Design and Implementation of Next Generation Ethernet-Based Hybrid Wired/Wireless Broadband Access Networks for Delivering Differentiated Services

By

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Abstract

DESIGN AND IMPLEMENTATION OF NEXT GENERATION ETHERNET-BASED HYBRID WIRED/WIRELESS BROADBAND ACCESS NETWORKS FOR DELIVERING DIFFERENTIATED SERVICES

By

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This thesis addresses the important problem of how to provide the broadband services available at the backbone to the end users with cost effective solution that is robust and reliable. The work is focused on developing, modeling and analysis of various architectures and algorithms to provide broadband services to wired as well as wireless users. We proposed hybrid wired/wireless architectures based on PON to provide broadband differentiated services to end users

The main body of the work is divided in to two main overlapping phases.

The first phase addresses a novel hybrid FSO/RF architecture based on PON. The proposed architecture provides cost effective solution for broadband services to end users that no other technology can compete. The first phase is divided in to 3 sections.

In the **first section**, we assess the feasibility, cost and the performance of the proposed architecture with that of a typical EPON architecture using the IPACT and limited DBA scheme. In simulation an event-driven packet-based simulation model was developed using C++. Two simulation programs with identical network parameters were developed, one for typical EPON using IPACT architecture and the other for the proposed architecture.

In the **second section** Fair Queuing Algorithms are provided to deliver differentiated services to the end users. To achieve fair queuing across different independent network flows in upstream direction, a fairness algorithm need to be implemented. A number of algorithms have been proposed to achieve fair queuing across different independent network flows in the literature, e.g., Deficit Round Robin (DRR) , Weighted Fair Queuing (WFQ), and Fair Queuing with Round Robin . Considering the fact that DRR is an important fair queuing algorithm that is used in CISCO's Gigabit Switch Router 12016, we are going to implement DRR for fair queuing algorithm at the OLT (Optical Line Terminal).QoS is implemented in down stream direction by OLT and in upstream direction by DU.

In the **third section** we proposes a novel fair queuing algorithm for hybrid wired and wireless access network with wireless compensation to provide QoS bounds to

Ethernet traffic for wired as well as wireless users with wireless error compensation. Scheduler maps priorities and weights for QoS of the Ethernet into wireless MAC.

In the second phases of the work we modified the architecture to deliver broadband differentiated services to mobile end users. The proposed Passive Optical Network (PON) based UMTS broadband wireless access network architecture provide multimedia services (video telephony, video streaming, mobile tv, mobile emails etc) to mobile users. In the conventional wireless access networks, the base stations (Node B) and Radio Network Controllers (RNC) are connected by point to point T1/E1 lines (Iub interface). the T1/E1 lines are not capable of supporting bandwidth (BW) required by next generation wireless multimedia services proposed by High Speed Packet Access (HSPA, Rel.5) for Universal Mobile Telecommunications System (UMTS) and Evolution Data only (EV-DO) for Code Division Multiple Access 2000 (CDMA2000).The proposed PON based back haul can provide Giga bit data rates and lub is shared by Node Bs. We also propose a novel algorithm to provide end to end QoS (between RNC and user equipment).The algorithm provides QoS bounds in the wired domain as well as in wireless domain with compensation for wireless link errors.

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Chapter 1

1. Introduction

1.1. Introduction

Recent rapid advances in broadband access technologies in wireline and wireless networks promises seamless, all-over access to unlimited information and entertainment to consumers and business users. End users are becoming more sophisticated and rich multimedia and real-time services are becoming more popular Mobile and fixed-line service providers (SPs) worldwide are being challenged to deliver new and innovative offerings beyond voice and basic data services to their customers. To meet these challenges, SPs are introducing dramatic changes in both their fixed and mobile broadband access networking infrastructures.

On the fixed broadband access side, there is a growing perception that copper access networks will soon no longer be able to meet the ever-growing consumer demand for bandwidth. Clever utilization of twisted pair has given some consumers network access of 128 Kb/S to 2.3 Mbps even though most access of this kind through digital subscriber line (DSL) is limited to 144 Kb/S [6].Cable modems (CM) can provide access at rates of about 30 Mbps. Because multiple users share same cable that reduces the data rate for per user

drastically in densely populated areas. Also DSL and CM have distance limitation. The maximum distance between central office and customer should be not more than 5.5 km. This, along with a combination of regulatory and competitive forces, as well as recent rapid advances and standardizations of Passive Optical Network (PON) technology, are finally prompting carriers around the world to consider PON-based Fiber-To-The-Home (FTTH) systems as possible successor to current copper-based access solutions. FTTH is the ultimate level of access, allowing end users to access the backbone networks through the gigabit capacity of a fiber optic cable.

A PON connects a group of Optical Network Units (ONUs) located at the subscriber premises to an Optical Line Terminal (OLT) located at the service provider's facility. It consists of a single, shared optical fiber (trunk) connecting an OLT to a passive star coupler (SC), which splits the downstream signal to multiple ONUs over dedicated short optical fiber. Traffic from an OLT to an ONU is called "downstream" (point-to-multipoint), and traffic from an ONU to OLT is called 'upstream' (multipoint-to-point). Among the various PON-based FTTH solutions, single channel Time-Division Multiplexed PON (TDM-PON) architecture is currently the most viable solution.

Due to their reduced operational and equipment costs, TDM-PONs have been widely accepted as a viable technology for the implementations of FTTH solutions, and are being deployed in the field in several places around the world. TDM-PON-based FTTH access solutions including broadband PON (BPON), gigabit PON (GPON), and Ethernet PON (EPON) are finally emerging into the mainstream and are set to revolutionize the access infrastructure worldwide. It is widely anticipated that, over the next decade, copper access networks worldwide will be largely replaced by fiber access networks, marking the beginnings of a new era of a mass migration to PON-based FTTH solutions.

On the wireless side, the mobile telecommunications industry is experiencing rapid growth both in terms of subscribers as well as high-speed mobile data services. Emerging mobile data services are expected to see the same explosive growth in demand that Internet and wireless voice services have seen in recent years. It is widely anticipated that demand for data-centric mobile services including web surfing, IPTV, music and video downloads, mobile TV, and other streaming services, will emerge as an equally significant market driver as that of the traditional mobile voice services. To meet the growing demands of both existing and forthcoming high-speed data services, service providers continue to evolve their mobile networks and services form 2G (second generation) to 2.5G to 3G and to beyond 3G (4G).

The first mobile services, which are commonly referred to as First Generation (1G) wireless, were analog. Analog systems were primarily based on circuit-switched technology and designed for voice, not data. In the 1990s, mobile services based on digital mobile technologies signaled the creation of the Second Generation (2G) of wireless services. 2G used several technologies including GSM (Global System for Mobile Communications), TDMA (Time Division Multiple Access), and CDMA (Code Division Multiple Access).CDMA and TDMA were deployed in the various parts of the U.S., while GSM was deployed as the common standard in Europe. 2G is also mostly based on circuit-switched technology. Extension of 2G system is introduced in 2.5 G systems such as General Packet Radio Service (GPRS) and Enhanced Data Rates for Global Evolution (EDGE). GPRS and EDGE are two interim technologies, which supported higher data rates than that of 2G networks and provided a migration path to the Third Generation (3G) of wireless services. Current 2G networks (e. g. GSM and CDMAone) are designed primarily for voice traffic with support of low speed data via GPRS or at enhanced rates using EDGE or CDMA20001x. Neither the first nor the second generations of wireless technologies were designed for multi-media services, such as the Internet.

The need to support a wider range of advanced services along with the diverse quality of service (QoS) and rate requirements set by these services have led to the development of 3G mobile telecommunications systems. The concept of 3G wireless technology represents a shift from voice-centric services to multimedia-oriented (voice, data, video, fax) services.3G wireless technology represents the convergence of various 2G wireless telecommunications systems into a single global system that includes both terrestrial and satellite components. 3G networks can use a variety of present and future wireless network technologies, including GSM, CDMA, TDMA, WCDMA, CDMA2000, UMTS and EDGE. Third generation networks such as Universal Mobile Telecommunications System (UMTS) and CDMA2000 1x EV-DO support higher data rates (up to 2Mb/s) and are currently well into deployment offering a variety of data services including mobile video and TV.

In anticipation of an increased demand for more advanced multimedia applications which require data rates higher than those supported by 3G, High-Speed Packet Access (HSPA) standards, which represent an ongoing evolution from 3G to 3.5G services, including High-Speed Downlink Packet Access (HSDPA) and High-Speed Uplink Packet Access (HSUPA) have been introduced in enhanced releases 6 and 7 of 3G UMTS. HSDPA delivers speeds comparable to or better than current fixed-line broadband access systems – up to 14.4 Mb/s peak air throughput per user, while HSUPA provides improved up-link performance of up to 5.76 Mb/s theoretically. Since 2006, UMTS networks in many countries have been or are in the process of being upgraded with HSDPA, with some networks offering full HSDPA 14.4 Mb/s downlink capacity and full HSUPA 5.76 Mb/s uplink capacity.

The ultimate vision is the evolution to a fully IP-based mobile infrastructure capable of providing 100 Mb/s and 1 Gb/s speeds both indoors and outdoors, with premium quality and high reliability. This is the envisioned Fourth-Generation wireless systems (4G), also known as beyond 3G. A 4G system will be able to provide a comprehensive IP solution where voice, data and streamed multimedia can be given to users on an "Anytime, Anywhere" basis, and at higher data rates than previous generations. The introduction of High Speed OFDM Packet Access (HSOPA) and Ultra Mobile Broadband (UMB) technologies are paving a smooth evolutionary path towards 4G and are often referred to as pre-4G.

High Speed OFDM Packet Access (HSOPA) is a Long Term Evolution (LTE) upgrade path for UMTS systems, which is also often referred to as Super 3G. HSOPA is currently under development, aiming for maximum transfer rates of 100 Mb/s for down-link and 50 Mb/s for uplink. UMB is part of a project within 3GPP2 to improve the CDMA2000 mobile phone standard for next generation applications and requirements. The system is based upon IP networking technologies running over a next generation radio system, with peak rates of up to 280 Mb/s down-link and 75 Mb/s uplink. The UMB standardization is expected to be completed soon, with commercialization taking place around mid-2009.

1.2. Thesis statement

Mobile networks and services have gone beyond voice-only communication services and are rapidly evolving towards data-centric services. Emerging mobile data services are expected to see the same explosive growth in demand that Internet and wireless voice services have seen in recent years. It is widely anticipated that demand for data-centric mobile services including web surfing, IPTV, music and video downloads, mobile TV, and other streaming services, will emerge as an equally significant market driver as that of the traditional mobile voice services. With this trend, mobile networks will certainly experience the same paradigm shift that has already occurred in wireline networks - namely the shift from voice-centric to data-centric networks.

In anticipation of this paradigm shift, mobile operators are currently planning to migrate their existing TDM-based mobile network infrastructure to a fully packet-based network. However, current mobile backhaul infrastructure remains the main hurdle to such a transition. The problem is further exacerbated by the fact that mobile backhaul accounts for up to 40% of total operational cost in 2G networks, and 60% or more in 3G. The migration to a packet-based mobile backhaul infrastructure that can support the emerging 3G/4G data-centric services along with the diverse QoS and rate requirements set by these services faces several technical as well as economical hurdles

To address some of these key challenges, this thesis argue for an immediate migration form current TDM-based mobile environment to fully packet-based mobile networks with a fiber based access infrastructure. Specifically, this work proposes and devises cost-effective multiservice Ethernet-based mobile backhaul networking architecture with a fiber based access infrastructure. The proposed architecture extends the reach of Ethernet access to base stations and provides a scalable mobile backhaul infrastructure that can accommodate emerging 3G/4G and legacy traffic and services as well as subscriber growth.

The key for a successful migration strategy rests entirely upon the cost-effectiveness of the proposed architecture. To achieve the most cost-effective migration strategy, rather than deploying a totally new packet-based mobile infrastructure, the proposed architecture utilizes the already existing EPON access infrastructure such that both RNC and OLT are collocated at the central office, while the BS is collocated with the ONU. The RNC along with each BS is equipped with an 802.3ah interface. Utilizing the existing wireline EPON infrastructure as a single packet-based radio access transport network enables a seamless migration path to fixed/mobile convergence and creates the potential of supporting several powerful and cost-effective packet-based fixed/mobile access architectural models.

There are several technical and economical drivers behind selecting native Ethernet as a viable alternative to TDM leased lines or ATM for transporting cellular traffic. However, Ethernet presents a major technical challenge when used as a transport technology in the RAN. Traditionally, Ethernet provides best effort traffic delivery and doesn't ensure end-to-end QoS guarantees. QoS support is missing due to the lack of connection admission control and/or packet scheduling mechanisms, which could introduce packet loss, jitter, and delay. To address this problem, the first phase of this work will focus on developing novel fully distributed DBA, packet scheduling, and queue management schemes to ensure that Ethernet can support differentiated class of services including voice, video, and best effort data, at least in a typical wireline EPON access environment.

As this work will show, integration of both inter-ONU/BS scheduling (upstream bandwidth allocation) and intra-ONU/BS scheduling (queue management and priority queuing at the ONU/BS) mechanisms facilitates the support of differentiated QoS across the proposed RAN. In contrast to traditional centralized EPON architectures, where the OLT arbitrates upstream transmissions by allocating an appropriate timeslot to each ONU, the integration process requires each ONU/BS to independently perform the process of upstream bandwidth allocation. It also requires direct intercommunication among ONUs/BSs. This in turn requires a fully distributed control plane among the ONUs/BSs for ONU/BS-ONU/BS communication as well as upstream access to the OLT/RNC.

This means that the conventional centralized EPON infrastructure where the process of upstream bandwidth provisioning is located at the distant OLT/RNC, and where there is no inter-ONU/BS communications at all, is not adequate for supporting the proposed RAN architecture. The challenge is to introduce a novel fully distributed EPON architecture, which forms the corner stone for the proposed RAN architecture, with as minor changes as possible to the conventional centralized architecture, while maintaining the IEEE 802.3ah traditional EPON standards.

To achieve our overall objectives, this work will be divided the main body of the work is divided in to two main overlapping phases.

The first phase addresses a novel hybrid FSO/RF architecture based on PON. The first phase is divided in to 3 sections.

In the first section, we assess the feasibility, cost and the performance of the proposed architecture with that of a typical EPON architecture using the IPACT and limited DBA scheme. In simulation an event-driven packet-based simulation model was developed using C++. Two simulation programs with identical network parameters were developed, one for typical EPON using IPACT architecture and the other for the proposed architecture.

In the second section Fair Queuing Algorithms are provided to deliver differentiated services to the end users. To achieve fair queuing across different independent network flows in upstream direction, a fairness algorithm need to be implemented. A number of algorithms have been proposed to achieve fair queuing across different independent network flows in the literature, e.g., Deficit Round Robin (DRR), Weighted Fair Queuing (WFQ), and Fair Queuing with Round Robin. Considering the fact that DRR is an important fair queuing algorithm that is used in CISCO's Gigabit Switch Router 12016, we are going to implement DRR for fair queuing algorithm at the OLT (Optical Line Terminal).QoS is implemented in down stream direction by OLT and in upstream direction by DU.

In the third section we proposes a novel fair queuing algorithm for hybrid wired and wireless access network with wireless compensation to provide QoS bounds to Ethernet traffic. Scheduler maps priorities and weights for QoS of the Ethernet into wireless MAC.The algorithm provides the means to transform each Ethernet traffic class into necessary parameters for the wireless channel scheduling. Each connection QoS is guaranteed by calculating bounds on delay and reserving the Bandwidth (BW).

The Ethernet traffic is divided in to three flows, each flow has different requirement for loss and delay. By supporting existing Ethernet traffic, scheduler avoids the need to redefine QoS parameters for wireless channel. Real time traffic uses fair-queuing model, non real time data use weighted round robin with re-circulating queue to accommodate wireless link error. Best effort traffic use FIFO mechanism. The algorithm is adjusted for wireless link failures which are location dependent and bursty.

In the **second phases** of the work we modified the architecture to deliver broadband differentiated services to mobile end users. The proposed Passive Optical Network (PON) based UMTS broadband wireless access network architecture provide multimedia services (video telephony, video streaming, mobile tv, mobile emails etc) to mobile users. In the conventional wireless access networks, the base stations (Node B) and Radio Network Controllers (RNC) are connected by point to point T1/E1 lines (Iub interface). The T1/E1 lines are not capable of supporting bandwidth (BW) required by next generation wireless multimedia services proposed by High Speed Packet Access

(HSPA, Rel.5) for Universal Mobile Telecommunications System (UMTS) and Evolution Data only (EV-DO) for Code Division Multiple Access 2000 (CDMA2000).

In addition, today's T1/E1 lines are not suitable for next generation wireless networks due to the following reasons: 1)T1/E1, which provide symmetric bandwidth in both uplink and downlink, so they are not well suited for bursty and asymmetric data traffic; 2)T1/E1s are source of reliability problems as adding redundancy through additional point to point links is expensive; 3) T1/E1s provisioning can take significant times (in some cases months) limiting the service providers ability to react changing demands; 4) T1/E1s are dedicated and can not shared dynamically.

The proposed PON based back haul can provide Giga bit data rates and lub is shared by Node Bs. We also propose a novel algorithm to provide end to end QoS(between RNC and user equipment).The algorithm provides QoS bounds in the wired domain as well as in wireless domain with compensation for wireless link errors.

Thesis Motivations:

It is thus quite clear from the above discussion that the technology is already in place to support quite advanced mobile data services, specifically IP/Ethernet-based services. It is also clear that mobile networks and services have gone beyond voice-only communication services and are rapidly evolving towards data-centric services. HSPA and CDMA 1x Evolution-Data Optimized (EV-DO) devices now on the market support peak download rates of several Mbit/s, which has led to a significant increase in data traffic volumes. With this trend, mobile networks will certainly experience the same paradigm shift that has already occurred in wireline networks - namely the shift from voice-centric to data-centric networks.

In anticipation of this paradigm shift, mobile operators are currently planning to migrate their existing TDM-based mobile network infrastructure to a fully packet-based network. However, current mobile backhaul infrastructure remains the main hurdle to such a transition. The problem is further exacerbated by the fact that mobile backhaul accounts for up to 40% of total operational cost in 2G networks, and 60% or more in 3G. The migration to a packet-based mobile backhaul infrastructure that can support the emerging 3G/4G data-centric services along with the diverse QoS and rate requirements set by these services, that is the focus of this thesis, faces several technical as well as economical hurdles including:

First, as outlined above, each new wireless generation provides wider range of services as well as new capabilities. Such high performance, however, comes at a price: an exponential increase in the bandwidth required to backhaul cellular traffic across the Radio Access Network (RAN) from individual Base Stations (BSs) to the Base station Controller (BSC) or Radio Network Controller (RNC). The BSC is used in 2G networks, while the RNC is used in 3G and later generation networks. Radio Access Network

(RAN) provides a connection between the end-user mobile station over air interface and the core network over a landline transmission network usually called – mobile backhaul.

The fundamental problem is that the majority of today's RANs are built on legacy TDM-based circuit switched infrastructures, where each BS is connected directly to the BSC/RNC (in "star topology" architecture) over dedicated point-to-point T1/E1 lines. Though acceptable for voice and low data rate applications (simple messaging services and email), T1/E1 capacity is inadequate for higher mobile data rates.

In addition, today's T1/E1 lines are not suitable for next generation wireless networks due to the following reasons: 1)T1/E1, which provide symmetric bandwidth in both uplink and downlink, so they are not well suited for bursty and asymmetric data traffic; 2)T1/E1s are source of reliability problems as adding redundancy through additional point to point links is expensive; 3) T1/E1s provisioning can take significant times (in some cases months) limiting the service providers ability to react changing demands; 4) T1/E1s are dedicated and can not shared dynamically.

With early deployment of HSDPA and HSUPA under way and HSOPA as well as UMB on the horizon, the exponential spike in capacity required to transport these data-intensive applications – with peak air throughput per user reaching 14-200 Mb/s, is far beyond what can realistically be achieved using today's T1/E1 leased lines. Assuming that the requisite additional T1/E1 lines were readily available (few tens to more than a hundred T1/E1 lines per BS) from the landline operator, they would be cost-prohibitive.

Another major limitation facing current traditional RAN architectures is the fundamental problem of inefficient utilization of limited available network resources. Because of the static-mapping nature of TDM transport, for each and every channel on the air interface, regardless if carrying data or idle, the appropriate channel resource must be allocated to that channel on the outgoing T1/E1 link. Furthermore, the TDM bandwidth is dedicated on a point-to-point basis from each base station all the way back to the BSC/RNC. This problem is more pronounced under typical non-uniform network traffic loads scenario, i. e., when some BSs, for a given interval, are heavily loaded while others are underutilized or are totally idle. In this case, the unused dedicated channel capacities of those lightly loaded/idle BSs cannot be shared by any of the other heavily loaded BSs attached to the BSC/RNC, leading to the waste of scarce network resources. There is no way to dynamically move capacity from a heavily loaded BS to a lightly loaded/idle BS even within the same cell site. In other words, additional backhaul expense results from unused bandwidth that is often stranded in the "wrong place at the wrong time." Therefore, it is essential that future RAN architectures must support dynamic bandwidth allocation (DBA) and sharing.

In addition to the aforementioned technical hurdles, which can be addressed, the most serious challenge to this migration scenario, however, is an economical one, namely, the decoupling of operating costs and revenues. It is well known that, as was the case in wireline, "the revenue per bit for data services is significantly lower than that for voice services“and, thus, this anticipated increase in data traffic will probably not be associated with an equal increase in revenue. It is therefore imperative to drastically reduce the cost per bit of transporting data traffic over tomorrow's packet-based mobile backhaul infrastructure and into the core network." Furthermore, it is equally important that this infrastructure must has the capability of efficiently supporting both existing services (which are paying for operating costs), and emerging 3G/4G data-intensive services, minimizing the bandwidth required and thereby the cost of supporting these services.

In view of the above discussion and reasoning, the following summarizes the status of current mobile networks: First, optimized for slow growing, narrowband, circuit-switched voice traffic, current mobile backhaul networks can't cope with the dynamic and bursty traffic pattern of the emerging 3G/4G data-centric multimedia services. Second, these T1/E1- based access transport networks lack the dynamic functionality and scalability needed to keep pace with the increasing volumes and unpredictability of data traffic. Third, the transition from current mobile backhaul infrastructure that is built primarily for voice traffic to a network supporting increasingly large volumes of data and ultimately multimedia traffic is not feasible via conventional approaches that apply point-solutions and work-arounds to the currently existing infrastructure.

Taking into account the current status of mobile backhaul infrastructure along with the fact that the ongoing evolution of mobile wireless services is dominated by technologies that support advanced data-intensive mobile services and are inherently packet-oriented, such as HSDPA and HSUPA, migration to a fully-packet-based RAN infrastructure is inevitable.

Thesis Statement:

To address some of these key challenges, this thesis argue for an immediate migration form current TDM-based mobile environment to fully packet-based mobile networks with a fiber based access infrastructure. Specifically, this work proposes and devises cost-effective multiservice Ethernet-based mobile backhaul networking architecture with a fiber based access infrastructure. The proposed architecture extends the reach of Ethernet access to base stations and provides a scalable mobile backhaul infrastructure that can accommodate emerging 3G/4G and legacy traffic and services as well as subscriber growth.

The key for a successful migration strategy rests entirely upon the cost-effectiveness of the proposed architecture. To achieve the most cost-effective migration strategy, rather than deploying a totally new packet-based mobile infrastructure, the proposed architecture utilizes the already existing EPON access infrastructure such that both RNC and OLT are collocated at the central office, while the BS is collocated with the ONU. The RNC along with each BS is equipped with an 802.3ah interface. Utilizing the existing wireline EPON infrastructure as a single packet-based radio access transport network enables a seamless migration path to fixed/mobile convergence and creates the potential of supporting several powerful and cost-effective packet-based fixed/mobile access architectural models including:

1. An independent stand alone Ethernet-based RAN architecture model that enables the backhaul of next generation (3G/4G) voice and data traffic and services over a single fully packet-based RAN infrastructure. It is shown that this architecture utilizes available network resources (bandwidth) in both upstream (BSs to RNC) and downstream (RNC to BSs) directions much more efficiently compared to that of a typical TDM-based circuit switched RAN architecture. This reduces the aggregate bandwidth consumed per BS freeing up capacity in the network to support heavily loaded BSs and/or new mobile data services. This is achieved via utilizing efficient DBA and sharing schemes, which is independently implemented by each BS in the upstream direction, as well as statistical multiplexing of voice and data traffic.
2. A near-term stand alone Ethernet-based RAN transitional model that enables the backhaul of current and next generations (2G/3G/4G) of voice and data traffic and services over a single packet-based RAN infrastructure. This model consolidates 2G/3G/4G access networks and provides a basis for graceful transition to a NG fully-packet-based network. Since 2G BSs have E1/T1 interfaces, a TDM-over-packet converter is required at each 2G BS to map T1/E1 circuits into Ethernet frames.

3. A hybrid model that support EPON's traditional fixed users as well as the packet-based RAN's mobile users. In this architecture, an ONU and a BS can be integrated into a single module in terms of both software and hardware. This model provides the best overall system performance in terms cost-effectiveness, bandwidth utilization, and QoS. This is because the integrated control module housed at each ONU/BS has global information about the entire fixed/mobile network status including the aggregate bandwidth requirements of both fixed and mobile users. Thus, the processes of bandwidth allocation and packet scheduling as well as prioritizing different class of services (for either fixed or mobile users) are globally optimized.

There are several technical and economical drivers behind selecting native Ethernet as a viable alternative to TDM leased lines or ATM for transporting cellular traffic including: 1) the inherent benefits of simplicity, flexibility, and low cost have uniquely positioned Ethernet as the leader for the inexpensive transport of packet-based technologies; 2) Ethernet naturally support IP services, which are expected to dominate NG 4G mobile services; 3) Ethernet offers more capacity at finer granularity; 4) some radio equipment manufacturers have already introduced Ethernet ports in their EV-DO BSs and their UMTS equipment; and 5) Ethernet services are provided over a standard, widely available and well-understood Ethernet interface.

Despite these advantages, Ethernet presents a major technical challenge when used as a transport technology in the RAN. Traditionally, Ethernet provides best effort traffic delivery and doesn't ensure end-to-end QoS guarantees. Although IEEE 802.1Q specifies three priority bits, Ethernet has no true class of service provision, such as DiffServ, and therefore cannot mark packets for prioritization, scheduling, and policing. QoS support is missing due to the lack of connection admission control and/or packet scheduling mechanisms, which could introduce packet loss, jitter, and delay. To address this problem, the first phase of this work will focus on developing novel fully distributed DBA, packet scheduling, and queue management schemes to ensure that Ethernet can support differentiated class of services including voice, video, and best effort data, at least in a typical wireline EPON access environment.

As this work will show, integration of both inter-ONU/BS scheduling (upstream bandwidth allocation) and intra-ONU/BS scheduling (queue management and priority queuing at the ONU/BS) mechanisms facilitates the support of differentiated QoS across the proposed RAN. In contrast to traditional centralized EPON architectures, where the OLT arbitrates upstream transmissions by allocating an appropriate timeslot to each ONU, the integration process requires each ONU/BS to independently perform the process of upstream bandwidth allocation. It also requires direct intercommunication among ONUs/BSs. This in turn requires a fully distributed control plane among the ONUs/BSs for ONU/BS-ONU/BS communication as well as upstream access to the OLT/RNC.

Note that the processes of moving the functionality of the packet scheduler and the upstream DBA module from the RNC to the BSs as well as achieving direct intercommunication among BSs are in full compliance with 3G standards and the vision of 4G. This also leads to efficient radio resource scheduling, optimum upstream bandwidth allocation, and faster re-transmission. Another important advantage is that some of the burden has been off loaded form the RNC to the BSs. This means that the conventional centralized EPON infrastructure where the process of upstream bandwidth provisioning is located at the distant OLT/RNC, and where there is no inter-ONU/BS communications at all, is not adequate for supporting the proposed RAN architecture. The challenge is to introduce a novel fully distributed EPON architecture, which forms the corner stone for the proposed RAN architecture, with as minor changes as possible to the conventional centralized architecture, while maintaining the IEEE 802.3ah traditional EPON standards.

Chapter 2

2.1. Broadband wireless access Networks and Last-mile Problem

2.1.1. Introduction

Fiber optic network exist worldwide and the amount of installed fiber will grow with the implementation of wavelength division multiplexing (DWDM) the information carrying capability of fiber networks has increased enormously. The worldwide demand for broadband communications is being met in many places by installed fiber networks. At least 10 Tb/S of capacity on a single fiber has been demonstrated of early 2002.

However there is still a significant “last/first mile “problem. Which seriously limits the availability of broadband to the subscriber. External bandwidth of 10 Mb/s, 100 Mb/s or even more are not unusual for business.

Passive Optical Networks (PON) has been proposed as an alternate solution. But in metropolitan area the optical fiber network is still limited due to high cost of fiber installations or simply right of way issue. High ways ,rivers and bridges are also problems. Typically only small buildings are connected by fiber. However vast majority is with in 1 Km, even in many cases much shorter.

We proposed a PON based hybrid free space optical (FSO) and Radio frequency(RF) communication systems as solutions to provide high-speed local loop connectivity in metropolitan area network environments.

Research demonstrated that FSO can provide the same data capacity like a DWDM [1].For reliable operation over 1 km range an optical wireless system can have easily foot print diameter at the receiver of only 50 mm (after pointing and adjustment, with out that 1 to 2 meter .)[2]

Four multiplexed 2.5 Gb/s channel[3-5] of 1550 nm over a 4.4 Km has been demonstrated and modeled by lucent. Terra link 8-155 did an experiment which provide a full duplex wireless communication link at data rates of up to 230 Mbps at distance of 8 km. OW systems may provide high bandwidth communication channel over several km .In the remaining sections of this chapter we will explain the theory and back ground of PON networks and FSO technology.

2.2. Free Space Optics

Free Space Optics (FSO) is a fibreless, laser driven technology that supports high bandwidth, with easy to install connections for the last-mile and campus environment. The light pulses are transmitted through the atmosphere in a small conical shaped beam by means of low power lasers or LED’s. Free space optics require line of sight availability between the laser/receiver units .The units can be mounted on building tops ,sides or even behind windows. The units are full duplex meaning that data can flow in both directions simultaneously. The lasers are low power and do not constitute a risk to the nacked eye or any bird or animal that might get in the laser path. The frequencies used by the lasers are between 750 and 1550 Ghz and do not require licensing like other wireless devices.

Unlike fiber systems, free space optical link propagate with in a medium randomly changing. So there are atmospheric effects on the the light signal passing through it

2.2.1. Atmospheric effects

Atmospheric effects[6-8] can deteriorate free space link transmission by two ways:

- Overall reduction in detected optical power level called atmospheric attenuation.
- Random optical fluctuations in received beam deformation, scintillation effects and beam wander.

a. Atmospheric attenuation:

Atmospheric absorption from particulates and aerosols interacting with optical beam over the link span. The phenomena had accumulate effects upon over all received power level signal as well as causing fluctuations.

The transmitted optical power at any specific distance is given by Beer's Law

$T[R]=P[R]/P[0] = e^{-\sigma*R}$

Where

T[R]=Transmittance at range R

P[R]=Link power at range R

P[0]=Initial launched data link optical power

σ=Attenuation constant per unit length

σ=σm+σa+βm+βa

σm =Molecular absorption coefficient .They have high absorption in the infrared band which include absorption by water , CO_2 and ozone molecules.

σa =Aerosol absorption coefficient.

Which is the result of dispersed solid and liquid particles with in the atmosphere, such as ice, dust and organic particles varying in size to maximum of 40 µm diameters.

βm=Rayleigh scattering coefficient

Which is the result of light interaction with particles which are much smaller than the propagating wave length. The Rayleigh scattered power coefficient follows λ-4 law and decrease with wave length.

βa=Mei scattering coefficient

Mei scattering dominates the total attenuation coefficient when the size of atmospheric particles are of the order of wavelength. It can be expressed as

= σ=[3.91/ v][λ/550] -Q

V= visibility (km)

λ=Wave length (nm)

Q=Size distribution of particles

=1.6V>50 km high visibility

=1.3 average visibility (6<V<km)

=.585 V1/3 for low visibility (V<6km)

b. Scintillation:

As the beam propagates through the atmosphere it experiences deterioration and deformation of its wave front. These degradation are because of the atmospheric turbulence, from wind or from temperature difference between the ground and the air, produce temporary packets of air with slightly different temperature, density, and index of refraction.

These packets are in constant motion and are also continuously changing size as they are being created, mixed and destroyed A laser beam wave front propagating through turbulence becomes deformed and distorted since these packets act as small lenses and prisms.

A random pattern of dark and light spots will form as constructive and destructive interference produce a spatial redistribution of light intensity across the beam front. These dark and light spots are in constant motion across the receiver aperture. These fluctuations in received power are similar to the twinkling of distant star.

The time scale of these fluctuations is comparable to the time it takes the volume of air to move across the beam path due to wind. When the scintillation are week (fluctuations are small). The scintillation is given by log normal distribution as predicted by Rytov theory. But it tend to saturate at a value

$\boldsymbol{\sigma x}$=3.35

The log normal distribution (normalized so that that the integrated probability is 1 and the mean intensity is 1) is given by

$P (I,\sigma x^{2}) = 1/ 2I [2\Pi\sigma x^{2}]^{½} \quad \exp \ -(\ln I)+2\sigma x^{2}) /(8\sigma x^{2})$

The decaying exponential distribution provides a better fit to large fluctuation data exp(-I).The actual distribution probability lies in between the two distributions.

c. Geometrical beam expansion:

Geometrical loss is the fixed loss for a given free space optical data link system. It is the loss encountered as the beam travels and expands along its trajectory. The coupling loss in decibels (dB) at a specified link span is given by the following expression

10 * log (Arec /Aw.f)

Where

Arec = Effective receive area for the telescope

Aw.f =Effective cross-section area of the propagating wave front at a range R .

d. Optical miss pointing:

Moving building –Buildings often sway from side to side or oven settle further in to the ground. They are also introduced because of mechanical vibration of all components with in the optical link The degree to which both transmitting and receiving telescopes are aligned and remain align will determine. Auto tracking capability can be used to over come this loss.

e. Atmospheric effects:

Optical power link budget

T[R]=P[R]/P[0]

$$=\sigma m+\sigma a+\beta m+\beta a +10 * \log (Arec /Aw.f)+Sc$$

2.2.2. Compensation for Atmospheric effects:

Scintillation introduced fades can be reduced by [19]

- Aperture averaging :Increased receiver aperture size decreases the variance (log amplitude) of the received signal
- Stochastic smoothing: Multiple incoherent beams are transmitted to produce independent speckle pattern at the receiver .The receiver sums over the fades produced by the multiple beams. This is form of spatial diversity.
- Retransmission: Transmitting two copies of the same data through the same path one is delayed and the other is not delayed .At the receiver maximal combining techniques are used to recover the data. The delay needs to be greater than the scintillation fade duration. two fold techniques to improve it. Two-fold technique has penalty of 3 db but provide 20-30 db gain on the scheme [38-4].

- Active and adaptive optics: These are new technologies for commercial FSOC application. Active optics is a powerful techniques for beam steering that combine with other approaches yields significant benefits to FSOC applications. Adaptive optics helps to reduce the atmospherically induced wave front distortions. This method is currently expensive and complex.

- Error correction coding and interleaving:

2.2.3. Optical wireless advantages:

- The primary advantages of FSO over fiber are its rapid deployment and significant cost saving [1].
- Allowing transmission of digital computer data, video, voice over IP, multiplexed data or ATM. Optical wireless virtual point allows each subscriber able to connect to base site using any protocol or data rate independently of others in the cell.
- Building to building connectivity .LAN to LAN and MAN, corporate and university campuses, military bases, hospitals to doctor's offices.
- **High quality video transmission**
    - ♦Sporting event
    - ♦Surveillance
    - ♦Traffic monitoring
    - ♦Video conferencing
- **Deployable or temporary connectivity Increase band**
    - ♦Natural disaster disruption.
    - ♦Relay of military test rang data.
    - ♦Link establishment before fiber installation.
    - ♦Current wireless systems not adequate.

- **Increase bandwidth to or from Internet service providers (ISP's) LAN to WAN**
    - ♦Replacement of high priced leased line.
    - ♦High bandwidth services to apartment, town house, mall, and hotel complex.
- Eliminate right of way access problems: Major high ways, rivers, lakes, terrain, that is rocky or forested, urban areas land where obtaining right of way not possible.
- Security –optical wireless uses narrow beams narrow beams (typically .1 -.5 degrees) with no side lobes or antenna rear emission
- Transparency to all protocols (connect to various network in seamless manner)
- Bandwidth on demand
- Market (Equipment available in the market)
- Instant access to all information through databases, high-speed link and cost effective and maintenance free.

2.2.4. Optical Wireless Disadvantages:

1- Require totally clear line of sight.

2- If obstruction such as birds and aircraft fly through the laser, they will impede data transmission temporary

- Most protocols automatically retransmit any lost data
- Meshed systems always provide another path.

2.3. A Millimeter wave Broadband Wireless Access Technology

Millimeter wave technology at 60 GHz is unlicensed due to atmospheric losses, and is capable of higher capacity than frequencies at higher wavelengths [9-11]. There are three components of loss: free space loss, gaseous loss and loss due to particulate scattering ,primarily from precipitation. Millimeter wave systems at 60 GHz are limited by the FCC to output powers [12] of about 500 mW(total radiated power)for license free operation. System designers are free to choose the antenna size ,which generally dictates its gain. The size of antenna determines the amount of intercepted millimeter wave energy as well as determine the beam divergence, since the system is diffraction limited. For reasonably sized systems ,with an antenna size near 13 inch in diameter ,the clear air range is 1500 meters[12].

Rain is the primary obscurant affecting the performance of 60 GHz systems. Particularly in regions where heavy rain is common ,rain induced attenuation limits the carrier grade range to less than 500 meters.

2.4. PON Access Communication

A PON [13-15] consist of equipment located in a service provider’s local exchange (OLT). Which is connected to few optical network unit terminal (ONU or ONT) located in building, curb or home.

A Passive Optical Network (PON) is a group of technologies originally created by the Full Service Access Network (FSAN) working group and now standards of ITU-T and IEEE, allowing fiber as the first mile (or last mile) to the customer premises. A PON consists of a central office node Optical Line Termination (OLT) at the service providers office and a number of Optical Network Units (ONUs) near end users, and the fibers and splitters between them, called the optical distribution network (ODN). The OLT provides the interface between the PON and the backbone network, while the ONT provides the service interface to the end user. PON is a converged infrastructure that can carry multiple services such as voice (plain old telephony service or voice over IP), data, video, and/or telemetry, in that all of these services are converted and encapsulated in a single packet type for transmission over the PON fiber.

In a PON configuration, downstream signals are broadcast to each premises sharing a fiber. Encryption is used to prevent eavesdropping. Upstream signals are combined using a multiple access protocol, invariably time division multiple access (TDMA). This allows for two-way traffic on a single fiber optic cable. The main fiber run on a PON network can operate at 155 Mps, 622 Mbps, 1.25 Gbps or 2.5 Gbps using APON/ BPON, EPON or the emerging GPON standards. Bandwidth allocated to each customer from this aggregate bandwidth can be static or dynamically assigned in order to support voice, data and video applications. In PON we can have four different approaches as shown in figure 2.1. We will use approach 3 (PON : a distributed switch) through out

in our work because of its benefits over the other two approaches.

<u>i. Point-to-point links</u>

a) Two many fiber lines

b) 2N transceivers.

<u>ii. Concentration switch in the neighborhood</u>

a) Power in the field

b) Band receivers used are 2N+2

iii. PON a distributed switch

a) Minimum fiber

b) N+1 transceivers

c) Path transparency

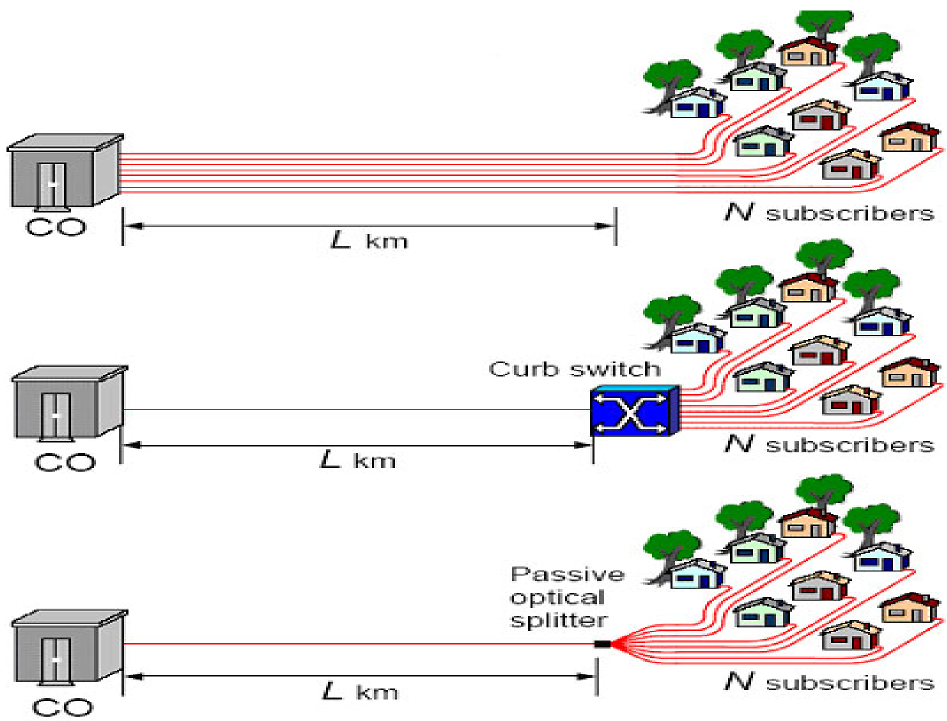


Figure 2.1. Various approaches for PON architecture

There are various flavors of PON technologies including APON, BPON, EPON and GPON.

2.4.1. APON/BPON (ITU-T G.983):

APON, ATM PON, is the initial PON specifications defined by the FSAN committee used ATM as their layer 2 signaling protocol. Use of the term APON led users to believe that only ATM services could be provided to end-users, so the FSAN decided to broaden the name to Broadband PON (BPON). BPON systems offer numerous broadband services including Ethernet access and video distribution.

APON systems are based upon ATM as the bearer protocol. Downstream transmission is a continuous ATM stream at a bit rate of 155.52 Mb/s or 622.08 Mb/s with dedicated Physical Layer OAM (PLOAM) cells inserted into the data stream. Upstream transmission is in the form of bursts of ATM cells, with a 3 byte physical overhead appended to each 53 byte cell in order to allow for burst transmission and reception.

2.4.2. GPON (ITU-T G.984):

Gigabit PON is a PON technology operating at bitrates of above 1 Gb/s. Apart from the need to support higher bitrates, the overall protocol has been opened for re-consideration and the sought solution should be the most optimal and efficient in terms of support for multiple services, OAM&P functionality and scalability.

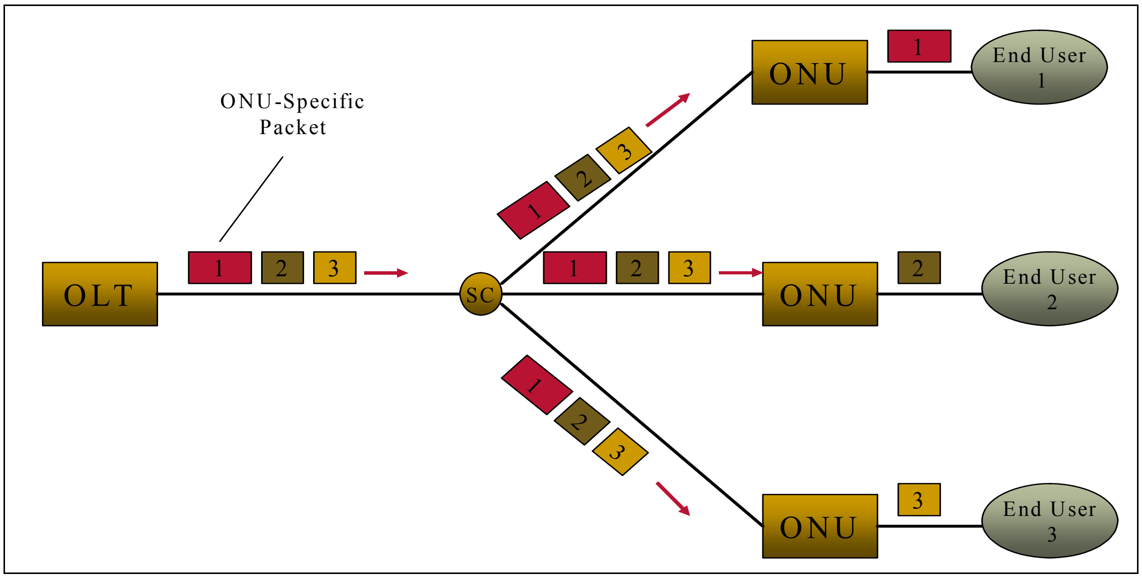
The main GPON requirements are:

- Full Service Support C including voice (TDM, both SONET and SDH), Ethernet (10/100 BaseT), ATM, leased lines and more.
- Physical reach of at least 20 km with a logical reach support within the protocol of 60 km.
- Support for various bitrate options using the same protocol, including symmetrical 622 Mb/s, symmetrical 1.25 Gb/s, 2.5 Gb/s Downstream and 1.25 Gb/s Upstream and more.
- Strong Operation Administration and Maintenance (OAM&P) capabilities offering end to end service management.
- Security at the protocol level for downstream traffic due to the multicast nature of PON.

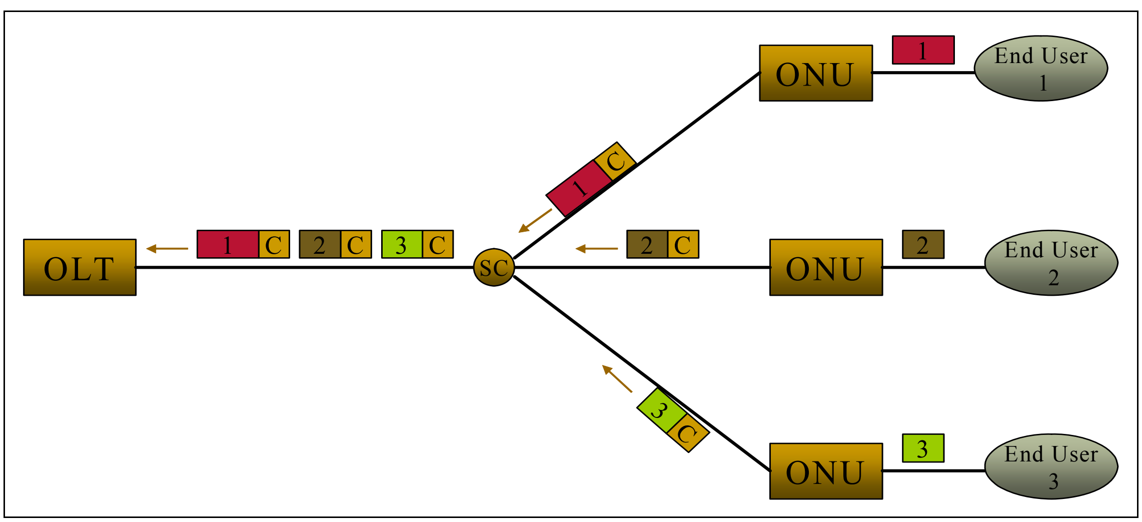
2.4.3. EPON (IEEE 802.3ah)

Ethernet Passive Optical Network (EPON) is a point to multipoint (Pt-MPt) network topology implemented with passive optical splitters, along with optical fiber PMDs that support this topology. EPON is based upon a mechanism named MPCP (Multi-Point Control Protocol), which uses messages, state machines, and timers, to control access to a P2MP topology. Each ONU in the P2MP topology contains an instance of the MPCP protocol, which communicates with an instance of MPCP in the OLT. At the basis of the EPON/MPCP protocol lies the P2P Emulation Sublayer, which makes an underlying P2MP network appear as a collection of point to point links to the higher protocol layers (at and above the MAC Client). It achieves this by prepending a

Logical Link Identification (LLID) to the beginning of each packet, replacing two octets of the preamble. In addition, a mechanism for network Operations, Administration and Maintenance (OAM) is included to facilitate network operation and troubleshooting.



EPON Downstream overview



EPON Upstream overview

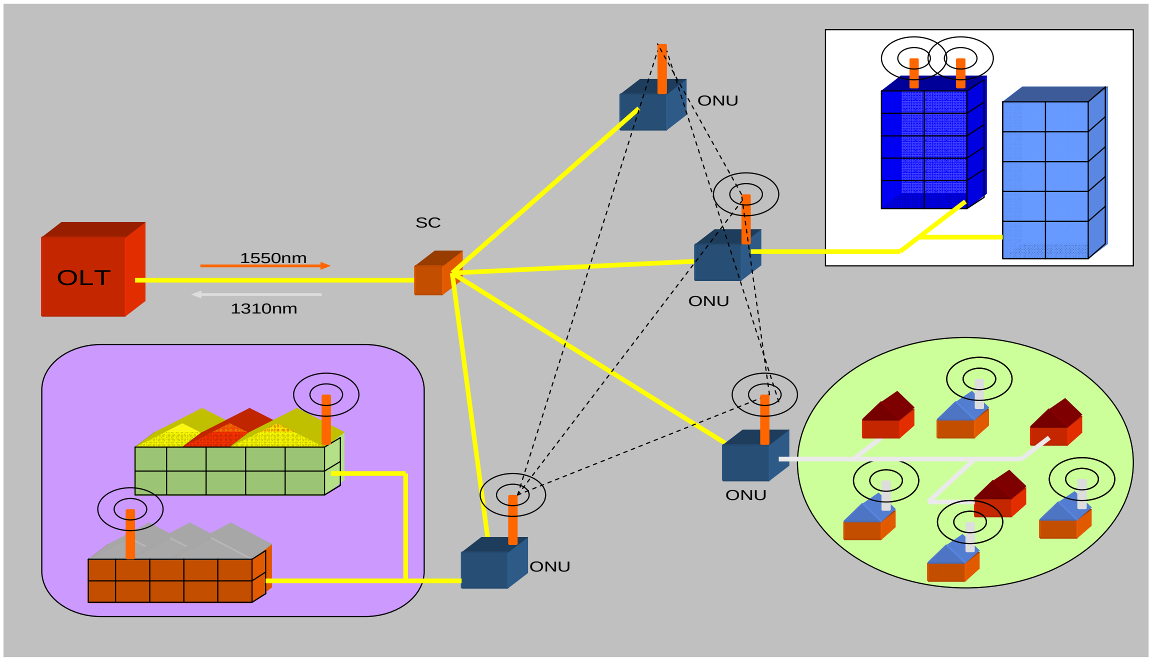


Figure 2.2. PON: Passive Optic Network Technologies (APON/BPON, EPON and GPON)

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Chapter 3

3.1. Self-similar traffic

3.1.1. Introduction

Ethernet local area network (LAN) traffic is statistically self-similar, that none of the commonly used traffic models is able to capture this fractal behavior, and that such behavior has serious implications for the design, control,and analysis of high-speed, cell-based networks. Intuitively, the critical characteristic of this self-similar traffic is that there is no natural length of a "burst": at every time scale ranging from a few milliseconds to minutes and hours, similar-looking traffic bursts are evident; we find that aggregating streams of such traffic typically intensifies the self-similarity ("burstiness") instead of smoothing it.

Throughout our simulations we will use constant bit rate (CBR) for voice traffic ,video and data traffic we will use self similar traffic. Understanding the nature of network traffic is critical in order to properly design and implement computer networks and network services like the World Wide Web.

Ethernet LAN traffic is very different both from conventional telephone traffic and from currently considered formal models for packet traffic (e.g., pure Poisson or Poisson-related models such as Poisson-batch or Markov-Modulated Poisson processes (Heffes and Lucantoni (1986)), packet-train models (Jain and Routhier (1986)), and fluid flow models (Anick et al. (1982)), etc.). These differences require a new look at modeling the traffic and performance of broadband networks. Rather, measurements of real traffic indicate that significant traffic variance (burstiness) is present on a wide range of time scales.

Traffic that is bursty on many or all time scales can be described statistically using the notion of self-similarity. Self-similarity is the property we associate with fractals --- the object appears the same regardless of the scale at which it is viewed. In the case of stochastic objects like timeseries, self-similarity is used in the distributional sense: when viewed at varying scales, the object's distribution remains unchanged.

Since a self-similar process has observable bursts on all time scales, it exhibits long-range dependence; values at any instant are typically correlated with values at all future instants. Surprisingly (given the counterintuitive aspects of long-range dependence) the self-similarity of Ethernet network traffic has been rigorously established.

3.1.2. Generating self-similar traffic

The self-similar or long-range-dependant (LRD) network traffic can be generated by multiplexing several sources of Pareto-distributed ON and OFF periods [3]. In a context of a packet-switched network the ON periods correspond to packet train – packets transmitted back to back, or separated only by a relatively small preamble (as defined in IEEE standard 802.3, for example). OFF periods are the periods of silence between packet trains.

Multiple sources contributing to resulting synthetic traffic trace may be thought of as individual flows (connections). It is reasonable to assume that packet sizes within a connection remain constant. Different connections, however, will have packets of different sizes.

To generate a Pareto-distributed sequence of ON periods, one can generate a Paretodistributed sequence of packet train sizes. The minimum train size is 1, which corresponds to a single packet transmitted.

Pareto distribution has the following probability density function:

$$P(x) = \frac{\alpha b^{\alpha}}{x^{\alpha+1}} \quad , \quad x \geq b \qquad \ldots\ldots 1$$

Where α is a shape parameter (tail index), and *b* is minimum value of *x*. When $2 \leq a$, the variance of the distribution is infinite. When $1 \leq a$, the mean value is infinite as well.

For self-similar traffic, α should be between 1 and 2 [4,5]. The lower the value of α, the higher the probability of an extremely large *x*. Figure 1 shows the density functions for various values of α.

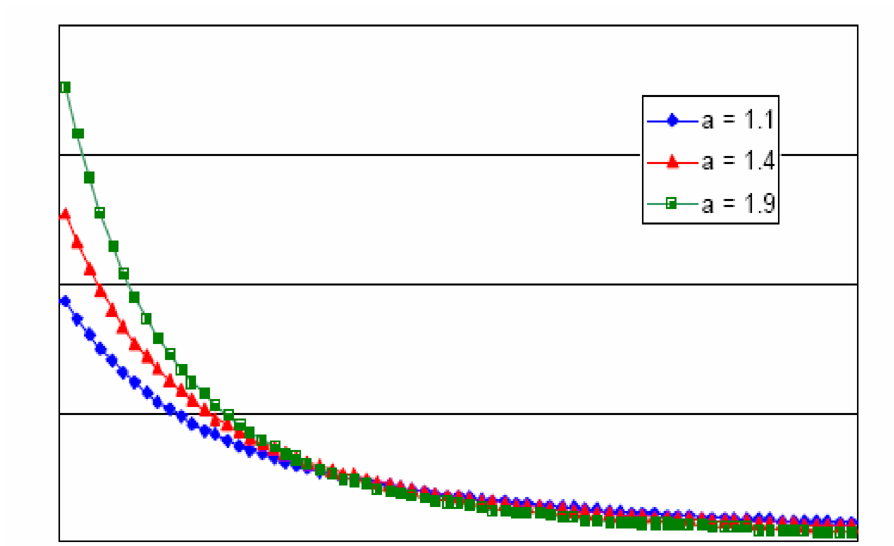

Figure 3.1. Probability density functions for Pareto distribution with a = 1.1, 1.4, 1.9

Mean value of a Pareto distribution equal is

$$E(x) = \frac{\alpha b}{\alpha - 1} \qquad \text{……...2}$$

The formula to generate a Pareto distribution is

$$X_{PARETO} = \frac{b}{U^{1/\alpha}} \qquad \text{……...3}$$

where U is a uniformly distributed value in the range (0, 1)

Very often it is desirable to generate a synthetic traffic of a predefined load. Obviously, the resulting load *L* is just a sum of loads *L_i* generated by each individual source *i*. Given N sources,

$$L = \sum_{i=1}^{N} L_i \qquad \text{……...4}$$

Thus, it is important to be able to get a good estimation of the load generated by one source. The load generated by one source is mean size of a packet train divided over mean size of packet train and mean size of inter-train gap, or putting it differently, it is a mean size of ON period over mean size of ON and OFF periods.

$$L_i = \frac{\overline{ON_i}}{\overline{ON_i} + \overline{OFF_i}} \qquad \text{……………………………………………………………...……………5}$$

Formula (2) gives the mean value of a true Pareto distribution. However computers using equation (3) generate a pseudo-Pareto distribution. One of problems comes from the fact that computers cannot generate arbitrarily large value. However, any true Pareto distribution of sufficiently large length will have values that exceed the range generated by computers. Thus, what we have is a truncated-value distribution.

Let's denote *S* to be the smallest non-zero value that uniform random generator may produce. Then, the generated Pareto-distributed values will not exceed *q*:

$$q = \frac{b}{S^{1/\alpha}} \qquad \text{……………………………………………………………………………………………6}$$

Then, the mean value of a Pareto distribution can be calculated as shown below:

$$E(x) = \int_b^q x\, f(x)\, dx = \int_b^q x \frac{\alpha b^\alpha}{x^{\alpha+1}} dx = \alpha b^\alpha \int_b^q \frac{dx}{x^\alpha}$$

$$= \alpha b^\alpha \left. \frac{x^{1-\alpha}}{1-\alpha} \right|_b^q = \frac{\alpha b}{\alpha - 1}\left[1 - \left(\frac{b}{q}\right)^{\alpha-1}\right] \qquad \text{…………...……7}$$

Substituting (6) into (7) we get

$$E(x) = \frac{\alpha b}{\alpha - 1}\left[1 - S^{\frac{\alpha - 1}{\alpha}}\right] \qquad \ldots\ldots 8$$

Equation (8) gives the mean value of a truncated-value Pareto distribution. Now, if we are given load *Li* and the packet size *k* for a given source, we can find the minimum value of the OFF period.

First, let's find the mean value of OFF period. From equation (2) we get

$$\overline{OFF_i} = \overline{ON_i}\left(\frac{1 - L_i}{L_i}\right) \qquad \ldots\ldots 9$$

Let's denote M_{OFF} and M_{ON} to be the minimum ON and OFF periods respectively. We mentioned above that the minimum packet train size is just one packet, i.e., M_{ON} =

$$\frac{M_{OFF}\alpha_{OFF}}{\alpha_{OFF} - 1}\left[1 - S^{\frac{\alpha_{OFF} - 1}{\alpha_{OFF}}}\right] = k\frac{M_{ON}\alpha_{ON}}{\alpha_{ON} - 1}\left[1 - S^{\frac{\alpha_{ON} - 1}{\alpha_{ON}}}\right]\left(\frac{1 - L_i}{L_i}\right) \qquad \ldots\ldots 10$$

where α_{ON} is the shape parameter for the ON periods, and α_{OFF} is the shape parameter for the OFF periods.

$$T_{ON} = \frac{\alpha_{ON} - 1}{\alpha_{ON}} \text{ and } T_{OFF} = \frac{\alpha_{OFF} - 1}{\alpha_{OFF}}$$

we get

$$M_{OFF} = k\frac{T_{OFF}}{T_{ON}} \times \frac{1 - S^{T_{ON}}}{1 - S^{T_{OFF}}} \times \left(\frac{1}{L_i} - 1\right) \quad \text{.........11}$$

Thus, given values for k, L_i, α_{ON}, and α_{OFF}, the formula (10) gives us the value for M_{OFF} that would result in link load closer to L_i.

However, if we generate traffic using the above formulas, we will notice the mean values for ON and OFF periods in the generated series still slightly off.

The problem appears to be in the way computers generate Pareto-distributed values (formula 3). While Pareto distribution assumes continuous sample space, computers generate discrete values with uniform probability. The Pareto-like distribution is achieved by having higher density of samples toward lower end of the scale.

If we build distribution function by aggregating samples over some window size, we will get a plot somewhat close to the one shown in Figure 1. But still, no matter how large our window is, at the tail end the distance between two neighboring points will exceed the window size. That means that some windows will contain zero samples, even if number of samples approach infinity. Of course, that introduces an error to the mean size of ON and OFF periods.

To correct for this error, we found that the calculated values $\overline{ON}$ and $\overline{OFF}$ should be multiplied by coefficient C

$$C = (1.19\alpha - 1.166)^{-0.027} \quad \text{.........12}$$

Thus, formula (11) becomes

$$M_{OFF} = k \times \frac{C_{ON}}{C_{OFF}} \times \frac{T_{OFF}}{T_{ON}} \times \frac{1 - S^{T_{ON}}}{1 - S^{T_{OFF}}} \times \left(\frac{1}{L_i} - 1 \right) \qquad \text{………………………………..…..13}$$

On a final remark, if we choose α_{ON}, and α_{OFF} to be the same, the equation (13) will reduce to

$$M_{OFF} = k \times \left(\frac{1}{L_i} - 1 \right) \qquad \text{…………………………………………………………………………….14}$$

That, however, may limit the usefulness of the traffic generator. It is very reasonable to assume that in real traffic, probability of having extremely large OFF period is higher then the probability of having extremely large ON period. That means that the shape parameter α_{ON} should be larger than α_{OFF} . The above heuristic coefficient results in generated load being very close to the specified load with all combinations of α_{ON}, and α_{OFF} .

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Chapter 4- Design and Performance of Hybrid FSO/RF architecture

Chapter 4

4.1. Design and Performance of Hybrid FSO/RF architecture for Next Generation Broadband Access Networks

4.1.1. Introduction

Recent advances in optical networking technologies have fueled tremendous growth in both backbone and metropolitan access network (MAN) capacity. At the same time, end users are becoming more sophisticated and rich multimedia and real-time services are becoming more popular. The conduits linking the high speed end- user equipment to the high capacity backbone networks, however remain a bottleneck. These connections are commonly referred to as “last/first mile”.

The recent advancements in last mile technology have increased capacity from the range of 56kb/s for a dial-up modem to a few Mb/s. Clever utilization of twisted pair has given some consumers network access of 128 Kb/S to 2.3 Mb/S even though most access of this kind through digital subscriber line (DSL) is limited to 144 Kb/S [1].Cable modems can provide access at rates of about 30 Mb/S. Because multiple users share same cable that reduces the data rate for per user drastically in densely populated areas. Also

DSL and CM (cable modem) have distance limitation. The maximum distance between central office and customer should be not more than 5.5 km.

If we use point to point RF wireless as an alternative, there are unwanted side lobes. For a transmission aperture of diameter D the diffraction angle is θdiff ~λ /D. Consequently for equivalent sized apertures a microwave signal at 2 GHz has a diffraction angle almost 100,000 times larger than a laser operating at 1.55 mm. So the microwave signal has spread in the area 100,000 times larger than the highly directional laser beam. The high carrier frequency for example 1.55 mm has 100,000 times higher carrying capacity than a 2 GHz microwave signal [1].

Fiber optical networks passive optical network (PON) based-solutions are emerging as viable choices that might address the first/last mile access problems[10-12].

PON based broadband access networks include broadband PON(BPON), Gigabit PON(GPON), and either PON (EPON)., However, PON have been very limited due to the high cost of fiber installation. There are also regulatory restrictions and right-of-way issues. The current cost of installing fiber optic infrastructure within a city in north America can be up to $1 million/mile[1] (FSO/RF link the cost for 1km link is around approximate10,000-18,000 dollars). There is also sunk cost in case the end users move to some other location. The buried fiber links are difficult to reconfigure, and scalability problems. That's why very few commercial buildings in an access network are connected to a high-speed networking backbone (carrier's central office) via optical fiber, even though a vast majority of these buildings are within 1 km or less from central office.

Taking these limitations in to account, utilizing line-of-sight broadband wireless communications e.g. Free Space Optics (FSO) and/or RF microwave systems offer the potential of lowering the cost and expediting the implementation of broadband access network.

The newly emerged FSO technology can provide much more capacity than any other competing wireless technologies such as RF or microwave systems. The FSO communication technology offers multi giga-bit capacity without requiring fiber connectivity between link ends. The optical communication path is line of sight and directed through the atmosphere. However the main impediant facing FSO technology is the adverse weather conditions such as dense fog or steam (FSO is transparent to rain), which limits the optical wireless link to about 300 meters or less (under dense fog condition).FSO not only provide massive data rate but also freedom regulation , and offer physical link security because of their fundamentally low probability of intercept or detection. Free space optical links can be easily redirected and reconfigured in a multiple connected topology to improve network performance.

The unlicensed RF 60 GHz (millimeter) wave can provide high bandwidth like FSO .Millimeter wave(MMW) is transparent to fog but highly susceptible to rain attenuation ,clear oxygen absorption and is limited to 500 meters or less under heavy rain condition .But RF is transparent to fog. Thus as for as weather impact, optical and microwave technologies are complementary.

By combining the benefits of both FSO and RF technologies we proposed a novel Ethernet-based broadband access networking architecture that utilizes the existing wired trunk feeder fiber of typical EPON infrastructure (from OLT-SC) along with a hybrid FSO/RF reliable wireless connectivity to the end-users (ONUs). Among all the PON networks, we selected EPON because Ethernet currently accounts for more than 85 % of all installed connections and more than 95% of all LANs[10].

In the distribution segment (from SC to ONUs), two main features distinguish the proposed wired/wireless architecture from that of typical EPON architecture:

1.The passive star coupler(SC) is replaced by an active Distribution Unit(DU).

2.In contrast to typical EPON architecture where end users (ONUs) are connected by short distribution fibers, the proposed architecture provides reliable wireless broadband access using a hybrid FSO/RF link.

The architecture for EPON is shown in figure 4.1.

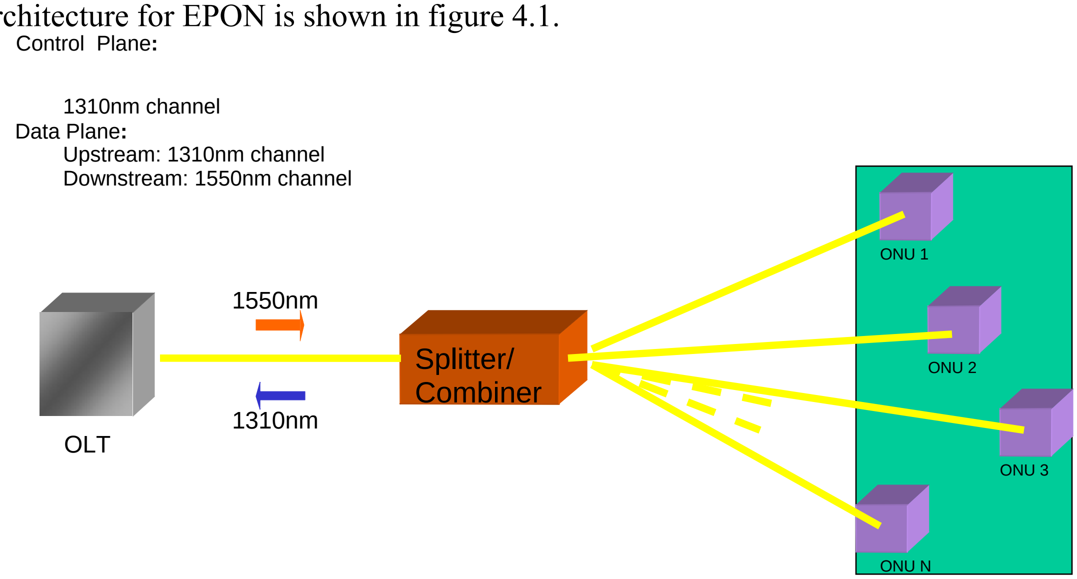

Figure 4.1: EPON architecture

An EPON is a point-to-multipoint fiber optical network with no active elements in the signal's path. It consists of a single, shared optical fiber connecting a service provider's central office (head end) to a passive star coupler (SC)/optical splitter/combiner, which is located near residential customers. The SC is intentionally positioned a substantial distance away from the central office (CO), but close enough to the customers in order to save fiber. Each customer receives a dedicated short optical fiber but shares the long distribution trunk fiber. All transmissions in a PON are performed between an Optical Line Terminal (OLT) and Optical Network Units (ONUs). Traffic from an OLT to an ONU is called 'downstream' (point-to-multipoint), and traffic from an ONU to the OLT is called 'upstream' (multipoint-to-point)

Two wavelengths are used: typically 1310 nm (λup) for the upstream transmission and 1550 nm (λd) for the downstream transmission. The OLT resides in the central office, connecting the optical access network to the metro or backbone network, where the ONU is located at either the curb (Fiber To The Curb; FTTC solution) or the end-user location (Fiber To The Building and Fiber To The Home; FTTB and FTTH). A single PON typically serves from 16-64 customers. PONs can be deployed in a 1:N tree, tree-and-branch, ring, or bus topology.

In the downstream direction, an EPON operates as a broadcast and select network. The OLT has the entire bandwidth of the channel to broadcast standard formatted 802.3 Ethernet frames to all ONUs. Each ONU extracts those packets that contain the ONU's unique Media Access Control (MAC) address. In the upstream direction, multiple ONUs share the transmission channel. Thus, the ONUs need to employ some arbitration mechanism to avoid collisions. In that case, each ONU transmits within a dedicated time slot and the OLT receives a continuous stream of collision-free frames from multiple ONUs.

4.1.2. The Proposed Hybrid Wired/Wireless Access Architecture

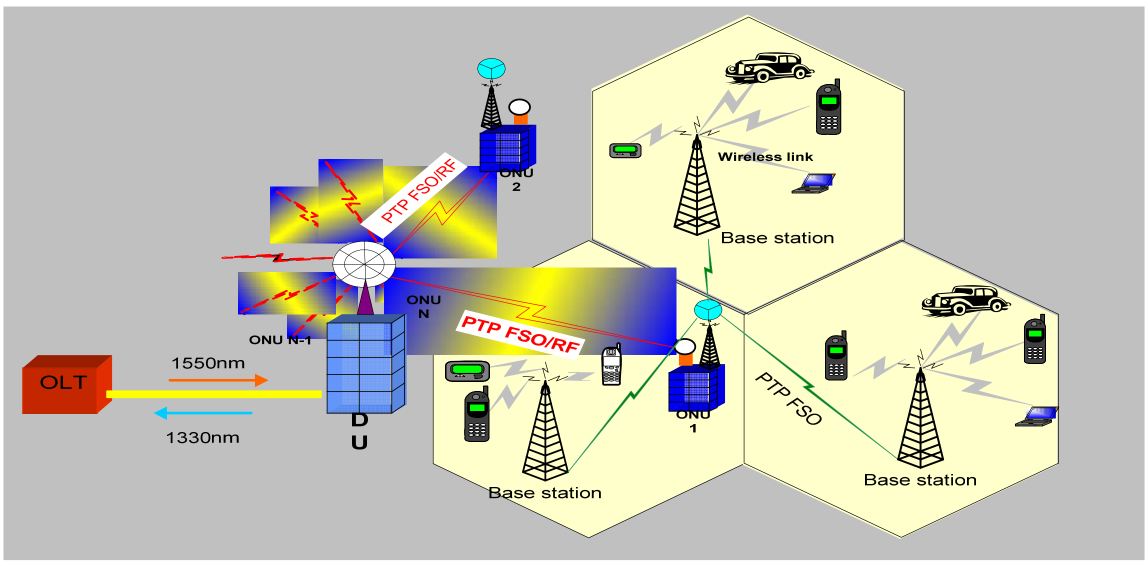

Figure 4.2: Proposed Hybrid FSO/RF Architecture

Fig. 4.2 illustrates the proposed architecture. An OLT is connected to N number of Optical Network Units (ONUs) via a 20-km trunk feeder fiber, an active Distribution Unit(DU), and a hybrid FSO/RF (HFR) wireless links. The average distance form a given ONU to the active DU at the end of the trunk is assumed to a 1-km[10-12].

Fig. 4.3 shows detailed DU architecture. In the downstream direction, the architecture serves as broadcast. All the traffic from OLT goes to an active Distribution

Unit (DU). DU has cell tower, which is divided into different sectors. The degree of each sector is measured by the number of ONUs. Each sector holds hybrid RF/FSO transceiver set to communicate between DU and ONU.

After receiving traffic from OLT, the CWDM at DU will fed it in to 1X8 splitter (figure 4.3). The splitter will split the downstream signal into 8 channels. Each of these channels is again spitted into 8 channels by using 1X8 splitter and this process can be repeated to support the desired number of ONUs (Usually 32-64 [10-12]). If we use more ONUs the signal after splitting so many times can becomes weak but we can amplify the signal by using EDFA and then use transceivers to transmit the signals.

After splitting, each signal is fed into O/E converter. After O/E the signal is fed into processor/MUX. The function of processor/MUX in downstream direction is to get signal from O/E and then transmit it to ONU by using FSO and RF transceivers. The traffic from basement to the rooftop and vice versa at DU will be sent by using FSO/RF coupler [14] and vice versa.

Each ONU has also has hybrid RF/FSO transceiver. After receiving signals from both FSO and RF paths, the signals are fed into processor/MUX situated at each ONU location. Where both signals are compared by using cyclic redundancy check (CRC) bit inserted in each frame and after the comparison whichever frame comply with CRC, that frame is passed to the users and the corrupted one is dropped. By using two paths, one RF and other FSO, transmitting same data . we have obtained two main advantages. First when the path conditions are such that one topology is starting to fail and the other is taking over, The user does not see the loss of even single bit. even if the path is rapidly switching back and forth due to changing weather conditions or other obstruction on the either path. A good example of regular occurring obstruction is the periodic interruption by bird fly through the links. This seems minor, yet the data lost on high speed connection can be significant and unacceptable for real time communications such as voice and video. Bird interruptions occur on all the system regardless of the size of aperture or number of beams deployed. The second advantage is if the equipment of either link fail still the other will be available.

After receiving the correct signal, each ONU will separate all wired and wireless traffic by looking at destination (MAC) addresses. The wired traffic is brought to the basement by using FSO/RF coupler[14] and then send to users by wired connection at ONUs (DSL, cable, coaxial cable etc). The wireless traffic is broadcasted with point-to-point HFR to all base stations in the service area of that specific ONU by using sectored transceivers. The base stations use conventional RF signals to provide traffic to the mobile users. Because signals are broadcasted, there will not be need of any handover if the users move from one sector to another sector covered by the same ONU.

In the upstream direction, the architecture serves as multipoint to point. All traffic from wireless and wired units is multiplexed at ONUs by using processor/MUX and sent to DU by using HFR transceivers situated at the rooftop on each DU site.

After receiving the signals from both FSO and RF paths (figure 4.3), DU unit will feed them in to processor/MUX. The processor compares them by using cyclic redundancy check (CRC) bit inserted in each frame and after the comparison whichever frame comply with CRC, that frame is passed to the designated buffer of the ONU and the corrupted one is dropped. The comparison is done frame by frame basis. Since the frames are pipelined there is absolutely no bit loss or Delay introduced in the system. All framing and bit stuffing is removed prior to the data stream being dropped to the user. So as far as user is concerned, the system is transparent.

After that the data at the buffers are multiplexed by using a multiplexer and multiplexed signal is converted to optical signal by driving DFB laser and then sent to OLT through Coarse Wave Length Division Multiplexer (CWDM). The function of CWDM in upstream direction is get signal from DFB laser and feed to upstream direction. In the downstream direction it takes signal from fiber and feed it to splitter.

Because all the traffic from different buffers of all ONUs at DU is sharing the medium between DU and OLT some kind of scheduling algorithm is needed to give fair share to every ONU.

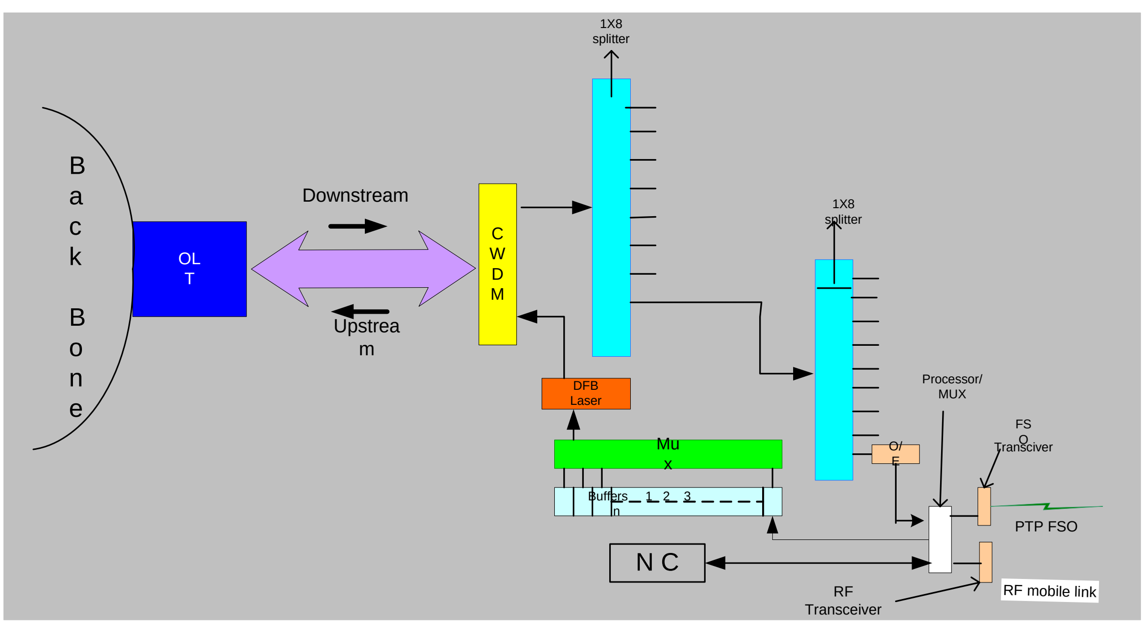

Figure 4.3. Transceiver architecture at DU

4.1.3. Performance Evaluation

In this section, we assess the feasibility, cost and the performance of the proposed architecture with that of a typical EPON architecture using the IPACT and limited DBA scheme [10,12]. We need some algorithm scheme to provide share for each user(ONU), because like EPON data from all the ONUs has to share single optical fiber link of 1 Gbps between DU and OLT. We choose DBA because it is dynamic and very efficient.[10,12]. In simulation an event-driven packet-based simulation model was developed using C++. Two simulation programs with identical network parameters were developed, one for typical EPON using IPACT architecture and the other for the proposed architecture.

To compare the upstream channel performance results of the proposed architecture with that of a typical EPON using IPACT [10], we use the same system parameters used therein; a system with 16 ONUs, access link data rate from users to an ONU of 100 Mb/s, and a 1 Gb/s upstream EPON line rate (from an ONU to the OLT). Maximum cycle time TMAX = 2ms. The guard time, TG, separating two consecutive transmission windows for typical EPON architecture is set to 5 μ s [10]. Whereas for the proposed architecture, we set TG = 0, since guard time slots are not needed. Buffer size in each ONU is 10 Mbytes [10,12].

The traffic model used here is the same as that reported in [10] where each ONU has a number of ON/OFF sources, each with a Pareto distribution governing the lengths of the ON/OFF periods, in order to capture the self-similar nature of Ethernet traffic. All arriving frames are then queued in a first-in-first-out buffer. Each point on the following plots corresponds to a sample of 50 million packets, averaged over four different runs.

Figure 4.4 - 4.6 compares the upstream average packet delay, mean packet drop and channel utilization of the proposed architecture versus that of a typical EPON using IPACT. As can be seen from simulation results, the performance of the proposed architecture outperforms that of the typical EPON architecture in terms of both upstream channel utilization and packet delay. Utilizing an active DU in the proposed architecture (instead of the SC in a typical EPON architecture) ensures:

a) Collision-free upstream data transmission without resorting to the typical use of guard time bands.

b) Eliminates the typical EPON architecture need for both control messages (REPORT message and GATE message).

The bandwidth wasted by these messages become a big problem especially in the following two cases [12].

Case1. High downstream load and light upstream load:

Light upstream traffic and traffic, each ONU will reply its Request message with no data or very little data.

This mean ONU will poll the next ONU much sooner that means Grants will be generated more often. This may result lot of bandwidth loss because of Grant messages in the downstream direction where load is heavy already.

Case2. Blocking the Grant behind a long data packet:

If the OLT determines that the next Grant message should leave at time t, but the channel is busy transmitting a long data packet ,the Grant messages will be delayed ,so will be delayed the transmission from the corresponding ONU and thus more bandwidth will be lost.

This saving of both guard time's and control massages overhead increases the available upstream transmission bandwidth and minimizes queuing delay.

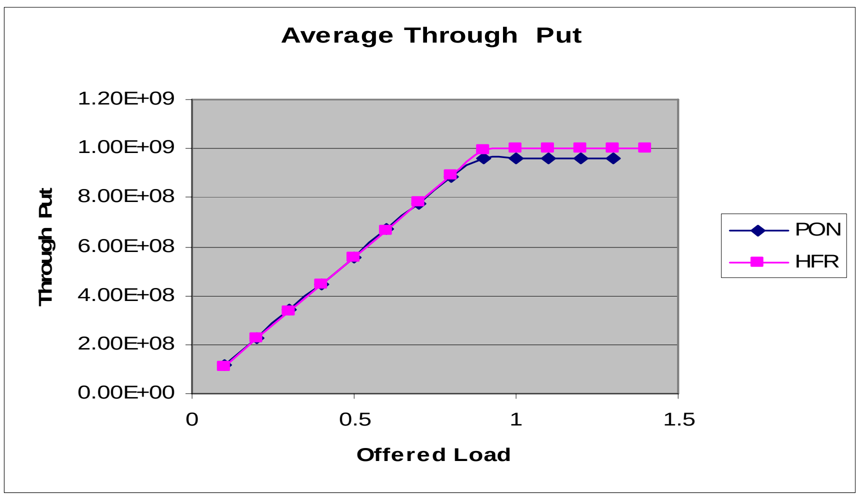

Figure 4.4. Average through put for HFR and EPON architectures

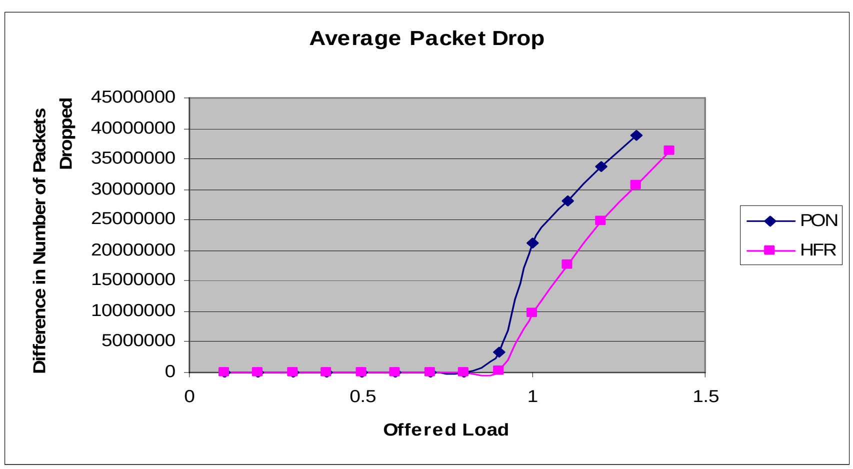

Figure 4.5. Average packet drops for HFR and EPON architectures

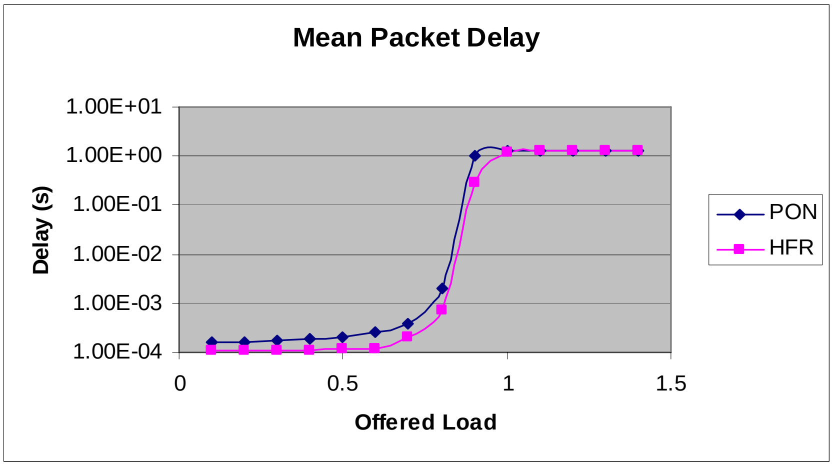

Figure 4.6. Average packet delay for HFR and EPON architectures

4.2. Conclusion

There is tremendous bandwidth requirements for broadband access network. FTTH/FTTC/FTTB/FTTN meet the requirement but the implementation is not cost effective, bridges, highways, rivers and also right of way problems.

We have proposed a novel Ethernet-based wired/wireless broadband access networking architecture that utilizes the existing wired trunk feeder fiber of typical EPON infrastructure along with a hybrid FSO/RF reliable wireless connectivity to the end-users (ONUs). By combining the benefits of both FSO and RF technologies, the proposed integrated networking solution can provide a downstream bandwidth of up to 2.5 Gbps per wavelength and 99.999% availability at a range of 1 km in all weather conditions fewer packet drops, less delay and better through put. The architecture is designed in such a way that the maximum distance covered by HFR link will be 1 km or less.

The proposed architecture can provide cost effective broadband access network solution that no other existing technology can compete.

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Chapter 5- Fair queuing algorithms to Deliver Differentiated QoS

Chapter 5

5.1. Fair Queuing Algorithms to Deliver Differentiated Quality of Service

5.1.1. Introduction

In the proposed architecture to achieve fair queuing across different independent network flows in upstream direction, a fairness algorithm need to be implemented at the Distribution Unit (DU). A number of algorithms have been proposed to achieve fair queuing across different independent network flows in the literature, e.g., Deficit Round Robin (DRR) [1], Weighted Fair Queuing (WFQ) [2], and Fair Queuing with Round Robin [3]. Considering the fact that DRR is an important fair queuing algorithm that is used in CISCO's Gigabit Switch Router 12016, we are going to implement DRR for fair queuing algorithm at the OLT. QoS is implemented in down stream direction by Optical Line Terminal (OLT) and in upstream direction by DU.

5.1.2. Deficit Round Robin Algorithm (DRR)

DRR algorithm works in rounds. A round is one complete cycle through all backlogged buffers of sending assigned amount of bytes per buffer. Note that the time duration of a round (cycle) depends on the number of backlogged buffers. Each buffer is allowed to send out packets in the first round no more than Quantum. Quantum is a fixed amount of bits. If there are no-more packets in queue "i" after the queue has been serviced a state variable called Deficiti is reset to 0. Otherwise the remaining amount is stored in the state variable Deficiti. In subsequent rounds, the amount of bandwidth usable by this queue is the sum of Deficiti of the previous round and Quantum. To avoid examining empty queues, an auxiliary list called ActiveList which is a list of indices of queues that contain at least one packet is used. For detailed explanation of DRR readers are referred to the [1].

ITU G.114 recommends that the maximum tolerable delay of voice traffic in an access system be 1.5 ms. To implement DRR with any traffic, because the low packet processing overhead of DRR (O (1)) is achieved when Quantum is bigger than or equal to maximum packet size, P_{max}[1], the lower limit of Quantum, Q_{min}, is set to P_{max} bytes. Upper limit of Quantum, Q_{max}, is bounded with maximum tolerable delay of voice traffic. Maximum buffer slot, S_{MAX}, which any buffer, Ni, at the OLT can transmit maximum number of bytes at its turn is defined as;

$S_{MAX} = Q_{max} + P_{max -1}$, $i \in 1 \ldots N$, Where $P_{max -1}$: The maximum deficit that can be left from previous round (cycle) for any buffer.

Maximum round slot, S^{T}_{MAX}, which is one iteration of processing each buffers which are utilizing their maximum assigned slots fully can be found with the following formula;

$$S^{T}_{MAX} = \sum_{i=1}^{N} N_i * \mathrm{S_{MAX}},$$

where N is number of buffers at a given DU

After setting the upper and lower limit of Quantum, we decided to use Quantum value of 2389 bytes for DRR algorithm so that the maximum scheduling delay of a packet at a buffer is equal to ~500 microseconds as in[6]. We call the connection from an SP to a user as a flow. We will consider that an OLT is serving to 16 ONUs. So each OLT has 16 buffers and each buffer can allocate packets up to 1MByte. Each buffer is composed of 3 different sub-queues to realize QoS to different CoS packets.

In this architecture, each subscriber's traffic in the downstream is directed into three priority queues in a given DU: Q_0 (CBR), Q_1 (VBR) and Q_2 (Best Effort data, (BE)). These queues are going to deliver voice, video, and data traffic respectively. The traffic used for Q_1 and Q_2, is self similar traffic with long range dependence with variable packet size ranging from 64 byte to 1518 bytes, while for Q_0, we emulate T1 connection which is 4.48 Mbps [7]. In our traffic model, downstream traffic (loads) is generated as follows; for 0.1 load (100 Mbps), 100 Mbps is divided between 16 buffers those are residing at the DU to find each buffer share (buffer load) for traffic generation. Since every buffer receives constantly 4.48 Mbps of Q0 traffic, after subtracting this value from each buffer load, remaining load is split equally between Q1 and Q2 traffic. These steps are repeated for every load generation by keeping constant value for Q0 at every load and splitting the remaining load generation equally between Q1 and Q2.

Description	Value
Quantum	2389 Byte
Number of priority classes at each buffer	3
Number of buffers at the DU(Each buffer composed of 3 queues)	16
Minimum advance time for scheduler (Inter Frame Gap period)	96 ns

Table 5.1: Parameters used in the simulation

5.1.3. Dynamic Weighted Queuing with Threshold (DWQ-T)

When scheduling the packets from the buffer (of sub-queues) which can deliver BufferWindow amount of packets at one round, scheduler first schedules all voice packets that are backlogged at the voice sub-queue. Then the remaining window from BufferWindow, if any, is going to be split between the video and data sub-queues based on the weights and the accumulation of the total packet size at the video and data sub-queues. After scheduling the packets at a given buffer, if any sub-queue of the buffer can not use some of its share and there are still packets waiting in the any sub-queue of the buffer then it will be added to deficit to be used in the next cycle with the same buffer. The algorithm for splitting the remaining window between video and data sub queues is given below

In DWQ-T algorithm the weight of the video sub-queue, θ, and the data sub-queue, β, are the values between 1 and 0 and their sum adds up to 1(θ+β =1). Total size of the packets that are accumulated at the video sub-queue is going to be multiplied by the video weight, θ, and the resultant value is going to be called weightedVideo and the packets that are accumulated at the data sub-queue is going to be multiplied by the data weight, β, and the resultant value is going to be called weightedData. The remaining BufferWindow is going to be split between Video sub-queue and data sub-queue based on the ratio of the weightedVideo and weightedData. The weight of the video and data sub-queues will be dynamically changed based on the drop ratio of the Video packets at the video sub-queue. According to [8], [9], mpeg-2 video quality is unintelligible when packet loss rate exceeds 12%). In DWQ-T, for a given video sub-queue of a buffer, if drop ratio of the video packets exceed 12% then scheduler will increment not only the video sub-queue weight by some amount, Δ, while decrementing data sub-queue weight with same amount to add up weights to 1, but it also start displacing data packets from the tail of the data sub-queue, if any, to place arriving video packets into the buffer in case buffer is full for arriving video packets. As long as drop ratio of the video packets is below 12%, when buffer is full of packets, scheduler will not displace any data packets to place arriving video packets and will simply drop video packets. If buffer is full, arriving voice packets displace data packets, if any. If there is no data packets but only voice and video packets, then arriving voice packets displace video packets from the tail of video sub-queue. After incrementing video weight, if video drop pushed down to below 12%, then video weight would be decremented while data drop is incremented and also arriving video packets will not displace data packets from the tail of data queue anymore even though buffer is full. Once video weight reduced to 0.65, which is the minimum value video weight can be, it will not go below that value under any condition In our algorithm we set the initial video and data weights as 0.65 and 0.35 respectively. The increment, Δ, is set to 0.5. To give more priority to video packets we give higher weights to video packets. Initial weights might be changed based on the traffic engineering requirement or SLA between SP and its end-user. As can be seen from the algorithm that voice packets are not included in the weighted algorithm. Simply, they are assigned the required window to deliver all voice packets that are waiting in the voice sub-queue.

5.1.4. DRR integration with Strict Priority Queuing (SPQ)

Second algorithm we use as QoS scheduler is Strict Priority Queuing (SPQ) algorithm (defined in P802.1D). In SPQ, arriving voice packets displaces first data packets from the tail of data queue if buffer is full without any condition. In case buffer is full while there exist no data packets in the buffer then arriving voice packets displace video packets, if any. Voice packets will be dropped only if buffer is full of voice packets and there is no place for arriving voice packets. When scheduling an active buffer which can deliver BufferWindow amount of packets, scheduler first delivers all the voice packets from the voice sub-queue. Then the remaining window from BufferWindow, if any, is going to be used for scheduling of all video sub-queue packets and if there is some window left from BufferWindow, it will be used for data sub-queue traffic. After servicing all the sub-queues of a given buffer, if there is still some window left to be used, but all the buffer packets are scheduled, then deficit will be reset. But if there is still a packet waiting at the head of a sub-queue which has the size more than the window left to be used then that window will be used at the next round as deficit. The biggest problem with SPQ algorithm that data packets can wait forever at the data sub-queue at the high load just because BufferWindow is used with all voice and video packets and nothing will be left for the data sub-queue.

5.1.5. Fair Non-Strict Priority Scheduling (F-NSPQ)

Algorithm starts transmitting all voice packets first then video packets and then data packets. After servicing a buffer, algorithm keeps record of the total number of packets that are waiting at each sub-queue of the buffer. After scheduler completes its round and come back to schedule the packets at the same buffer in next round, it checks its record to find out the packets from the each sub-queue which are waiting from the previous round and transmit them first and subtract total value of all these packets from the BufferWindow. After subtracting it, if there is still some window left from the BufferWindow that window is going to be used for the packets which came between first and second round scheduling time of the same buffer. Assume a buffer has in the record W1vo amount of voice, W1vi amount of video and W1dt amount of data and W2vo, W2vi and W2dt amount of packets came between two scheduling time of the buffer. Assume scheduler has enough BufferWindow to schedule all these packets. The order will be W1vo+ W2vo, W1vi+ W2vi and W1dt+ W2dt. So every time voice has the highest priority while data is the lowest. But if there is not enough windows to transmit W2vo, W2vi and W2dt, then only W1vo, W1vi and W1dt will be transmitted.

5.2. Performance evaluation

Results in Figure 5.1 shows that average buffer delay for voice traffic for DWQ-T, SPQ and F-NSPQ algorithms. To find the total delay between OLT and DU, one must take into consideration the propagation delay as well. Maximum distance of 20 km assumed between OLT and DU which has propagation delay of ~100 µs. Even under heavy load conditions figure 3 proves that average voice delay is below the maximum tolerable delay requirement (1.5 ms) of voice packets at the access networks for all 3 algorithms.

Since voice packets which come between 2 consecutive rounds time might wait next round to be transmitted at F-NSPQ, voice delay is the higher at F-NSPQ at heavy loads.

Figure 5.2 shows average buffer delay for video traffic for DWQ-T, SPQ and F-NSPQ algorithms. As predictable, higher video delay is encountered at DWQ-T algorithm due to till 12% video packets drop reached; DWQ-T neither increment the weight of video nor arriving video packets displace data packets to be placed to into the video sub-queue when buffer is full. So the treatment of video and data is same except the weights which are fixed till 12% video drops.

Figure 5.3 shows average buffer delay of data traffic for DWQ-T, SPQ and F-NSPQ algorithms. As expected data delay is the lowest at DWQ-T algorithm due to till

12% video packets drop reached; DWQ-T does not displace data packets from data sub queue even buffer is full.

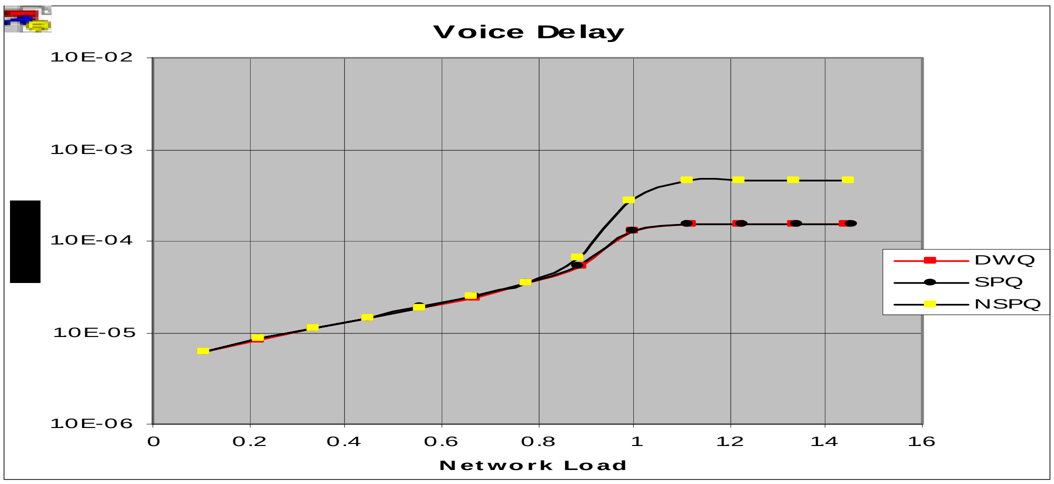

Figure. 5.1 Mean Buffer Delay of Voice packets of DWQ-T, SPQ and F-NSPQ as a function of ONL Video

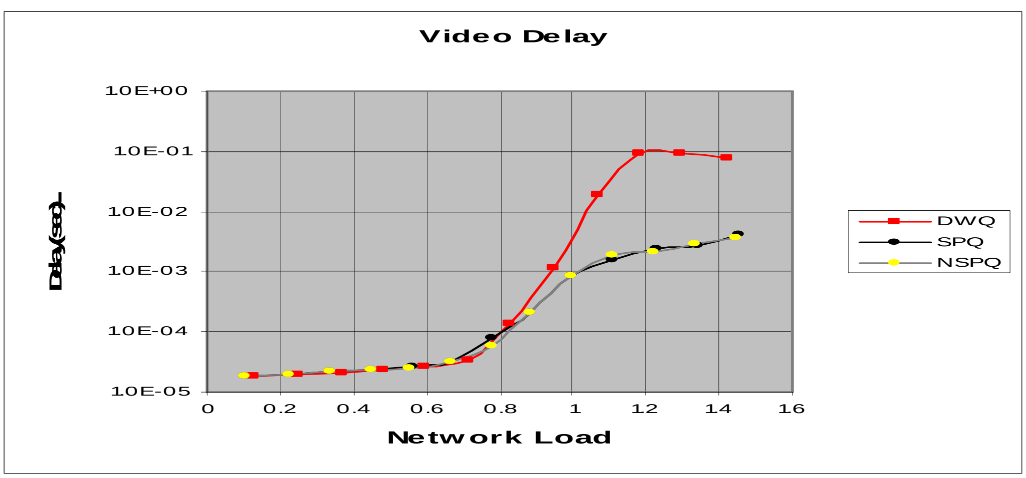

Figure. 5.2 Mean Buffer Delay of Video packets of DWQ-T, SPQ and F-NSPQ as a function of ONL

So the treatment of video and data is based on their fixed weight until 12% video drops. F-NSPQ also performs better than SPQ in data delay, because even though higher priority packets comes between two consecutive rounds, at the scheduling time of former round of a given buffer all non-serviced available packets at the time of former scheduling will be transmitted first at next round even though new voice or video packets come before the scheduling of next round.

Results in Figure 5.3 show video drop ratio for DWQ-T, SPQ and F-NSPQ as a function of ONL. Video drop ratio is the ratio between the arriving video packets and dropped video packets at buffer. When the buffer is full at DWQ-T, till drop ratio reaches 12% of all video packets, video drop increases linearly but once it reaches to 12%, it stays there because scheduler not only starts displacing data packets to place arriving video packets but also it increases the video weight while decreasing data weight.

Figure 5.5 shows data drop ratio for DWQ-T, SPQ and F-NSPQ as a function of ONL. As expected drop ratio for data packets less in DWQ-T as compared to SPQ and F-NSPQ. Since voice packets are not dropped under any load condition, it is not included in the graph.

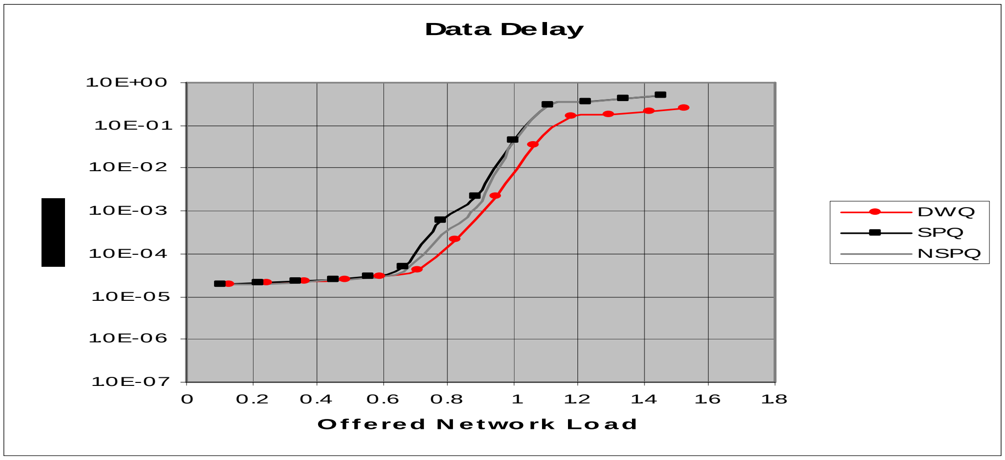

Fig.5.3 Mean Buffer Delay of Data packets of DWQ-T, SPQ and F-NSPQ as a function of ONL

Video Drop Ratio

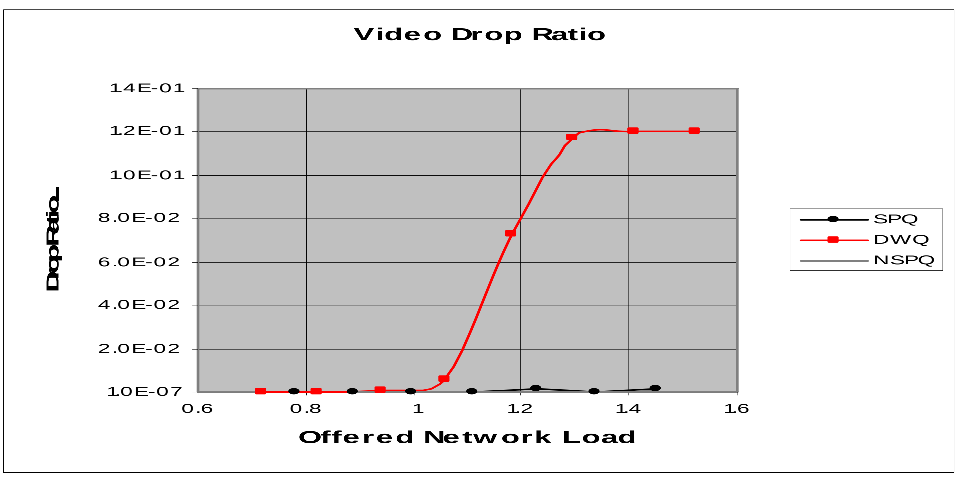

Fig. 5.4. Video Drop Ratio for DWQ-T, SPQ and F-NSPQ as a function of ONL

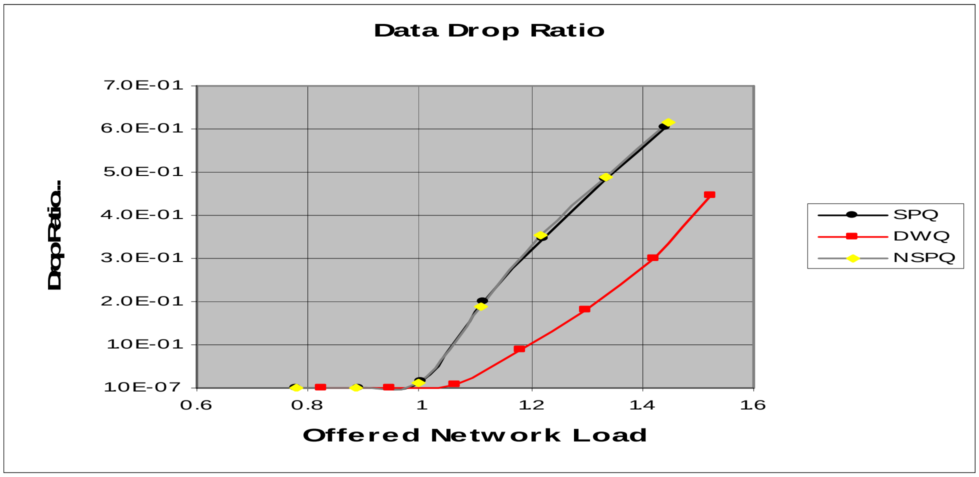

Fig. 5.5.Data Drop Ratio for DWQ-T, SPQ and F-NSPQ as a function of ONL

5.3. Conclusion

We have proposed a novel Ethernet-based wired/wireless broadband access networking architecture that utilizes the existing wired trunk feeder fiber of typical PON infrastructure along with HFR wireless connectivity to the end-users (ONUs).By combining the benefits of both FSO and RF technologies we proposed a novel Ethernet-based broadband access networking architecture that utilizes the existing wired trunk feeder fiber of typical PON infrastructure (from OLT to SC) along with a reliable HFR wireless connectivity between SC and ONUs. We have developed two algorithms (DWQ-T and F-NSPQ) and compared it to standard SPQ algorithm by integrating all of them to DRR for fairness among the customers of SP. We assumed all the end-users have the same SLA with the SP. We emulate the algorithm by simulations and observe improvement in the delays and better performance by our proposed algorithm

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Chapter-6 Fair queuing algorithm to provide QoS with wireless compensation

Chapter 6

6.1. Fair Queuing Algorithms to Provide Differentiated Quality of Service with wireless compensation for wireless Link errors

6.1.1. Introduction

Recent advances in optical networking technologies have fueled tremendous growth in both backbone and metropolitan access network (MAN) capacity. At the same time, end users are becoming more sophisticated and rich multimedia and real-time services are becoming more popular. The conduits linking the high speed end- user equipment to the high capacity backbone networks however remain a bottleneck. These connections are commonly referred to as “last/first mile”. We proposed hybrid FSO/RF architecture [15,16,17]. This paper is extension to provide QoS bounds to all Ethernet classes as well as compensation for wireless link.

Works has been done to support certain bandwidth and delay requirements over bandwidth limited link [6-8] .In the wire-line domain a large amount of work has been done with packet scheduling schemes. We have proposed hybrid FSO/RF architecture [15] and some algorithms [16, 17] to provide QoS in wired part of the architecture. In this paper we modified the algorithms [16, 17] to provide end to end QoS with wireless compensation for wireless link.

The wireless links are said to be in error when the users can not communicate with the BS (Base Station). Due to the inherent problems with wireless channels , there can be certain times allocated to a host when that host is unable to transmit data. These channel errors can occur from multi-path fading, shadow fading, or interference from another device.

It is unnecessary to redefine QoS requirements for wireless link when these have been well defined and supported in Ethernet wired environment. In order to maintain end to end QoS , QoS in Ethernet is mapped into wireless MAC together with wireless compensation.

The contribution of this paper is the development and analysis of scheduling algorithm to provide QoS bounds to all Ethernet traffic classes as defined in [1-4] .The system allows wireless media to use traffic with current Ethernet QoS specifications .This work builds on Weighed Priority Scheduling(MWPS) [9], with modifications to support Ethernet QoS parameters for wired and wireless links. The algorithm provides the means to transform each Ethernet traffic class into necessary parameters for the wireless channel scheduling .Each connection QoS is guaranteed by calculating bounds on delay and reserving the Bandwidth (BW).

6.2. Weighted Priority Scheduling (MWPS) Algorithm for Wired and Wireless Domain

Weighted priority queuing allows a traffic scheduler to allocate network resources according to the required bandwidth proportions while providing fair access to all nodes within the necessary delay bounds.

In the rest of this paper we will use the notations specified in table 6.1

Abbreviation	**Description**
Ft	*Set of all the flows*
B(t)	*Set of all the backlogged flows*
B(CBR)	*Set of backlogged flows in CBR*
B(VBR)	*Set of backlogged flows in VBR*
B(BE)	*Set of backlogged flows in BE*
Pi(t_1,t_2)	*Packets for flow I between time t1 and t2*
Wi(CBR)	*Weight of flow i in CBR*
Wi(VBR)	*Weight of flow i in VBR*
Wi(BE)	*Weight of flow i in BE*
Wt	*Weight of total flows*
Npi(CBR)	*Packets served in CBR in bytes*

Npi(VBR)	*Packets served in VBR in bytes*
Npi(BE)	*Packets served in BE in bytes*
Rc	*Rate of channel*
Wsi(CBR)	*Window size served in CBR*
Wsi(VBR)	*Window size served in VBR*
Wsi(BE)	*Window size served in BE*
D_{CBR}	*Delay bound for HOL packet in CBR*
D_{VBR}	*Delay bound for HOL packet in VBR*
D_{BE}	*Delay bound for HOL packet in BE*

Table 6.1: MWPS Notations

6.3. Channel Model Setting

We assume a single channel for uplink and downlink both for signaling and user traffic. Thus at most one packet transmission at a time between BS and user. Each packet transmission involves a RTS-CTS hand shake between the mobile user and BS.

We assume a medium for possibly imperfect prediction of the channel state. This is reasonable because the channel errors are highly correlated between successive slots. Errors in the wireless channel typically occur over a short bursts and are highly correlated in successive slots but uncorrelated over long time windows. Errors can be well represented by a two state Markov model. No more than ei errors can occur in Ti time window.

Weighted Priority Scheduling (MWPS) algorithms is implemented by mapping the specific levels of QoS of Ethernet into individual weights for use by the wireless MAC. These weights are determined by the QoS algorithm. The Ethernet traffic is divided in to three types of classes each with different priority.

1. First priority: Voice (Tight bounds on delay)
2. Second priority: Video (Requires guaranteed BW but not tight delay bounds)
3. Third priority: Data (Not tight bound on delay or BW)

The total buffer size is shared between different classes of traffic. If the upper priority levels have buffer space available lower priority flows may use it, other wise dropped. The upper priority traffic can overwrite the buffers holding the lower priority traffic. However on average the higher priority buffers will not be full, allowing the lower priorities to buffer more data and reduce packet loss.

By scheduling flows within prioritized traffic classes, the algorithm distinguishes itself from other wireless algorithms.

6.4.The analytical model for the algorithm

In this section we set the BW allocation, calculate delay and look for channel capacity for all the three priorities for wired and wireless link

To apply these results into proposed packetized hybrid architecture we implemented Multiclass Weighted Priority Queuing (MWPQ) and the maximum cycle time choosed is 2ms[1-4].The buffer size choosed is 10 Mb.

6.4.1. MWPS Embedded at DU (For wired domain)

Using MWPS at the Distribution Unit (DU) provides QoS between the ONU and OLT. The scheduler at DU maps Ethernet QoS parameters into weights and priority levels.

Delay bounds are calculated and required Bandwidth(BW) are reserved to guarantee QoS for different priorities of traffic. A scheduling scheme specific to the priority of traffic type is used in order to maintain the QoS guarantees. We do not need to compensate for wireless because either the link is fiber or redundant between OLT and ONUs.

The total BW available at every cycle is divided between all the active available flows. If the flow is inactive it can not reclaim its share later from other users.

A vital part of MWPS algorithm is the analysis of delay bounds for the priority levels and for multiple classes themselves. Once these bounds are determined they can be used along individual connection weights for scheduling.

6.4.1.1. First priority

The first priority is Constant Bit Rate (CBR) which is voice traffic. At first priority level the traffic is scheduled according to single class fair queuing model. The first priority is sensitive for delay, the cycle time we use will be 2ms and max the CBR traffic can wait for 2 and ½ cycles.

i. Channel Capacity:

$C_{CBR} = C_T$ since the entire bandwidth is available.

C_T: Total channel capacity

C_{CBR}: Channel capacity available for CBR traffic

ii. Bandwidth bound :

$$\omega_{CBR(i)} = \frac{\omega_{i(CBR)}}{\sum_{j \in B(CBR)} \omega_j} . \omega_T$$

$\omega_{CBR(i)}$ =Available window size(in bytes) for CBR traffic type in flow i

$\omega_{i(CBR)}$ = Total back logged CBR type of traffic in flow i (in bytes)

ω_T = Total window available for CBR traffic type in the current cycle

iii. Delay bound:

Worst case a packet in a flow has to wait is time to send all its own packets plus time for all other flows to finish using their share.

$$D_{CBR(i)} = \sum_{i \in B(CBR)} \frac{N_{P_{i(CBR)}}}{r_c} + \sum_{j \in B(VBR+BE)} \frac{N_{P_{i(VBR+BE)}}}{r_c}$$

$N_{P_{i(CBR)}}$ =Number of packets (in bytes) served for the flow i in CBR

$N_{P_{i(vBR+BE)}}$ =Number of packets (in bytes) served for the flows i both in VBR and BE

$D_{CBR\ (i)}$ = Delay bound for HOL packet in CBR for flow i

6.4.1.2. Second priority

Traffic must wait for the higher level queues to completely drain before gaining channel access.

i. Channel Capacity:

This means level2 traffic has an effective throughput of

$C_2=C_T-C_1$

C_1 =Channel capacity used by all served packets of CBR type of traffic in all flows

ii. Bandwidth bound:

$$\omega_{VBR(i)} = \frac{\omega_{i(VBR)}}{\sum_{j\in B(VBR)} \omega_j}.(\omega_T - \sum_{k\in B(CBR)} W_{S_k(CBR)})$$

$\omega_{i(VBR)}$ = window size(in bytes) for VBR type of traffic in flow i

B(VBR): Backlogged CBR flows

iii. Delay bound:

The HOL packet in level 2, flow must wait until the level1 queue has drained plus all arriving level 1 packets have been serviced over the delay time.

The packet might then have to wait for all other flows in level 2 to get their share of the bandwidth before gaining access and sending the flow I packet.

$$D_{VBR(i)} = \sum_{i\in B(VBR)} \frac{N_{P\,i(VBR)}}{r_c} + \sum_{j\in B(CBR+BE)} \frac{N_{P\,i(CBR+BE)}}{r_c}$$

6.4.1.3. Third priority

The lowest priority in the system processes flows that require no guarantees on BW or delay. Packets are entered in the order of arrival. An arriving packet is places in the out going queue to be sent in FIFO order.

i. Channel Capacity:

$C_3 = C_T - C_1 - C_2$

C_3=Channel capacity available for class 3

ii. Bandwidth bound:

The bandwidth available for third priority is after the first two priorities served then remaining window is divided between third priority traffic types according to their weights.

$$\omega_{BE(i)} = \frac{\omega_{i(BE)}}{\sum_{j \in B(BE)} \omega_j} . (\omega_T - \sum_{k \in B(CBR+VBR)} W_{S_k(CBR+VBR)})$$

iii. Delay bound:

The lowest priority in the system processes flows that require no guarantees on BW or delay. Packets are entered in the order of arrival. An arriving packet is places in the out going queue to be sent in FIFO order.

$$D_{BE(i)} = \sum_{i \in B(BE)} \frac{N_{P\,i(BE)}}{r_c} + \sum_{j \in B(CBR+VBR)} \frac{N_{P\,i(CBR+VBR)}}{r_c}$$

6.4.2. MWPS embedded at BS (For Wireless Domain)

MWPS is modified for wireless. Flows are allowed to either lag or lead their schedule time, lagging flows then make up their lag by leading flows giving up their lead

The scheduling is done by BS (base station). It has knowledge of both the state of channel and which mobile station needs to transmit data to uplink direction.

Back logged flows are bounded to prevent a channel that is back logged for long period of time from exclusively grabbing the channel for a significant time upon receiving good channel.

The scheduling algorithm uses a one step prediction to determine if a mobile host will be able to transmit during its next assigned slot [12].

The protocol provides wireless compensation through a system of per flow credits and debits .These credits and debits account for lagging and leading flows.

If a flow is behind its ideal service due to wireless channel error .The flow is said to be lagging and thus has accrued credit. Conversely if a flow is ahead of its ideal service because of sending data when another flow has to wait, the flow is said to be leading and thus has accumulated debits. As scheduling progresses, flows with credits and debits are swapped to compensate for wireless channel error.

In effect, lagging flows can makeup their lag by causing leading flows to give up their lead. The BS scheduler assigns the traffic flows to the channel according to the hierarchy of priorities. The bounds on the buffer size can be determined for maximum burst size (MBS).

The MWPS parameters specified for the ideal weighted priority queuing case can now be applied to the wireless model. Since the MAC layer will use the system of credits/debits to swap flows to compensate for local errors. The delay bound for our prioritized classes can exceed the maximum delays calculated in section 3.1 because of added delay due to channel errors.

Unlike some previous algorithms maximum cycle time was not fixed but in Ethernet max cycle time is fixed so the backlogged flows can get credit that do not exceed the time cycle.

6.4.2.1. First priority

The cycle time is 2ms[1-4],as we know the delay acceptable for CBR is 5ms so to provide wireless compensation the packets can be hold in buffers for next cycle.

Here it should be noted that the use of wireless protocol does not change the time a flow is delayed due to higher level. All flows at the higher priorities must be drained prior to the lower level gaining channel access.

i. Channel Capacity :

$C_{CBR} = C_T - C_{B(CBR)e}$

$C_{B(CBR)e}$ = Backlogged CBR flows who were experiencing error in previous channel

ii. Bandwidth bound :

The backlogged flows which could not send traffic because of channel error also add up in next cycle if the channel become good.

$$\omega_{CBR(i)} = \frac{\omega_{i(CBR)}}{\sum_{j \in B(CBR)} \omega_j} . \omega_T \pm \omega_{comp(CBR)_i}$$

$\omega_{comp(CBR)_i}$ = Credit/Debit for the flow i from previous cycle

iii. Delay bound:

$$D_{CBR(i)} = \sum_{i \in B(CBR)} \frac{N_{P\,i(CBR)}}{r_c} + \sum_{j \in B(VBR+BE)} \frac{N_{P\,i(VBR+BE)}}{r_c} + \frac{N_{pb(CBR)}}{r_c}$$

$N_{pb(CBR)}$ =Number of packets served for CBR backlogged flows in the previous cycle who were experiencing channel error

6.4.2.2. Second priority

If the packet detect channel error, then its packet will be skipped in that round. It compensates errored flows by new allocated slots for a later time when it is good.

We use a simple MWPS scheme as in section 3.2.1 but with wireless channel compensation . If the packet detect channel error, then its packet will be skipped in that round. It compensates errored flows by new allocated slots for a later time when it is good.

i. Channel Capacity :

$C_{VBR} = C_{T} - C_1 - C_{B(CBR)e} - C_{B(VBR)e}$

ii. Bandwidth bound :

$$\omega_{VBR(i)} = \frac{\omega_{i(VBR)}}{\sum_{j \in B(VBR)} \omega_j} .(\omega_T - \sum_{k \in B(CBR)} W_{S_k(CBR)}) \quad \pm \omega_{comp(VBR)_i}$$

$\omega_{comp(VBR)_i}$ = Credit/Debit for the flow i from previous cycles

iii. Delay bound :

$$D_{VBR(i)} = \sum_{i \in B(VBR)} \frac{N_{P\,i(VBR)}}{r_c} + \sum_{j \in B(CBR+BE)} \frac{N_{P\,i(CBR+BE)}}{r_c} + \frac{N_{pb(CBR+VBR)}}{r_c}$$

$N_{pb(CBR+VBR)}$ = Number of packets served for CBR and VBR backlogged flows in the previous cycle who were experiencing channel error in previous cycles

6.4.2.3. Third priority

The lowest priority in the system processes flows that require no guarantees on BW or delay. Packets are entered in the order of arrival. An arriving packet is places in the out going queue to be sent .In case of the channel error the packets are held in buffer until channel is error free.

i. Channel Capacity :

$C_{BE} = C_T - C_1 - C_2 - C_{B(CBR)e} - C_{B(VBR)e} - C_{B(BE)e}$

ii. Bandwidth bound

$$\omega_{BE(i)} = \frac{\omega_{i(BE)}}{\sum_{j \in B(BE)} \omega_j} . (\omega_T - \sum_{k \in B(CBR+VBR)} W_{S_k(CBR+VBR)}) \pm \omega_{comp(BE)_i}$$

$\omega_{comp(BE)_i}$ =Credit/Debit for the flow i from previous cycles

iii. Delay bound

$$D_{BE(i)} = \sum_{i \in B(BE)} \frac{N_{P\,i(BE)}}{r_c} + \sum_{j \in B(CBR+VBR)} \frac{N_{P\,i(CBR+VBR)}}{r_c} + \frac{N_{pb(CBR+VBR+BE)}}{r_c}$$

$N_{pb(CBR+VBR+BE)}$ = Number of packets served for CBR, VBR and BE backlogged flows in the previous cycle who were experiencing channel error in the previous cycles

6.5. Conclusion

The MWPS algorithm is a novel multilevel scheduler that supports QoS over a packetized wireless channel for all types of Ethernet traffic. The scheduler maps QoS parameters specified at call setup into priorities and weights for the packetized wireless channel. Rather than redefining QoS parameters for the wireless channel, the model uses and enforces the wired Ethernet standards across the combined network.

We analyzed the delay bounds and through put .The MWPS algorithm not only provide minimum delay with highest precedence for voice traffic but also Video traffic is serviced quickly with minimum possible buffering. The algorithm is work conserving and minimum BW guarantees are satisfied.

6.6. References

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Chapter 7- UMTS High Speed Packet Access Networks Evolution

Chapter 7

7.1. UMTS High Speed Packet Access Networks Evolution: towards a new generation of broadband mobile services

7.1.1. Introduction

UMTS (Universal Mobile Telecommunications System) is a third-generation (3G) broadband, packet-based transmission of text, digitized voice, video, and multimedia at data rates up to 2 megabits per second (Mbps) [1-3]. UMTS offers a consistent set of services to mobile computer and phone users, no matter where they are located in the world. UMTS is based on the Global System for Mobile (GSM) communication standard. It is also endorsed by major standards bodies and manufacturers as the planned standard for mobile users around the world. Once UMTS is fully available, computer and phone users can be constantly attached to the Internet wherever they travel and, as they roam, will have the same set of capabilities. Users will have access through a combination of terrestrial wireless and satellite transmissions. Until UMTS is fully implemented, users can use multi-mode devices that switch to the currently available technology (such as GSM 900 and 1800) where UMTS is not yet available.

Previous cellular telephone systems were mainly circuit-switched, meaning connections were always dependent on circuit availability. A packet-switched connection uses the Internet Protocol (IP), meaning that a virtual connection is always available to any other end point in the network. UMTS also makes it possible to provide new services like alternative billing methods or calling plans. For instance, users can choose to pay-per-bit, pay-per-session, flat rate, or asymmetric bandwidth options. The higher bandwidth of UMTS also enables other new services like video conferencing or IPTV. UMTS may allow the Virtual Home Environment (VHE) to fully develop, where a roaming user can have the same services to either at home, in the office or in the field through a combination of transparent terrestrial and satellite connections.

The packet domain uses a packet-mode technique to transfer high-speed and low-speed data and signaling in an efficient manner. The packet domain optimizes the use of network and radio resources. Strict separation between the radio subsystem and network subsystem is maintained, allowing the network subsystem to be reused with other radio access technologies.

A common packet domain Core Network is used for both GSM and UMTS. This common Core Network provides packet-switched (PS) services and is designed to support several quality of service levels to allow efficient transfer of non real-time traffic (e.g., intermittent and bursty data transfers, occasional transmission of large volumes of data) and real-time traffic (e.g., voice, video). Applications based on standard data protocols and SMS are supported, and interworking is defined with IP networks. Charging should be flexible and allow to bill according to the amount of data transferred, the QoS supported, and the duration of the connection.

The Serving GPRS Support Node (SGSN) keeps track of the location of an individual MS and performs security functions and access control. The SGSN is connected to the GSM base station system through the Gb interface and/or to the UMTS Radio Access Network through the Iu interface. The SGSN also interfaces via the GPRS Service Switching Function with the GSM Service Control Function for optional CAMEL session and cost control service support.

The Gateway GPRS Support Node (GGSN) provides interworking with external packet-switched networks, and is connected with SGSNs via an IP-based packet domain PLMN backbone network.

The Charging Gateway Functionality (CGF) collects charging records from SGSNs and GGSNs.

The HLR contains GSM and UMTS subscriber information.

The SMS-GMSCs and SMS-IWMSCs support SMS transmission via the SGSN.

Optionally, the MSC/VLR can be enhanced for more-efficient co-ordination of packet-switched and circuit-switched services and functionality: e.g., combined GPRS and non-GPRS location updates.

In order to access the PS services, an MS shall first make its presence known to the network by performing a GPRS attach. This makes the MS available for SMS over PS, paging via the SGSN, and notification of incoming PS data.

In order to send and receive PS data, the MS shall activate the Packet Data Protocol context that it wants to use. This operation makes the MS known in the corresponding GGSN, and interworking with external data networks can commence.

User data is transferred transparently between the MS and the external data networks with a method known as encapsulation and tunnelling: data packets are equipped with PS-specific protocol information and transferred between the MS and the GGSN. This transparent transfer method lessens the requirement for the PLMN to interpret external data protocols, and it enables easy introduction of additional interworking protocols in the future.

7.1.2. Background and Motivation

Vast majority of mobile networks are designed and dimensioned to comply with the requirements of worst-case conditions in which busy hour or peek traffic demand can arise simultaneously in all of the backhaul network elements. To meet capacity demands, the TxRxs are assigned to BS according to busy hour call usage projections. The busier an area is, the more TxRxs need to be installed in the BSs to provide sufficient capacity, and the more each backhaul bandwidth is required to transport increased traffic. There is also a coverage demand, so the mobile service providers will add additional BSs to insure full service coverage for all areas (urban, sub-urban and rural), regardless of usage remaining low in some areas.

This will lead to situation where the traffic channels in one system are rarely simultaneously busy and the result is - inefficient allocation of limited network resources. Another issue that must be taken into account is inherent inefficiency. The service providers have no way of using extra capacity on seldom used lines to carry overflow traffic from busier lines. Consequently, while service providers can lease additional T1/E1 lines for BSs in high-traffic areas, they must absorb the cost of providing spare backhaul bandwidth for BSs in low-use areas

7.2. Main UMTS Concepts

In Iu mode, radio resources are allocated to MSs in a very flexible manner. Depending on the level of activity, MSs are allocated shared contention-based radio resources or dedicated radio resources for user packet transmission.

Three UMTS MS modes of operation are supported in Iu mode: The PS/CS mode of operation corresponds to class-A mode of operation in A/Gb mode. The PS mode of operation corresponds to class-C mode of operation in A/Gb mode. The CS mode of operation is the outside the scope of this specification.

UMTS security functionality is equivalent to or of higher functionality than the existing GSM security. UMTS may use a security algorithm different from GSM. The 3G-SGSN performs the authentication procedure, and the RNC performs the ciphering procedure based on the algorithm for UMTS.

In Iu mode, different levels of mobility procedures are executed depending upon the MS state. When an MS has an active RRC connection to UTRAN, the MS performs either UTRAN Registration Area updating procedures or handover or cell update procedures depending on the level that UTRAN is tracking the MS position. When an MS does not have an active RRC connection (i.e., it is in idle mode), the MS performs RA updating procedures. In all the procedures, the cell selection is controlled by the network by setting cell selection parameters and/or restriction information.

7.2.1. Logical Architecture

The packet domain Core Network functionality is logically implemented on two network nodes, the Serving GPRS Support Node and the Gateway GPRS Support Node. It is necessary to name a number of new interfaces. No inference should be drawn about the physical configuration on an interface from Figure 7.1

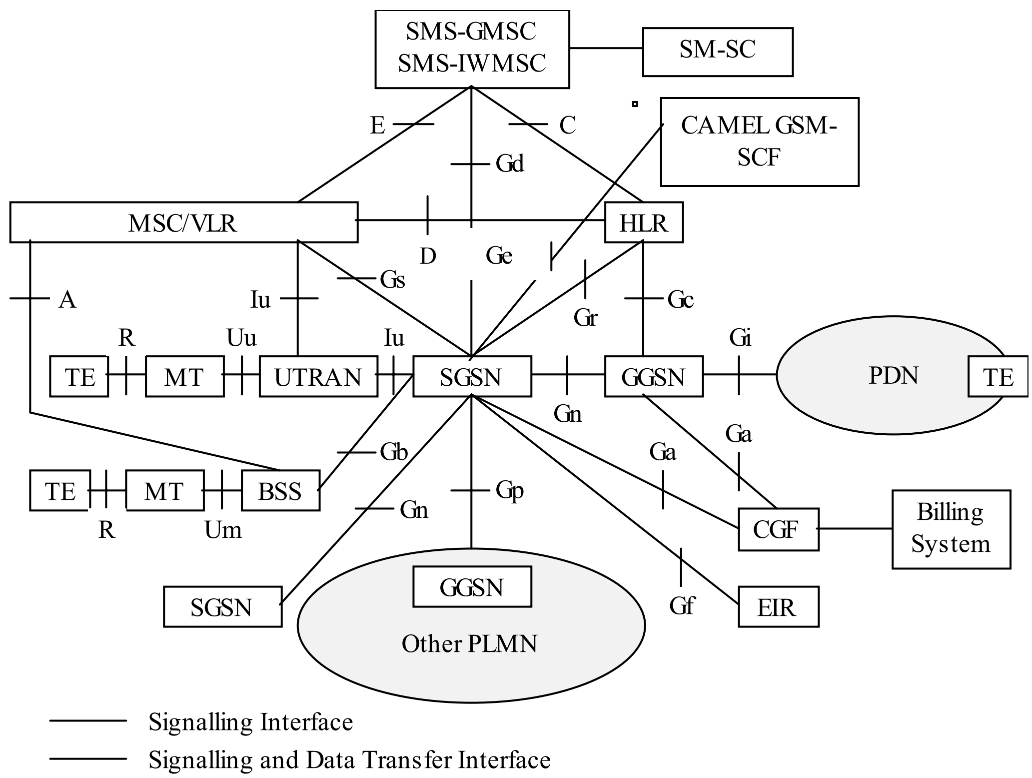

Figure 7.1: Overview of the Packet Domain Logical Architecture

7.2.2. UMTS Mobile Stations

A UMTS mobile station can operate in one of three modes of operation [1-4]. However, these operation modes are different from the ones in A/Gb mode GPRS due to the capabilities of UTRAN to multiplex CS and PS connections, due to paging co-ordination for PS services and CS services that are offered by the CN or the UTRAN, etc. The different UMTS mobile station operation modes are defined as follows:

- PS/CS mode of operation: The MS is attached to both the PS domain and CS domain, and the MS is capable of simultaneously operating PS services and CS services. This mode of operation is equivalent to the GSM GPRS class-A mode of operation.
- PS mode of operation: The MS is attached to the PS domain only and may only operate services of the PS domain. However, this does not prevent CS-like services to be offered over the PS domain (e.g., VoIP). This mode of operation is equivalent to the GSM GPRS class-C mode of operation.
- CS mode of operation: The MS is attached to the CS domain only and may only operate services of the CS domain. However, this does not prevent PS-like service to be offered over the CS domain. The CS mode of operation is outside the scope of this specification.

All combinations of different operation modes as described for GSM and UMTS MSs shall be allowed for GSM and UMTS multisystem terminals.

7.2.3. UMTS User Plane

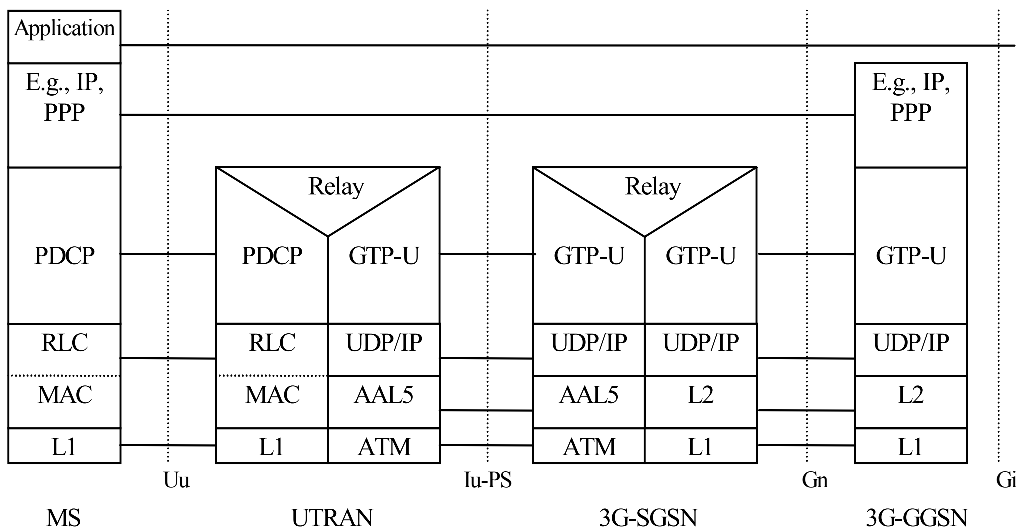

Figure 7.2: User Plane for UMTS

7.2.4. UMTS Control Plane

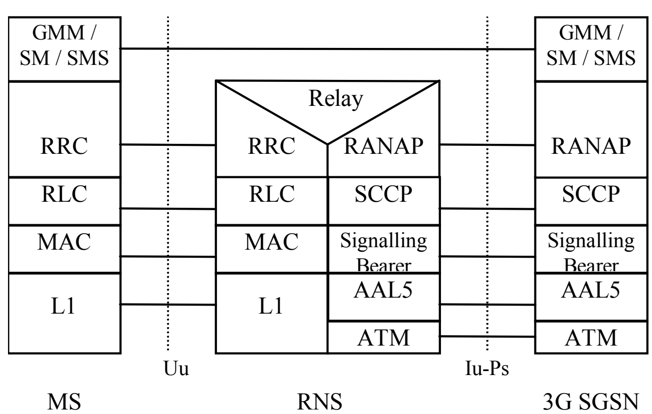

Figure 7.3: UMTS Control Plane

7.3. Current Architecture

Following figure shows current UMTS/GSM network architecture:

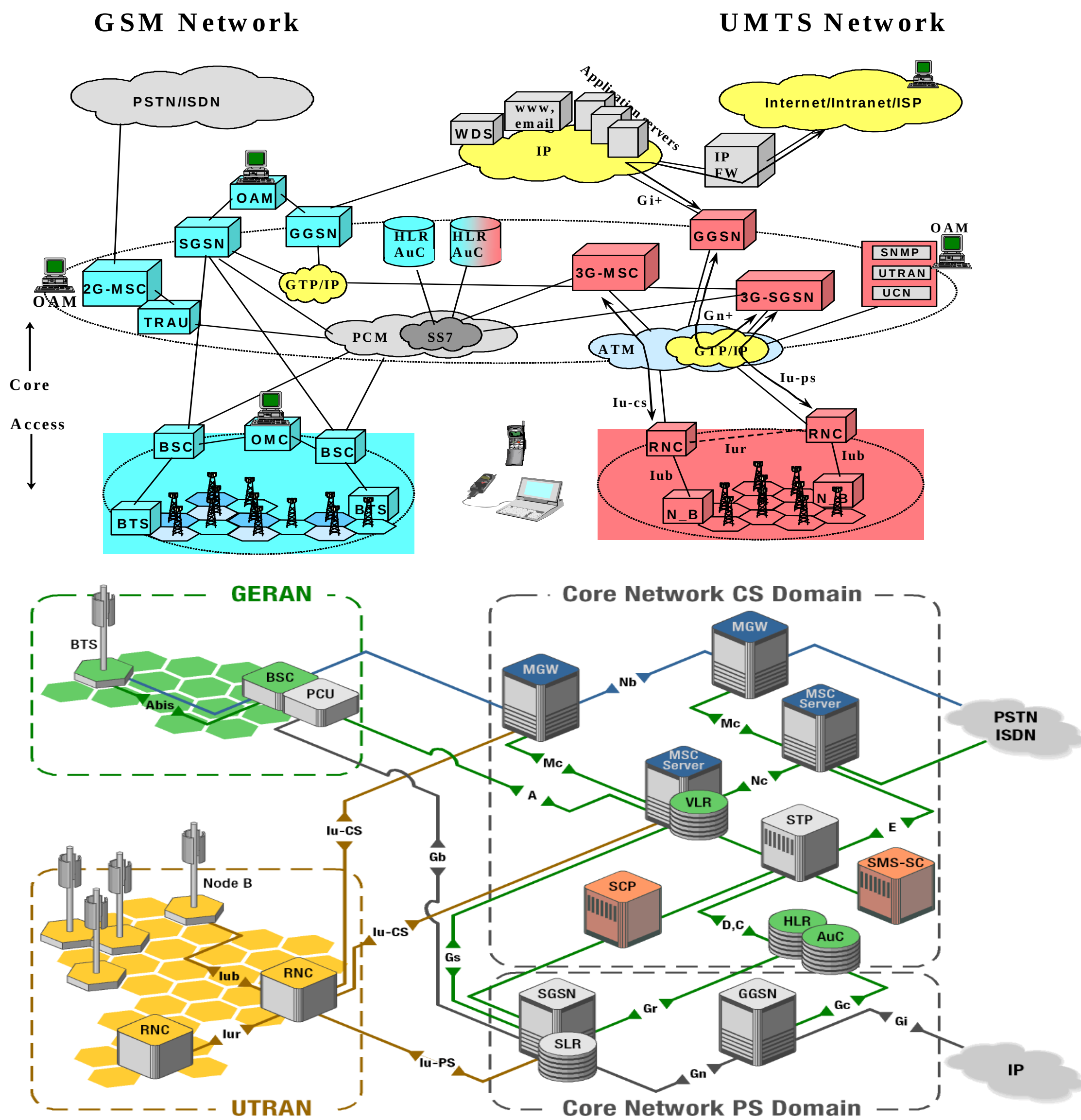

Figure 7.4: UMTS Reference Model

Node B - Base Station

RNC - Radio Network Controller

MSC/VLR - Mobile Switching Center / Visitor Location Register

GMSC - Gateway MSC

HLR - Home Location Register

SGSN - Serving GPRS (General Packet Radio Service) Support Node

GGSN - Gateway GPRS Support Node

7.3.1. UMTS Interfaces

Iu: UTRAN / CN interface specified by the RANAP (Radio Access Network Application Part) Protocol. Further separated by packet switched (Iu-ps) and circuit switched (Iu-cs)

Iur: Inter RNS interface specified by the RNSAP (RNS Application Part) protocol

Iub: RNC / Node B interface specified by the NBAP (Node B Application Part) protocol

Uu: Layer 1 and layer 2, physical and MAC layers are supported at the RNC by the RLC (Radio Link Control) protocol.

7.3.2. RNC FUNCTIONALITY

- **RNC Control Functions**

Cell setup, call admission control, call setup, handover control, Layer 3 signaling [4,5]

- **RNC OA&M Functions**

Configuration mgmt, Alarm management, Performance mgmt

- **RNC Traffic and Signaling Processing**

ATM termination, Signaling message routing, user data processing and routing, frame selection/distribution, ROLPC, ciphering

7.4. Recent advances

UMTS (Universal Mobile Telecommunications Service) is a third-generation (3G) broadband, packet-based transmission of text, digitized voice, video, and multimedia at data rates up to 2 megabits per second (Mbps). UMTS offers a consistent set of services to mobile computer and phone users, no matter where they are located in the world. UMTS is based on the Global System for Mobile (GSM) communication standard. It is also endorsed by major standards bodies and manufacturers as the planned standard for mobile users around the world. Once UMTS is fully available, computer and phone users can be constantly attached to the Internet wherever they travel and, as they roam, will have the same set of capabilities. Users will have access through a combination of terrestrial wireless and satellite transmissions. Until UMTS is fully implemented, users can use multi-mode devices that switch to the currently available technology (such as GSM 900 and 1800) where UMTS is not yet available.

Previous cellular telephone systems were mainly circuit-switched, meaning connections were always dependent on circuit availability. A packet-switched connection uses the Internet Protocol (IP), meaning that a virtual connection is always available to any other end point in the network. UMTS also makes it possible to provide new services like alternative billing methods or calling plans. For instance, users can choose to pay-per-bit, pay-per-session, flat rate, or asymmetric bandwidth options. The higher bandwidth of UMTS also enables other new services like video conferencing or IPTV.

UMTS may allow the Virtual Home Environment (VHE) to fully develop, where a roaming user can have the same services to either at home, in the office or in the field through a combination of transparent terrestrial and satellite connections.

7.5. HSPA: An evolution in UMTS to Provide Broadband Services

From the first launches of UMTS service in the Japanese Market by NTT Docomo in 2002, UMTS has shown itself to be the fastest growing cellular technology in history. As of August 2005, it is estimated that over 30 million subscribers were using UMTS services globally. Services such as video telephony, video streaming, mobile TV and mobile e-mail are now commonplace and the public awareness of the capabilities of 3G networks and terminals has helped considerably in moving the public perception of the mobile terminal from a pure voice and text communication instrument to a multimedia device. The levels of speed and interactivity offered improve utility to the end-user and significantly improve productivity and opportunities for operators, end-users and enterprises.

The challenge facing the mobile telecommunications industry today is how to continually improve the end-user experience, offer appealing services through a delivery mechanism which offers improved speed, service attractiveness and service interaction. Possibly the most important improvement is the arrival of a new series of technologies referred to as High Speed Packet Access (HSPA). These technologies will be available as a relatively straightforward upgrade to existing UMTS networks[6] and will offer improved bandwidth to the end-user, improved network capacity to the operator, and improved interactivity for data applications.

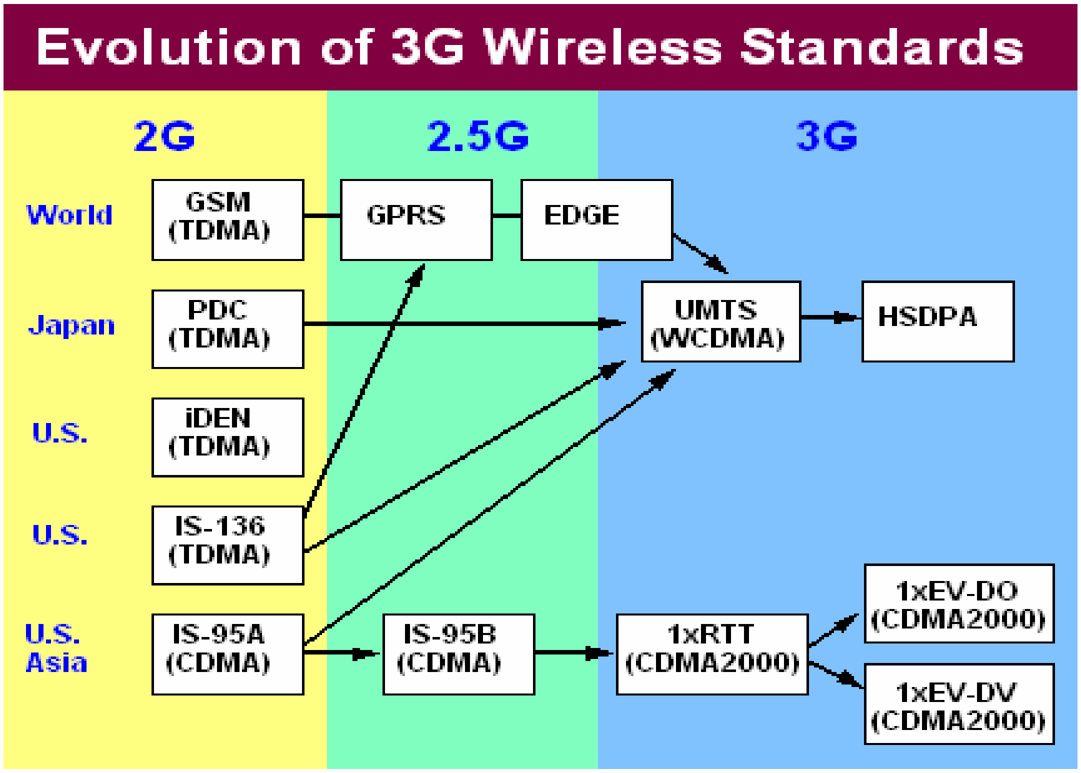

Figure 7.5. Evolution of wireless standards

In this article we explain in everyday industry terms what HSPA is, what it can do and what the benefits to all industry stakeholders will be. It is not an exhaustive technical description but an informational paper which seeks to raise industry awareness of how HSPA will affect the mobile telecommunications industry.

7.5.1. HSPA Definition

High Speed Packet Access (HSPA) is a generic term adopted by the UMTS Forum to refer to improvements in the UMTS Radio Interface in the Releases 5 and 6 of the 3rd Generation Partnership Project (3GPP) standards. HSPA refers to both the improvements made in the UMTS downlink, often referred to as High Speed Downlink Packet Access (HSDPA) and the improvements made in the uplink, often referred to as High Speed Uplink Packet Access (HSUPA) but also referred to as Enhanced Dedicated Channel (E-DCH).

HSDPA enables data transmission speeds of up to 14.4Mbit/s per user. Both HSDPA and HSUPA can be implemented in the standard 5 MHz carrier of UMTS networks and can co-exist with the first generation of UMTS networks based on the 3GPP Release 99 (R99) standard. As HSPA standards refer uniquely to the access network, there is no change required of the core network outside of the capacity increases that will be required to handle the expected increase in traffic generated by HSPA. The more important change that HSDPA makes is to move control of the medium access control (MAC) from the radio network controller (RNC) into the Node B.

7.5.2. HSPA Technology

A. HSDPA

HSDPA introduces a number of new technical capabilities to the radio access network, which when combined offer a significant improvement for both end users and operators. These capabilities are;

• A new common High Speed Downlink Shared Channel (HS-DSCH) which can be simultaneously shared by multiple users,

• The use of a shorter Transmission Time Interval (TTI) of 2ms, which enables higher speed transmission in the physical layer.

• The use of fast scheduling.

• The use of Adaptive Modulation and Coding (AMC).

• The use of fast retransmission based on fast Hybrid Automatic Response reQuest (HARQ) techniques.

The HS-DSCH is shared channel with a number of Spreading Factor 16 (SF-16) CDMA codes. Within each 2 ms TTI, a constant spreading factor of 16 is used with a maximum of 15 parallel channels in the HS-DSCH. These channels may all be assigned to one user during the TTI, or may be split amongst several HSDPA users. There is no Power Control with HSDPA and the HS-DSCH is transmitted at a constant power while the modulation, the coding and the number of codes are changed to adapt to the variations of radio conditions.

The shorter 2ms TTI (compared to TTI of between 10ms and 80ms in UMTS R99) means that the systems is more reactive to changing user or radio conditions and can quickly allocate capacity to users.

Fast data traffic scheduling means that variations arising from changing radio conditions can be accommodated and that the BTS is able to allocate as much of the particular cell's capacity to a particular user for a short period of time. This means that a user is able to receive as much data as radio conditions will allow.

Adaptive Modulation Coding (AMC) with fast link adaptation means that the modulation and coding formats can be changed in accordance with variations in the channel conditions, leading to a higher data rate for users with favorable radio conditions. Whereas UMTS Release 99 used only Quadrature Phase Shift Keying (QPSK) modulation, HSDPA provides the ability to use 16-QAM when the link is sufficiently robust, which can lead to a significant increase in data rate.

Fast H-ARQ enables erroneous packets to be resent within a 10ms window, ensuring that the TCP throughput remains high. In addition, in HSDPA the mechanisms for ARQ are moved to the BTS (from the RNC in R99). By using these approaches, all users, whether near or far from the base station, are able to receive the optimum data rate.

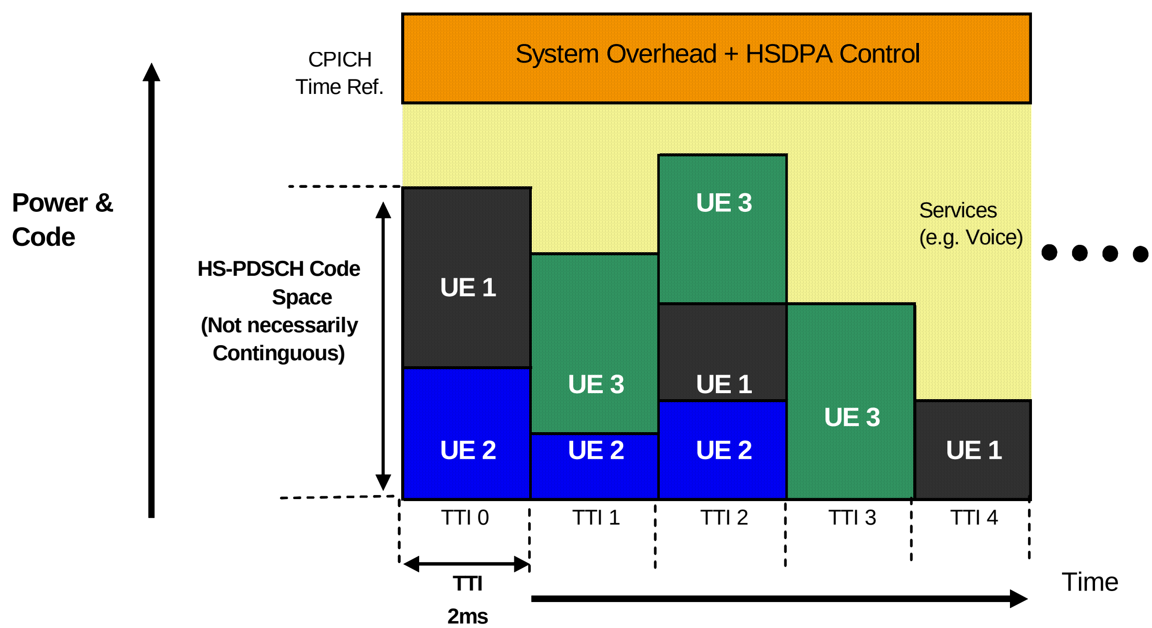

Figure 7.6 Time allocation for the UE

B. HSUPA

Similarly to HSDPA in the downlink, HSUPA defines a new radio interface for the uplink communication [10, 11]. The overall goal is to improve the coverage and throughput as well as to reduce the delay of the uplink dedicated transport channels.

From a 3GPP point of view, the first set of standards was approved in December 2004, and performance aspects were finalized during the summer of 2005. E-DCH is the name adopted in 3GPP for HSUPA which is in 3GPP Release 6.

Key technical capabilities introduced with HSUPA are;

- A new dedicated uplink channel.
- Introduction of H-ARQ.
- Fast Node B scheduling.

Unlike HSDPA, HSUPA remains based on a dedicated channel. A series of new channels are introduced for both signaling and traffic to improve overall uplink capabilities. Like HSDPA, HSUPA introduces fast retransmissions based on the Hybrid ARQ Protocol for error recovery at the physical layer [1-3].

7.5.3. HSPA Terminals

HSDPA will require new terminals. However, HSDPA and R99 capabilities will co-exist in these terminals and will be compatible with UMTS R99 networks. A typical upgrade strategy for an operator will be to upgrade the network to HSDPA progressively so that for a significant period both standards will be in operation. Therefore, all new terminals should support both R99 and HSDPA. The first terminals will be a data card enabling 1.8 Mbit/s of peak data rate (Category 12) and 3.6 Mbit/s of peak data rate (Category 6).

HS-DSCH Category	Max number of HS-DSCH codes (SF16) received	Minimum inter TTI interval	Modulation	Max peak rate
Category 1	5	3	QPSK & 16-QAM	1.2Mbps
Category 2	5	3	QPSK & 16-QAM	1.2Mbps
Category 3	5	2	QPSK & 16-QAM	1.8Mbps
Category 4	5	2	QPSK & 16-QAM	1.8Mbps
Category 5	5	1	QPSK & 16-QAM	3.6Mbps
Category 6	5	1	QPSK & 16-QAM	3.6Mbps
Category 7	10	1	QPSK & 16-QAM	7.3Mbps
Category 8	10	1	QPSK & 16-QAM	7.3Mbps
Category 9	15	1	QPSK & 16-QAM	10.2Mbps
Category 10	15	1	QPSK & 16-QAM	14.4Mbps
Category 11	5	2	QPSK only	900kbps
Category 12	5	1	QPSK only	1.8Mbps

Table 7.1: HSDPA Capability per 3GPP Category

HSUPA will also require new terminals. Similarly, they will co-exist with R99 capabilities and will be compatible with UMTS R99 networks. First HSUPA capable terminals are foreseen for 2H 2006. Like HSDPA, the capability of the terminal has been standardized per category.

E-DCH category	Maximum number of E-DCH codes transmitted	Minimum spreading factor	Support for 10 and 2 ms TTI EDCH	Maximum number of bits of an E-DCH transport block transmitted within a 10 ms E-DCH TTI	Maximum number of bits of an E-DCH transport block transmitted within a 2 ms E-DCH TTI
Category 1	1	SF4	10 ms only	7296	-
Category 2	2	SF4	10 ms and 2ms	14592	2919
Category 3	2	SF4	10 ms only	14592	-
Category 4	2	SF2	10 ms and 2ms	20000	5837
Category 5	2	SF2	10 ms only	20000	-
Category 6	4	SF2	10 ms and 2ms	20000	11520

Table 7.2: HSUPA Capability per 3GPP Category

Theoretically, the maximum physical throughput is 5.5 Mbit/s for the Uplink, which leads to 4 Mbit/s at the application layer. Recent studies have concluded that the use of HARQ with Node-B scheduling and 2ms TTI can lead the following improvements when compared with R99 systems;

• 50-70% improvement in UL capacity,

• 20-55% reduction in end-user packet call delay,

• Around 50% improvement in user packet call throughput.

7.5.4. HSPA Benefits

The key benefits of HSPA can be categorized in 3 ways;

• Improved speed for end user applications.

• Improved interactivity for end user applications.

• Improved network capacity for the operator.

UMTS networks deployed based on the 3GPP release 99 standard offer a maximum data throughput per user of 384kbit/s. With HSDPA there is the possibility to offer the end user up to 14.4Mbit/s. How often or how many users will be able to achieve this throughput will obviously depend on network and radio conditions as well as the type of terminal being used. This will bring new applications such as high quality video streaming as well as faster music and entertainment downloads, and improved time savings for ubiquitous corporate email services. Simulations show these features to further improve user data rates and network capacity. 3GPP standards beyond release 5 will aim to achieve further throughput increases, say peak data rates in the range 20 to 30 Mbit/s, by using Multiple Input Multiple Output (MIMO).No changes are required to the networks except increased capacity within the infrastructure to support the higher bandwidth.

One of the major improvements with HSPA technology is the improvement in network latency or round trip delay for data applications. First deployments of HSDPA indicate a round trip delay of as low as 60ms, meaning that many real time interactive services can be delivered over HSDPA. This will be true for Voice and Video but also for applications such as multi-user gaming where immediate real time interaction with other users is key to stimulate high levels of game usage. This will be the key enabler for the beginning of a new era of mobile multimedia over UMTS networks and terminals.

7.5.5. HSPA Applications

The new network and terminal capabilities introduced by HSPA [6-9] will drive new applications and stimulate new usage patterns. At the UMTS Forum we believe that these new applications and behaviors will fall into 3 categories;

• Introduction of new applications.

• Improvement of existing applications.

• Stimulation of new usage patterns.

II. Introducing new applications

New data applications require varying bandwidth to deliver the required end user experience. Mobile data applications can require anything from a couple of kilobits for text messaging to many hundreds of kilobits for high quality video streaming or conferencing.

As the network and terminal capabilities improve and the economics of delivering mobile broadband improve, the number of applications that can be delivered is increased as well as the number of users that can access these applications. This will provide a major stimulus to the mobile data market.

HSPA will stimulate many new applications, a large number of which have yet to be introduced or conceived. As an extrapolation of today's usage patterns, it can be expected that new applications in the following areas will present in the market.

High-Speed Internet Access

With HSPDA offering similar speeds to most DSL connections [7-9] , with the added value of ubiquitous mobility, UMTS can be expected to become to preferred connection medium for a range of users, whether it be via a laptop or a handheld terminal.

i. Voice over IP

Voice is clearly not a new application but when delivered over IP and coupled with other interactive media such as video and text, this new service mix will become very attractive.

ii. Multi-player Gaming

The improved interactivity of the networks supporting HSPA is expected to have a significant impact on the mobile gaming industry. Multi-user games, whether broadband or narrowband will benefit from the real time interactivity that will be possible and the end user experience will be significantly enhanced.

II. Improving Existing applications

HSPA will improve the end user experience for the many existing applications. The improvement in speed and interactivity will make a significant difference to many applications which previously appeared to be slow or tedious. The most often cited example of this will be the corporate email service based on Microsoft Outlook, where there is significant communication between the device and the server which in high latency networks can make the service appear slow. With the arrival of HSPA and the improvements in system latency, there will be major improvements and it is expected that the end user perception will be significantly improved.

i. Streaming Live TV

With the increased capacity of HSPA networks, more streaming services can be offered to more and more users. These TV streaming services have already shown themselves to be extremely popular in many markets and this trend can be repeated and improved on with HSPA.

ii. Video Telephony and Conferencing

Video conferencing or the delivery of multiple video streams to a single terminal will become more feasible now that the video services can be delivered using an IP stream.

iii. Driving new usage patterns

With an improved experience to the end-user for mobile data services, and an improved cost of delivery for the operator, the market is expected to demonstrate some significantly new behavioral and usage patterns.

These behaviors can be placed in a number of categories;

• New mobile business behaviors.

• New Peer to Peer application behaviors.

• New Mobile commerce behaviors.

For business users, it is expected that HSPA will drive significantly different usage patterns. In particular the usage of mobile email which is now synonymous with the Blackberry device, is today restricted to the text of an email and the attachments are discarded. This can be expected to change as networks and mobile devices become truly broadband and the storage capabilities of devices continue to increase. The use of the corporate intranet while mobile is expected to increase as HSPA enables a user experience which closer approximates to that of WLAN, while adding the extra benefit of ubiquitous mobility.

Peer to Peer applications drive the mobile industry. Whether it is voice or text, the desire for people to communicate with each other is insatiable and as HSPA rolls out and offers more users a broadband IP connection, the peer to peer possibilities will increase significantly. As well as communicating via voice and text, the end-user will use the same IP stream to communicate potentially by video, by sharing photos, by sharing presence information and by opening a gaming session. Just as text messaging and mobile email have driven the last major changes in end-user behavior patterns, peer to peer IP multimedia services offer the next major user behavior revolution. HSPA will be a key enabler of this revolution.

For mobile commerce, the increased omnipresence of mobile data will stimulate the purchasing opportunities for the end users. As i-Tunes has demonstrated, using consumer media while mobile meets our lifestyle demands and there is little reluctance to purchase this media over the internet. This offers a major opportunity to operators and content creators, as well as delivering an improved mobile experience to the end-user.

7.5.6. HSPA Availability

In March 2003, HSDPA was introduced in 3GPP Release 5 Specifications and the final and stable specifications were finally agreed in June 2004.

The first set of standards for E-DCH, or HSUPA, was approved in December 2004.

However, corrections have still to be done and the final version of 3GPP Specifications for HSUPA is expected in December 2005. The first demonstrations of HSDPA technology were made in the final months of 2004. The 3GSM conference in Cannes in February 2005 was where the first public demonstrations of the technology were made by infrastructure and terminal vendors together. By summer 2005 (the initial publishing date of this paper) commercial terminals in the form of data cards for PCs were available.

These data cards support connection speed of up to 1.8Mbit/s and will be category 12 terminals. It is expected that the first handset form factor terminals will be available during the first half of 2006. A wider variety of handsets are expected to become available at the beginning of the second half of the year. These commercial handsets are expected to be category 6. In each case network infrastructure is expected to be in place and capable of supporting these terminals and that commercial service will be launched during 2006 in many markets. The first HSUPA terminals are expected to become commercially available in the first half of 2007.

7.6. Conclusion

HSPA defines a series of straight forward upgrades to UMTS R99 networks which will offer improvements of a factor of ten in the speed of service delivery, improvements of a factor of five in network capacity and a significant improvement in service latency. HSPA refers to improvements in both the downlink and uplink of the radio access network, known as HSDPA and HSUPA respectively.

HSPA will thus offer cost effective wide-area broadband mobility and play a significant role in stimulating the demand for data services, whether they be consumer multimedia and gaming or corporate email and mobile access.

But to support the data rates proposed by HSPA ,the conventional backhaul wireless network(between the RNC and Node B) need to be replaced .Because star based connections between RNC and Node Bs are not composed or T1/E1 lines which are expensive and also laying multiple lines is not feasible solution.

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Chapter 8- PON Based UMTS Broadband Access networks

Chapter 8

8.1. A Novel PON Based UMTS Broadband Wireless Access Network Architecture With an Algorithm to Guarantee End to End QoS.

8.1.1. Introduction

Currently third generation wireless wide area networks based on UMTS are being deployed throughout the world. As of mid 2006 there were over 75 million IMT-2000/UMTS subscriptions world wide in more than 110 IMT-2000/UMTS networks launched commercially. To provide broadband packetized wireless access to end users 3GPP has proposed High Speed Packetized Access (HSPA) based on UMTS networks. HSPA splits the MAC layer between Radio Network Controller (RNC) and Node B. The standardized HSDPA can provide the data rates of 14.4 Mbps (category 6, Rel 5) in downlink direction (from Node B to mobile users) by carrying users data over High

Speed Physical Downlink Channels (HS-SCCH).The MAC packets RNC are than aggregated and sent over an Iub interface to Node B.

In these third generation wireless networks, the base stations (Node B) are connected with the RNCs by using point to point T1/E1 lines as shown in figure 8.1.These links(T1/E1) are expensive and add up to operating costs. Also the resources (transceivers and T1/E1)are designed for peak hours traffic, so most of the time the dedicated resources are idle and wasted. Further more the T1/E1 lines are not capable of supporting bandwidth (BW) required by next generation wireless multimedia services proposed by High Speed Packet Access (HSPA, Rel.5) for Universal Mobile Telecommunications System (UMTS) and Evolution Data only (EV-DO) for Code Division Multiple Access 2000 (CDMA2000).

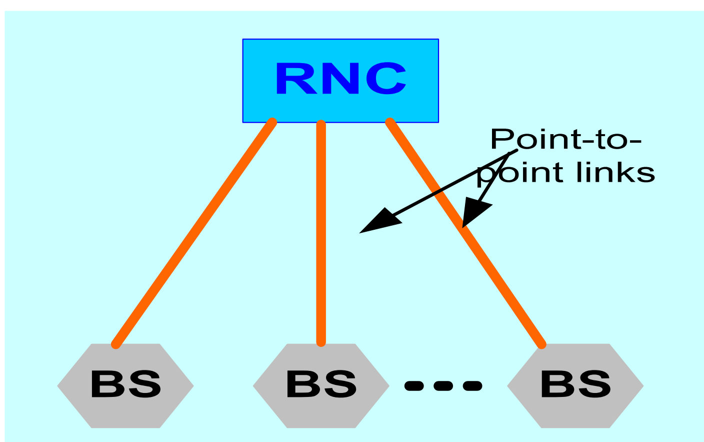


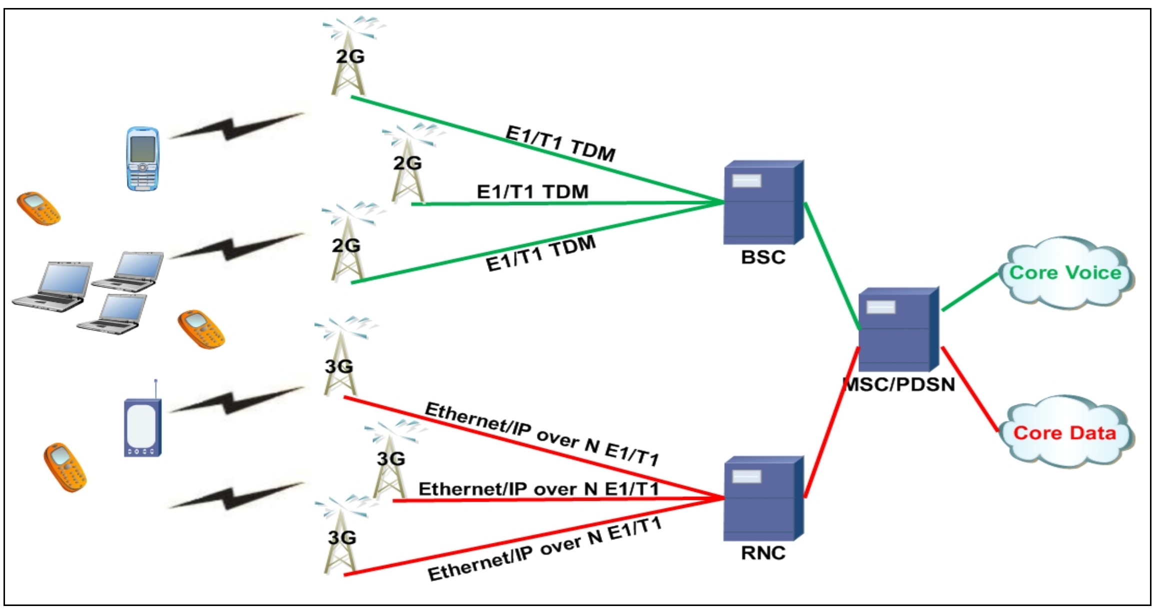
Figure 8.1. Conventional RAN architecture

In addition, today's T1/E1 lines are not suitable for next generation wireless networks due to the following reasons: 1)T1/E1, which provide symmetric bandwidth in both uplink and downlink, so they are not well suited for bursty and asymmetric data traffic; 2)T1/E1s are source of reliability problems as adding redundancy through additional point to point links is expensive; 3) T1/E1s provisioning can take significant times (in some cases months) limiting the service providers ability to react changing demands; 4) T1/E1s are dedicated and can not shared dynamically .

So the Iub link will be the bottleneck for next generation broadband wireless mobile networks. As an alternative solution if we use point to point fiber optics Iub links, which can provide Bandwidth(BW) of terabits, but it will be very expensive solution and resources will be wasted because they can not be shared dynamically between the users.

We propose a novel architecture (figure 8.2) by using a Passive Optical Networks (PON) as backhaul in the next generation wireless networks. novel Passive Optical Network (PON) based broadband wireless access network architecture to provide multimedia services (video telephony, video streaming, mobile TV, mobile emails etc) to mobile users. The proposed PON based back haul can provide Giga bit data rates and Iub interface can be dynamically shared by Node Bs. The BW is dynamically allocated and the unused BW from lightly loaded Node Bs is assigned to heavily loaded Node Bs. We also propose a novel algorithm to provide end to end Quality of Service (QoS) (between RNC and user equipment).The algorithm provides QoS bounds in the wired domain as well as in wireless domain with compensation for wireless link errors. Because of the air interface there can be certain times when the user equipment (UE) is unable to communicate with Node B (usually referred to as link error). Since the link errors are bursty and location dependent. For a proposed approach, the scheduler at the Node B maps priorities and weights for QoS into wireless MAC .The compensations for errored links is provided by the swapping of services between the active users and the user data is divided into flows ,with flows allowed to lag or lead. The algorithm guarantees (1)delay and throughput for error-free flows,(2)short term fairness among error-free flows,(3)long term fairness among errored and error-free flows,(4)graceful degradation for leading flows and graceful compensation for lagging flows.

The proposed algorithm deliver differentiated services and provide end to end Quality of Service (QoS) with wireless compensation for wireless links errors between Node B and UE. The links are said to be in errors when the UE is unable to communicate with the Node Bs. Due to inherent problems with wireless channels there can be certain times allotted to mobile host when that host is unable to transmit data. These channel error can occur from multipath fading, shadow fading or interface from other device. The scheduling algorithm uses a one step prediction to determine if a mobile host will be able to transmit during its next assigned slot.



Traditional Wireless Backhaul Architecture

8.2. Proposed Back haul architecture and its functionality

We propose a Passive Optical Network (PON) between RNC and Node Bs (Figure 8.2). A PON is a point-to-multipoint fiber optical network with no active elements in the signal's path. It consists of a single, shared optical fiber connecting a service provider's RNC (head end) to a passive star coupler (SC)/optical splitter/combiner, which is located near Node Bs.

The SC is intentionally positioned a substantial distance away from the RNC, but close enough to the Node Bs in order to save fiber. Each Node B receives a dedicated short optical fiber but shares the long distribution trunk fiber. All transmissions in a PON are performed between an RNC and Node Bs. Traffic from an RNC to Node B is called 'downstream' (point-to-multipoint), and traffic from an BS to the RNC is called 'upstream' (multipoint-to-point).

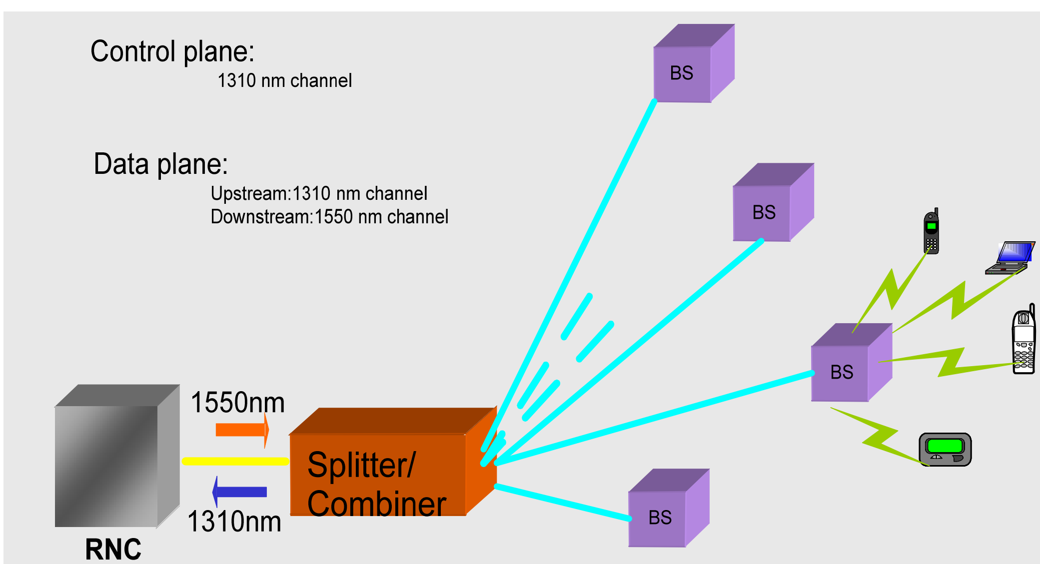


Figure 8.2. Proposed PON based RAN architecture

Two wavelengths are used: typically 1310 nm (λup) for the upstream transmission and 1550 nm (λd) for the downstream transmission. A single PON typically serves from 16-64 Node Bs. PONs can be deployed in a 1:N tree, tree-and-branch, ring, or bus topology.

In the downstream direction, PON operates as a broadcast and select network. The RNC has the entire bandwidth of the channel to broadcast standard formatted 802.3 Ethernet frames to all Node Bs. Each Node B extracts those packets that contain the Node B's unique Media Access Control (MAC) address. In the upstream direction, multiple Node Bs share the transmission channel. Thus, the Node Bs need to employ some arbitration mechanism to avoid collisions. In that case, each Node B transmits within a dedicated time slot and the RNC receives a continuous stream of collision-free frames from multiple Node Bs.

This work utilizes the 802.3ah standard as the BS will have an 802.3ah WAN interface. To control the Point-to-Multipoint fiber network, we will use the Multi-Point Control Protocol (MPCP). The MPCP specifies a control mechanism between a Master and Slaves units connected to a Point-to-Multi-Point (P2MP) segment to allow efficient transmission of data, MPCP is defined within the MAC Control layer, introducing new 64-byte control messages:

a). GATE and REPORT are used to assign and request bandwidth

b) REGISTER is used to control the auto-discovery process

MPCP provides hooks for network resource optimization. Bandwidth reporting satisfies requirements by BSs for DBA. Optical parameters are negotiated to optimize performance.

The proposed architecture offers the following significant advantages:

a) Proposed architecture can support triple play services as being proposed in 3rd generation packet based wireless networks (3G) and the emerging 4G networks with full end-to-end differentiated QoS support.

b) Proposed distributed architecture enables the base stations to communicate with each other, this is in compliance with the emerging 4G standards, whereas will be shown in upcoming sections, BSs can perform some of the functionality typically executed by the RNC thus offloading some of the RNC complexity.

8.3. Fair weighted Priority Queuing (FWPQ) Algorithm to Deliver differentiated services to end mobile users

Fair weighted priority queuing (FWPQ) allows a traffic scheduler to allocate network resources according to the required bandwidth proportions while providing fair access to all nodes within the necessary delay bounds. To work with in the context of the Weighted Fair Queuing (WFQ) algorithms, the specific levels of QoS of Ethernet need to be mapped into individual weights for use by the MAC. These weights are determined by the QoS algorithm. The Ethernet traffic is divided in to three flows as shown in table 8.1.

Priority	**Traffic Type**
1	*Voice(CBR)*
2	*Video(VBR)*
3	*Data(BE)*

Table 8.1: Ethernet prioritized Traffic

The total buffer size is shared between different flows. If the upper priority levels have buffer space available lower priority flows may use it, other wise dropped. The upper priority traffic can overwrite the buffers holding the lower priority traffic. However on average the higher priority buffers will not be full, allowing the lower priorities to buffer more data and reduce packet loss.

8.3.1. FWPQ Embedded at RNC (WIRED DOMAIN)

The total BW available at every cycle is divided between all the active available flows[6-8]. If the flow is inactive it can not reclaim its share later from other users.

1. The first priority is Constant Bit Rate(CBR) which is voice traffic. At first priority level the traffic is scheduled according to single class fair queuing model. The first priority is sensitive for delay ,the cycle time we use will be 2ms[1-5].As we know the acceptable delay time for CBR is 5ms, so maximum cycle time, the CBR traffic can wait for 2 and ½ cycles. The capacity C is give by

$C = C_{CBR}$ ……………………………………………………………………………………1

since the entire bandwidth is available.

The weight assigned to any flow 'i' in CBR (ω_{CBR_i}) among total 'j' back logged CBR flows ($j \in B(CBR)$) is given by

$$\omega_{CBR_i} = \frac{\omega_{t_{i(CBR)}}}{\sum_{j \in B(CBR)} \omega_j} \times \omega_T \quad \text{…………………………………………………………………… 2}$$

Where ω_T is total available window for CBR flows.

and $\omega_{t_{i(CBR)}}$ is total weight of CBR flow '*i*' in the buffer. After serving the total CBR traffic, the remaining window in 2 ms cycle time slot will be used by VBR and BE traffic .Thus in worst case a HOL packet in flows will be delayed while it waits for every other flow to send its share of channel BW, plus the time to send its own packet.

2. The second priority (VBR) in the system is used for flows that require guaranteed BW but not tightly delay bounds. The second priority level again uses the single class fair queuing model. However the additional delay due to higher traffic must be considered. Traffic must wait for the higher level queues to completely drain before gaining channel access.

This means level2 traffic has an effective throughput of

$$C_2 = C - \sum_{j \in F_1} \lambda_{1_j} \quad \text{………………………………………..……………………………………3}$$

λ_{1_j} =Arrival rate of flow i in class1

Fi = The set of all flows in class i

We use WRR scheme to ensure that all Node Bs get their fair share. The weight assigned to any flow '*i*' in VBR (ω_{VBR_i}) among total '*j*' back logged VBR flows ($j \in B(VBR)$) is given by

$$\omega_{VBR_i} = \frac{\omega_{t_{i(VBR)}}}{\sum_{j \in B(VBR)} \omega_j} . (\omega_T - \sum_{k \in B(CBR)} \omega_{S_k} (CBR) \quad \ldots\ldots 4$$

Where $\sum_{k \in B(CBR)} \omega_{S_k} (CBR)$ is the total served window size used by CBR flows. Thus the HOL packet in level 2, flow must wait until the level 1 queue has drained. The packet might then have to wait for all other flows in level 2 to get their share of the bandwidth before gaining access and sending the packet.

3. The lowest priority (BE) in the system processes flows that require no guarantees on BW or delay. Packets are entered in the order of arrival. An arriving packet is places in the out going queue to be sent in the assigned time slot.

$$C_3 = C_2 - \sum_{j \in F_1} \lambda_{2_j} \quad \ldots\ldots 5$$

λ_{2_i} =Arrival rate of flow i in class 2

The weight assigned to any flow '*i*' in *BE* (ω_{BE_i}) among total '*j*' back logged BE flows ($j \in B(BE)$) is given by

$$\omega_{BE_i} = \frac{\omega_{t_{i(BE)}}}{\sum_{j \in B(BE)} \omega_j}.(\omega_T - \sum_{k \in B(VBR+CBR)} \omega_{S_k}(VBR + CBR) \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots 6$$

Where $\omega_{t_{i(BE)}}$ is total weight of BE flow '*i*' in the buffer.

For level 3 HOL packet , flow must wait for both level 1 and level 2 queues to drain before being serviced .

8.3.2. FWPQ in embedded at Node B (Wireless Domain)

The algorithm is modified for wireless link errors. Flows are allowed to either lag or lead their schedule time, lagging flows then make up their lag by leading flows giving up their lead .The scheduling is done by Node-B (base station). It has knowledge of both the state of channel and which mobile station needs to transmit data to uplink direction.

Back logged flows are bounded to prevent a channel that is back logged for long period of time from exclusively grabbing the channel for a significant time upon receiving good channel.

Due to inherent problems with wireless channels there can be certain times allotted to mobile host when that host is unable to transmit data. These channel error can occur from multipath fading, shadow fading or interface from other device .

The protocol provides wireless compensation through a system of per flow credits and debits .These credits and debits account for lagging and leading flows.

If a flow is behind its ideal service due to wireless channel error .The flow is said to be lagging and thus has accrued credit. Conversely if a flow is ahead of its ideal service because of sending data when another flow has to wait, the flow is said to be leading and thus has accumulated debits.

As scheduling progresses, flows with credits and debits are swapped to compensate for wireless channel error. In effect, lagging flows can makeup their lag by causing leading flows to give up their lead.

The Node B scheduler assigns the traffic flows to the channel according to the hierarchy of priorities. The parameters specified for the ideal weighted fair queuing case can now be applied to the wireless model. Since the MAC layer will use the system of credits/debits to swap flows to compensate for local errors. The delay bound for our prioritized classes can exceed the maximum delays if the flows were error free, because of added delay due to channel errors. The lagging flows are restricted to completely grab the channel and short time fairness is assured by giving maximum of 75 % BW allocated to leading flows. The long time fairness is achieved by leading flows giving up their lead and lagging flows making up for the lag.

1.The first priority traffic (CBR) the delay acceptable for CBR is 5ms so to provide wireless compensation the packets can be hold in buffers for next cycle. If the packets wait more than 5 ms in buffer they are dropped.

Here it should be noted that the use of wireless protocol does not change the time a flow is delayed due to higher level. Additional delay will be added if the link was in error in the previous cycle. All flows at the higher priorities must be drained prior to the lower level gaining channel access.

2. For the second priority traffic (VBR), we use a simple WRR scheme before but with wireless channel compensation . If the packet detect channel error, then its packet will be skipped in that round. It compensates errored flows by new allocated slots for a later time when it is good.

3. The lowest priority (BE) in the system processes flows that require no guarantees on BW or delay. Packets are entered in the order of arrival. An arriving packet is places in the out going queue to be sent in order it was entered. In case of the channel error the packets are held in buffer until channel is error free.

8.4. Performance Evaluation

The scenario for simulation consists of 3 mobile hosts, each with a different distance from the BS. Errors are simulated using a two state Markov model as defined in [9] for the distances 10 ft(CBR) , 90 ft(VBR) and 130 ft (BE) .One mobile host rarely has any errors. While the other suffer from errors during 2 % and 8 % of the simulation time respectively. The simulations were run for 50 million packets. The duration of the simulations packets we received was around 4 seconds. To allow sufficient time for the algorithm to compensate for wireless link errors and observe the effect of algorithm, we assumed the error occurred between the first 2 seconds of the simulations. There are 3 flows CBR, VBR and BE.

HSPA can utilizes shared links in the upstream and downstream directions and if there are 40 users per BS still each user can have up to 14.4 Mbps (Rel. 5).

In our simulations the traffic was each flow was choose as follows [3].

The aggregated voice (CBR) flow arriving from the users is packetized by placing 24 bytes of data in a packet. Including Ethernet and UDP/IP headers results in a 70-byte frame generated every 125 μs. Hence, the CBR consumed the bandwidth equal to 4.48 Mbps.

The aggregated Video (VBR) flow consisted of a VBR stream with average bit rate of 16 Mbps. This corresponds to three simultaneous MPEG-2-coded video streams [10]. Since the VBR traffic is also highly bursty (LRD), it is possible that some packets in long bursts will be lost. This will happen if the entire buffer is occupied by higher priority packets.

The aggregated data (BE) flow in our simulations is an average load of 0.4 (40 Mbps).

For all the three flows we run the simulations for the following two scenarios:

1. All flows channel error free during the simulation time. (Figure. 8.3)

2. CBR flow is error free but the other two has link errors as we found in Markov model. For this scenario we run simulations by implementing FWPQ algorithm with out wireless error compensation (Figure 8.4) and FWPQ algorithm with wireless link error compensation (Figure. 8.5).

From Figure.8.3 we can see that the BE traffic suffered the most when the ambient load increased. Its delay increased and the simulation results showed that some packets were discarded when network load exceeded 80%. The VBR data also experienced an increased delay, but no packet losses were observed. The increased delay in the VBR traffic can be attributed to long bursts of data. Clearly, applying some kind of traffic shaping/policing limiting the burst size at the source would improve the situation. The CBR data experiences a very slight increase in both average and maximum delays. This is due to the fact that the packets were generated with a constant rate, i.e., no data bursts. The average delay in this case exactly followed the average cycle time, being one half of that. The maximum delay is equal to the maximum observed cycle time and for any effective network load is bounded by TMAX (2 ms with our chosen set of parameters).

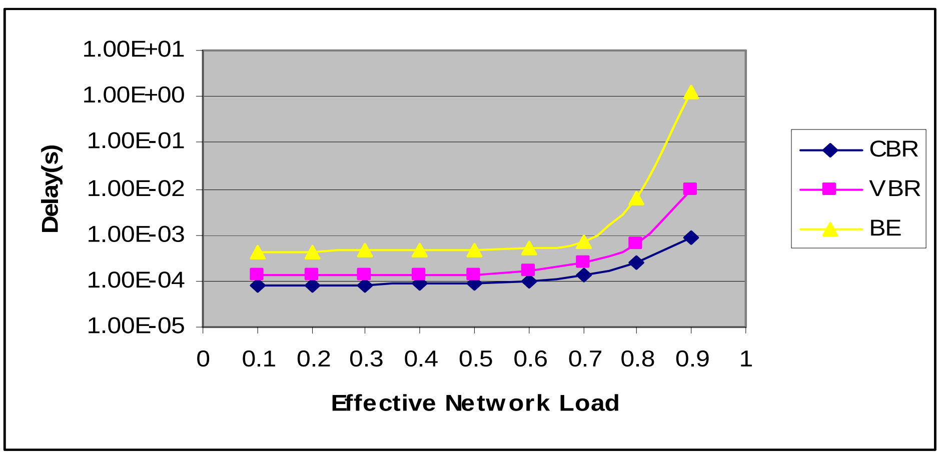


Figure.8.3. Average Packet delay for various classes of traffic (No flow in error)

From Figure.4 we see that when we include the link errors in our simulations the average packet delay, for both errored flows (VBR and BE), are much more as compare to error free case. This is, because BW allocated was wasted when there were wireless link errors.

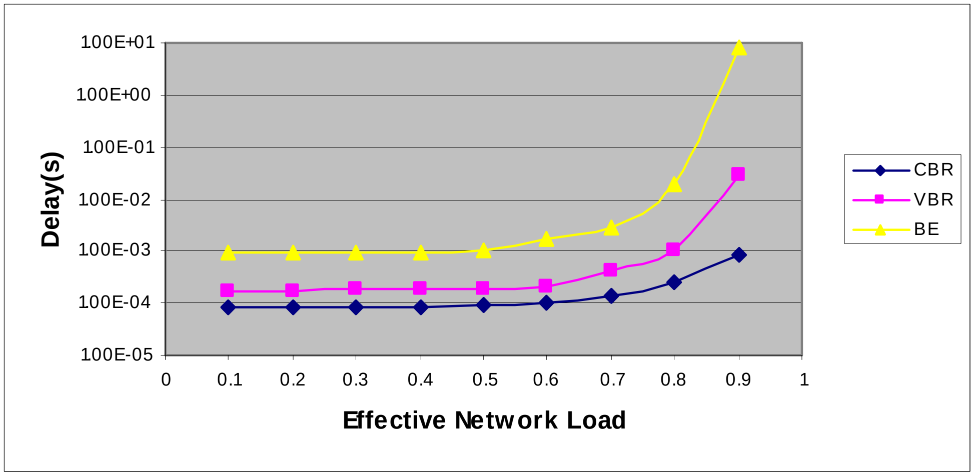


Figure8.4. Average Packet delay for various classes of traffic with out wireless compensation

(VBR and BE links error)

From figure.8.5, we can observe that by applying the FWPQ (Adjusted for wireless link errors compensation) improve the average packet delays and performance of the system. The average packet delays were improved drastically, because the BW was not wasted. If one UE was experiencing an error in the link, the Node B scheduler assigned the BW to error free flows and assign credits and debits to make the flows in sync afterwards by swapping the BW assigned to flows. Still the errors are little more than error free case because some times both VBR and BE were in error and BW was not utilized by any flow.

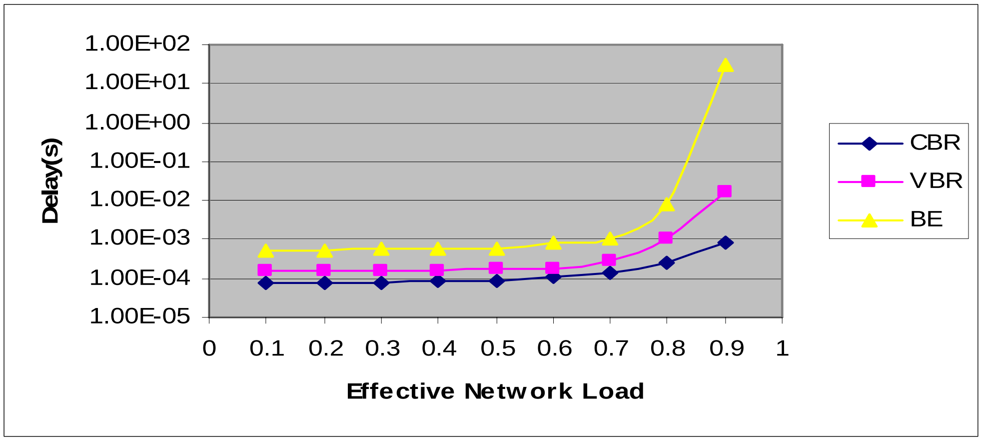


Figure8.5. Average Packet delay for various classes of traffic with wireless compensation algorithm (VBR and BE links error)

8.5. Conclusion

In this paper we proposed a PON based wireless radio access network to carry traffic in the next generation backhaul networks between the Node Bs and radio network controllers. The proposed backhaul is cost effective solution with no active elements, which will not only reduce the installation costs but also will reduce operating costs. The links are dynamically shared among the Node Bs, so if some Node Bs are lightly loaded the BW can be allocated to overloaded Node Bs. The proposed backhaul can provide the data rates of Giga bits which will be more than enough to handle all the next generations wireless requirements for the broadband wireless access networks.

Further more, in the proposed architecture we also provide a fair priority weighted queuing algorithm that not only deliver the differentiated services to the mobile end users but it also provide compensation for wireless link errors and improve the packet delays.

The scheduler maps QoS parameters specified at call setup into priorities and weights for the packetized wireless channel. Rather than redefining QoS parameters for the wireless channel, the model uses and enforces the wired Ethernet standards across the combined network.

The algorithm is work conserving and minimum BW guarantees are satisfied.

8.6. References

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Chapter 9- Conclusion and Future work

Chapter 9

9.1. Future work and conclusion

9.1.1. Conclusion

The work is composed of two overlapping phases. In the first phase of our research, we proposed a novel Ethernet-based wired/wireless broadband access networking architecture that utilizes the existing wired trunk feeder fiber of typical EPON infrastructure along with a hybrid FSO/RF reliable wireless connectivity to the end-users (ONUs). By combining the benefits of both FSO and RF technologies, the proposed integrated networking solution can provide a downstream bandwidth of up to 2.5 Gbps per wavelength and 99.999% availability at a range of 1 km in all weather conditions fewer packet drops, less delay and better through put. The architecture is designed in such a way that the maximum distance covered by HFR link will be 1 km or less. To deliver QoS in the proposed architecture we have developed two algorithms (DWQ-T and F-NSPQ) and compared it to standard SPQ algorithm by integrating all of them to DRR for fairness among the customers of SP. We assumed all the end-users have the same Service

Level Agreement(SLA) with the Service Provider (SP). We emulate the algorithm by simulations and observe the improvement in the performance by out proposed algorithm.

Further more because of wireless links in the proposed architecture there can be some times when the user equipment is unable to communicate with the host, so we propose a new algorithm to provide QoS service ,which provide the compensation for wireless link errors. We propose Multiclass Weighted Priority Scheduling (MWPS)algorithm .Because of different types of traffic (voice, video, data) ,each of which has different requirements for delay and bandwidth ,so in the proposed the algorithm we provide different queuing algorithm for different classes of service.

In the second phase of our work we modified the architecture to provide the broadband wireless access to the mobile end users. We proposed a PON based wireless radio access network to carry traffic in the next generation backhaul networks between the Node Bs and radio network controllers. The proposed backhaul is cost effective solution with no active elements, which will not only reduce the installation costs but also will reduce operating costs. The links are dynamically shared among the Node Bs, so if some Node Bs are lightly loaded the BW can be allocated to overloaded Node Bs. The proposed backhaul can provide the data rates of Giga bits which will be more than enough to handle all the next generations wireless requirements for the broadband wireless access networks.

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The scheduler maps QoS parameters specified at call setup into priorities and weights for the packetized wireless channel. Rather than redefining QoS parameters for the wireless channel, the model uses and enforces the wired Ethernet standards across the combined network.The algorithm is work conserving and minimum BW guarantees are satisfied.

9.1.2. Future Work

There are some very important issues which need to be discussed in future work.

First Issue:

The first issue need to be addressed is mobility management in the proposed architecture.

Mobility management is further divided in to two categories

1. **Location management**

i. Registering the mobile user with the Node B.

ii. Data delivery

2. **Handoff management**

When the signal power drop below a certain level then either it move to another Node B or we need to allocate new access channels with in the same Node B .If the signal fade due to moving to another Node B with in same RNC or some RNC covered area then it need to register with the new Node B and all the information from previous Node B and buffered data for the mobile user need to be a transferred to new Node B.

The handoff can be divided in two three categories

1- User assisted (initiated)

2- Network assisted

3- Combination of above two.

Two registers will be need

1- Home location register (HLR)

- It will keep the services subscribed ,billing information and home location.

2-Visitor location register (VLR)

- Lists pertinent data concerning location through out the network.

Second issue:

The further simulations need to be done on the dynamic allocation of the back bone links capacity so that if some links are under utilized it can be used by over loaded links .Also if more bandwidth is required we can use DWDM ,where the wave lengths are dynamically allocated and overloaded Node Bs can have more than one wave length and lightly loaded Node Bs can share the wave length assigned to them with some other BSs.

Third Issue:

We proposed PON based solutions for the backhaul bottleneck problem in wireless access networks .The PON network implemented was start based ,so if instead of using star based PON we can also utilize ring based PON. Where two rings will be deployed .Because all the Node Bs will be connected with each other and with the RNC through Ring .There are some benefits by using ring based PON ,The first benefit is protections and restoration will be available just in case one ring fail due to some fiber optic cuts the second can be used in the mean time the first one is restored .The second benefit can be achieved ,if we use two wave lengths one for the LAN traffic and one for the traffic going out of the area of RNC. Because all Node Bs are connected with each other so instead of going LAN traffic also to the RNC which are already congested because of .So the over head on the RNC will be reduced.

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Chapter 6

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