8 Ethernet Passive Optical Network (EPON)

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8.1 Introduction

In recent years the telecommunications backbone has experienced substantial growth; however, little has changed in the access network. The tremendous growth of Internet traffic has accentuated the aggravating lag of access network capacity. The "last mile" still remains the bottleneck between high-capacity Local Area Networks (LANs) and the backbone network. The most widely deployed "broadband" solutions today are Digital Subscriber Line (DSL) and cable modem (CM) networks. Although they are an improvement compared to 56 Kbps dial-up lines, they are unable to provide enough bandwidth for emerging services such as Video-On-Demand (VoD), interactive gaming or two-way video conferencing. A new technology is required; one that is inexpensive, simple, scalable, and capable of delivering bundled voice, data and video services to an end-user over a single network. Ethernet Passive Optical Networks (EPONs), which represent the convergence of low-cost Ethernet equipment and low-cost fiber infrastructure, appear to be the best candidate for the next-generation access network.

8.1.1 Traffic Growth

Data traffic is increasing at an unprecedented rate. Sustainable data traffic growth rate of over 100% per year has been observed since 1990. There were periods when a combination of economic and technological factors resulted in even larger growth rates, e.g., 1000% increase per year in 1995 and 1996 [1]. This trend is likely to continue in the future. Simply put, more and more users are getting online, and those who are already online are spending more time online and are using more bandwidth-intensive applications. Market research shows that, after upgrading to a broadband connection, users spend about 35% more time online than before [2]. Voice traffic is also growing, but at a much slower rate of 8% annually. According to most analysts, data traffic has already surpassed the voice traffic. More and more subscribers telecommute, and require the same network performance as they see on corporate LANs. More services and new applications will become available as bandwidth per user increases.

Neither DSL nor cable modems can keep up with such demand. Both technologies are built on top of existing communication infrastructure not optimized for data traffic. In cable modem networks, only a few RF channels are dedicated for data, while the majority of bandwidth is tied up servicing legacy analog video. DSL copper networks do not allow sufficient data rates at required distances due to signal distortion and crosstalk. Most network operators have come to the realization that a new, data-centric solution is necessary. Such a technology would be optimized for Internet Protocol (IP) data traffic. The remaining services, such a voice or video, will converge into a digital format and a true full-service network will emerge.

8.1.2 Evolution of the "First Mile"

The *first mile*? Once called the last mile, the networking community has renamed this network segment to the first mile, to symbolize its priority and importance*. The first mile connects the service provider central offices to business and residential subscribers. Also referred to as the subscriber access network, or the local loop, it is the network infrastructure at the neighborhood level. Residential subscribers demand first-mile access solutions that are broadband, offer Internet media-rich services, and are comparable in price with existing networks.

Incumbent telephone companies responded to Internet access demand by deploying Digital Subscriber Line (DSL) technology. DSL uses the same twisted pair as telephone lines and requires a DSL modem at the customer premises and Digital Subscriber Line Access Multiplexer (DSLAM) in the central office (CO). The data rate provided by DSL is typically offered in a range from 128 Kbps to 1.5 Mbps. While this is significantly faster than an analog modem, it is well shy of being considered "broadband," in that it cannot support emerging voice, data, and video applications. In addition, the physical area that one central office can cover with DSL is limited to distances less than 18000 ft (5.5 km), which covers approximately 60% of potential subscribers. And even though, to increase DSL coverage remote DSLAMs (R-DSLAMs) may be deployed closer to

* Ethernet in the First Mile Alliance was formed in December 2001 by Alloptic, Cisco Systems, Elastic Networks, Ericsson, Extreme Networks, Finisar, Intel, NTT, and World Wide Packets. For more information, visit www.efmalliance.org

subscribers, network operators, in general, do not provide DSL services to subscribers located more than a 12000 ft from CO due to increased costs [3].

Cable television companies responded to Internet service demand by integrating data services over their coaxial cable networks, which were originally designed for analog video broadcast. Typically, these hybrid fiber coax (HFC) networks have fiber running between a video head-end or a hub to a curbside optical node, with the final drop to the subscriber being coaxial cable, repeaters, and tap couplers. The drawback of this architecture is that each shared optical node has less than 36 Mbps effective data throughput, which is typically divided between 2000 homes, resulting in frustrating slow speed during peak hours. To alleviate bandwidth bottlenecks, optical fibers, and thus optical nodes, are penetrating deeper into the first mile.

The next wave of local access deployment promises to bring fiber to the building (FTTB) and fiber to the home (FTTH). Unlike previous architectures, where fiber is used as a feeder to shorten the lengths of copper and coaxial networks, these new deployments use optical fiber throughout the access network. New optical fiber network architectures are emerging that are capable of supporting gigabit per second speeds, at costs comparable to those of DSL and HFC networks.

8.1.3 Next-Generation Access Network

Optical fiber is capable of delivering bandwidth-intensive, integrated, voice, data and video services at distances beyond 20 kilometers in the subscriber access network. A logical way to deploy optical fiber in the local access network is to use a point-to-point (PtP) topology, with dedicated fiber runs from the CO to each end-user subscriber (Figure 8-1a). While this is a simple architecture, in most cases it is cost prohibitive due to the fact that it requires significant outside plant fiber deployment as well as connector termination space in the Local Exchange. Considering N subscribers at an average distance L km from the central office, a PtP design requires $2N$ transceivers and $N*L$ total fiber length (assuming that a single fiber is used for bi-directional transmission).

To reduce fiber deployment, it is possible to deploy a remote switch (concentrator) close to the neighborhood. That will reduce the fiber consumption to only L km (assuming negligible distance between the switch and customers), but will actually increase the number of transceivers to $2N+2$, as there is one more link added to the network (Figure 8-1b). In addition, curb-switched network architecture requires electrical power as well as back-up power at the curb switch.

Currently, one of the highest costs for Local Exchange Carriers (LECs) is providing and maintaining electrical power in the local loop.

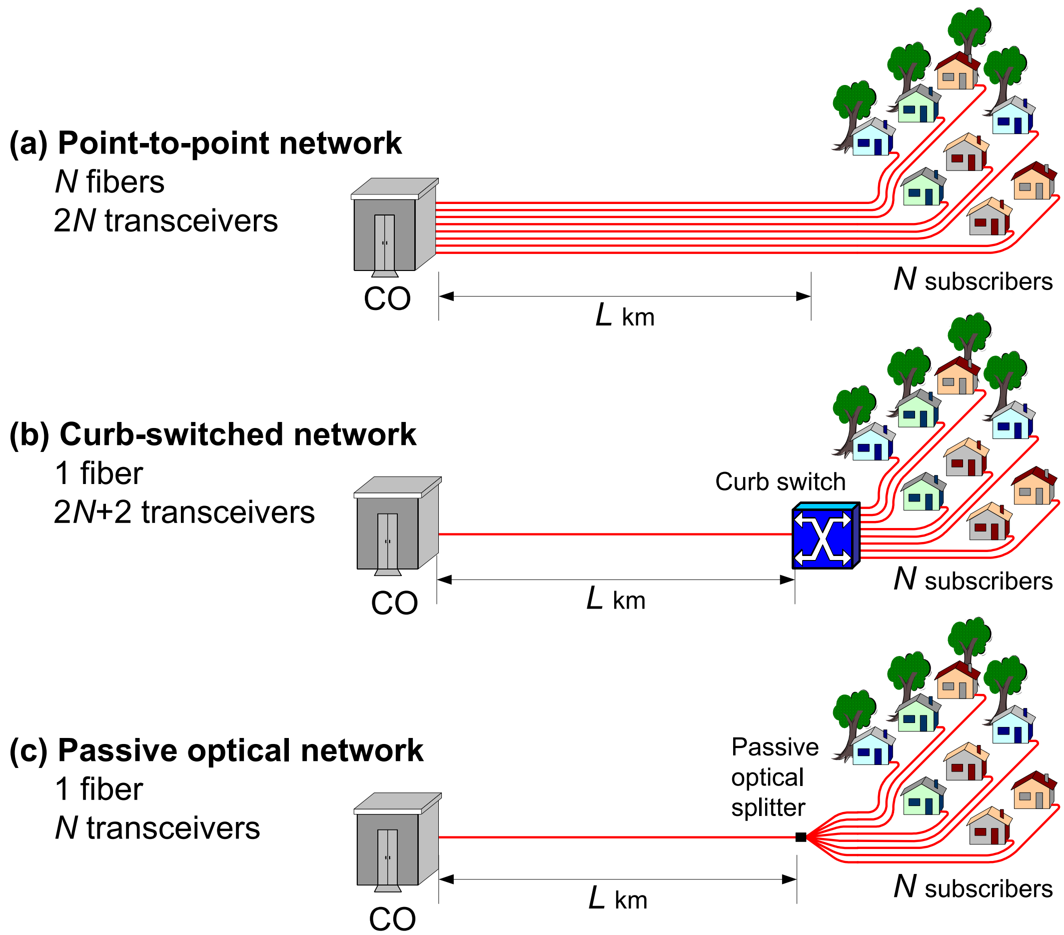


Figure 8-1. Fiber to the home (FTTH) deployment scenarios.

Therefore, it is logical to replace the hardened (environmentally protected) active curb-side switch with an inexpensive passive optical splitter. Passive Optical Network (PON) is a technology viewed by many as an attractive solution to the first mile problem [4, 5]; a PON minimizes the number of optical transceivers, central office terminations and fiber deployment. A PON is a point-to-multipoint (PtMP) optical network with no active elements in the signals' path from source to destination. The only interior elements used in PON are passive optical components, such as optical fiber, splices and splitters. An access network based on a single-fiber PON only require $N + 1$ transceivers and L km of fiber (Figure 8-1c).

8.2 Overview of PON technologies

8.2.1 Optical Splitters/Combiners

A passive optical network employs a passive (not requiring any power) device to split optical signal (power) from one fiber into several fibers and reciprocally, to combine optical signals from multiple fibers into one. This device is an optical coupler. In its simplest form, an optical coupler consists of two fibers fused together. Signal power received on any input port is split between both output ports. The splitting ratio of a splitter can be controlled by the length of the fused region and therefore is a constant parameter.

$N \times N$ couplers are manufactured by staggering multiple 2x2 couplers (Figure 8-2) or by using planar waveguide technology.

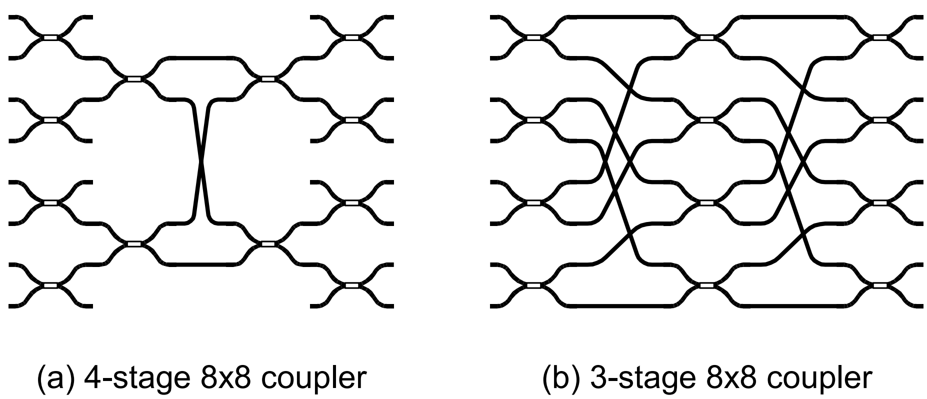


Figure 8-2. 8x8 couplers created from multiple 2x2 couplers.

Couplers are characterized by the following parameters:

Splitting Loss – Power level at the coupler's output vs. power level at its input, measured in dB. For an ideal 2x2 coupler, this value is 3 dB. Figure 8-2 illustrates two topologies for 8x8 couplers based on 2x2 couplers. In a 4-stage topology (Figure 8-2.a), only 1/16 of the input power is delivered to each output. Figure 8-2.b shows a more efficient design called *multistage interconnection network* [6]. In this arrangement, each output receives 1/8 of the input power.

Insertion Loss – Power loss resulting from imperfections of the manufacturing process. Typically, this value ranges from 0.1 dB to 1 dB.

Directivity – Amount of input power leaked from one input port to another input port. Couplers are highly directional devices with the directivity parameter reaching 40 – 50 dB.

Very often, couplers are manufactured to have only one input or one output. A coupler with only one input is referred to as a *splitter*. A coupler with only one output is called a *combiner*. Sometimes, 2x2 couplers are made highly asymmetric (with splitting ratios 5/95 or 10/90). This kind of couplers is used to branch off a small portion of signal power, for example, for monitoring purposes. Such devices are called tap couplers.

8.2.2 PON Topologies

Logically, the first mile is a PtMP network, with a CO servicing multiple subscribers. There are several multipoint topologies suitable for the access network, including tree, tree-and-branch, ring, or bus (Figure 8-3). Using 1:2 optical tap couplers and 1:N optical splitters, PONs can be flexibly deployed in any of these topologies. In addition, PONs can be deployed in redundant configurations such as double rings or double trees; or redundancy may be added to only a part of the PON, say the trunk of the tree (Figure 8-3d) (also refer to [7] for more redundant topologies).

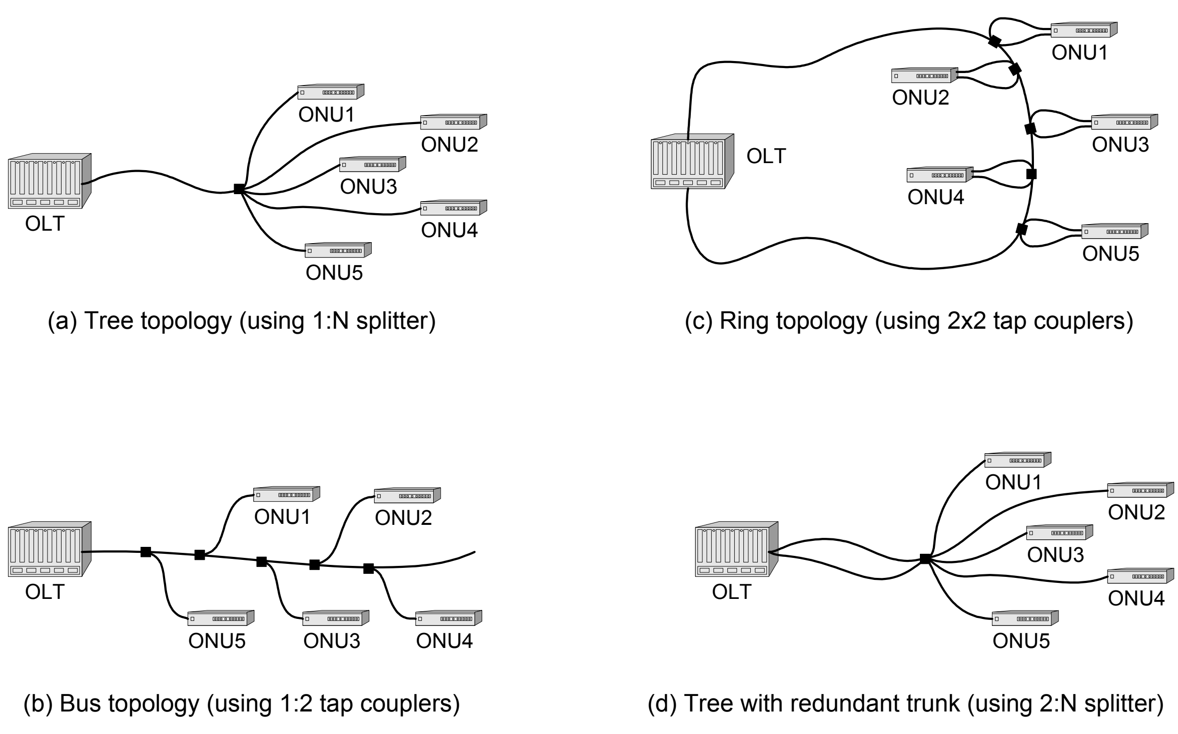


Figure 8-3. PON topologies.

All transmissions in a PON are performed between an Optical Line Terminal (OLT) and Optical Network Units (ONUs) (Figure 8-3). The OLT resides in the CO and connects the optical access network to the metropolitan area network (MAN) or wide area network (WAN) also known as backbone or long-haul network. The ONU is located either at the end-user location (FTTH and FTTB), or at the curb, resulting in fiber to the curb (FTTC) architecture.

The advantages of using PONs in subscriber access networks are numerous:

- PONs allow for long reach between the CO and customer premises, operating at distances over 20 km.
- PONs minimize fiber deployment in both the CO and the local loop.
- PONs provide higher bandwidth due to deeper fiber penetration, offering gigabit-per-second solutions.
- Operating in the downstream as a broadcast network, PONs allow for video broadcasting either as IP video, or analog video.
- PONs eliminate the necessity of installing active multiplexers at the splitting locations, thus relieving network operators from the gruesome task of maintaining active curb-side units and providing power to them. Instead of active devices in these locations, PONs use small passive optical splitters, located in splice trays, and deployed as part of the optical fiber cable plant.
- Being optically transparent end-to-end, PONs allow upgrades to higher bit rates or additional wavelengths.

8.2.3 WDM vs. TDM PONs

In the downstream direction (from OLT to ONUs), a PON is a point-to-multipoint network. The OLT typically has the entire downstream bandwidth available to it at all times. In the upstream direction, a PON is a multipoint-to-point network: multiple ONUs transmit all towards one OLT. Directional properties of a passive splitter/combiner are such that an ONU's transmission cannot be detected by other ONUs. However, data streams from different ONUs transmitted simultaneously still may collide. Thus, in the upstream direction (from user to network), PON should employ some channel separation mechanism to avoid data collisions and fairly share the trunk fiber channel capacity and resources.

One possible way of separating the ONU's upstream channels is to use a wavelength-division multiplexing (WDM), in which each ONU operates on a different wavelength. While it is a simple solution (from a theoretical perspective), it remains cost-prohibitive for an access network. A WDM solution would require either a tunable receiver, or a receiver array at the OLT to receive multiple channels. An even more serious problem for network operators would be wavelength-specific ONU inventory: instead of having just one type of ONU, there would be multiple types of ONUs based on their laser wavelength. Each ONU will have to use a laser with narrow and controlled spectral width, and thus will become more expensive. It would also be more problematic for an unqualified user to replace a defective ONU because a unit with wrong wavelength may interfere with some other ONU in the PON. Using tunable lasers in ONUs may solve the inventory problem, but is too expensive at the current state of technology. For these reasons, a WDM PON network is not an attractive solution in today's environment.

Several alternative solutions based on WDM have been proposed, namely wavelength-routed PON (WRPON). A WRPON uses an arrayed waveguide grating (AWG) instead of wavelength-independent optical splitter/combiner. We refer the reader to [8] for a detailed overview of these approaches.

In one variation, ONUs use external modulators to modulate the signal received from the OLT and send it back upstream. This solution, however, is not cheap either; it requires additional amplifiers at or close to the ONUs to compensate for signal attenuation after the round-trip propagation, and it requires more expensive optics to limit the reflections, since both downstream and upstream channels used the same wavelength. Also to allow independent (non-arbitrated) transmission from each of N ONUs, the OLT must have N receivers – one for each ONU.

In another variation, ONUs contain cheap laser-emitting diodes (LEDs) whose wide spectral band was sliced by the AWG on the upstream path. This approach still requires multiple receivers at the OLT. If, however, a single tunable receiver is used at the OLT, then a data stream from only one ONU can be received at a time, which in effect makes it a time-division multiplexed (TDM) PON.

In a TDM PON, simultaneous transmissions from several ONUs will collide when reaching the combiner. In order to prevent data collisions, each ONU must transmit in its own transmission window (timeslot). One of the major advantages of a TDM PON is that all ONUs can operate on the same wavelength and be absolutely identical component-wise. The OLT will also need a single receiver. A transceiver in an ONU must operate at the full line rate, even though the bandwidth available to the ONU is lower. However, this property also allows the TDM PON to efficiently change the bandwidth allocated to each ONU by changing the assigned timeslot size, or even employ statistical multiplexing to fully utilize the bandwidth available in the PON.

In a subscriber access network, most of the traffic flows downstream (from network to users) and upstream (from users to the network), but not peer-to-peer (user to user). Thus, it seems reasonable to separate the downstream and the upstream channels. A simple channel separation can be based on space division multiplexing (SDM) where separate PONs provided for downstream and for upstream transmissions. To save optical fiber and reduce cost of repair and maintenance, a single fiber may be used for bi-directional transmission. In this case, two wavelengths are used: typically 1310 nm ($\lambda 1$) for the upstream transmission and 1550 nm ($\lambda 2$) for the downstream transmission (Figure 8-4). The channel capacity on each wavelength can be flexibly divided between the ONUs.

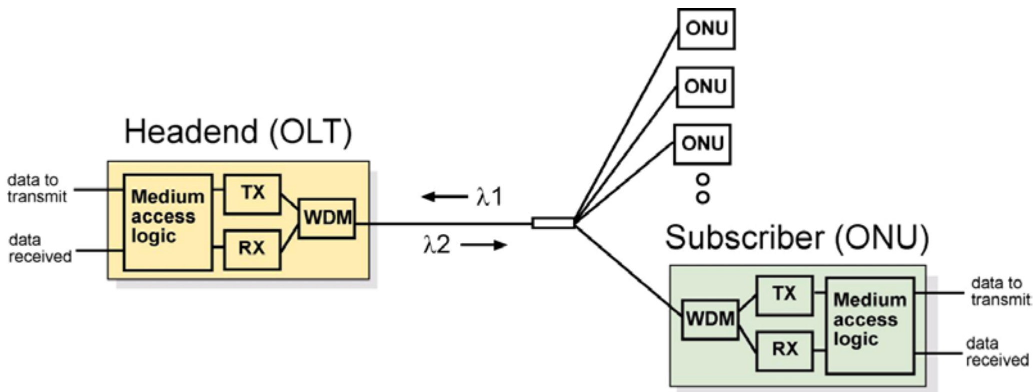


Figure 8-4. PON using a single fiber.

Time-sharing appears to be the preferred method today for optical channel sharing in an access network as it allows for a single upstream wavelength, such as 1310 nm, and a single transceiver in the OLT, resulting in a cost-effective solution.

8.2.4 Burst-Mode Transceivers

Due to unequal distances between CO and ONUs, optical signal attenuation in the PON is not the same for each ONU. The power level received at the OLT will be different for each timeslot (called the *near-far* problem). Figure 8-5 depicts power levels of four timeslots received by the OLT from four different ONUs in a TDM PON. As shown, one ONU's signal strength is lower at the OLT most likely due to its longer distance. If the receiver in OLT is adjusted to properly receive high-power signal from a close ONU, it may mistakenly read ones as zeros when receiving weak signal from a distant ONU. In the opposite case, if the receiver is trained on a weak signal, it may read zeros as ones when receiving a strong signal.

To properly detect the incoming bit stream, the OLT receiver must be able to quickly adjust its zero-one threshold at the beginning of each received timeslot, i.e., it should operate in *burst mode*. A burst mode receiver is necessary only in the OLT. The ONUs read a continuous bit stream (data or idles) sent by the OLT and do not need to re-adjust quickly.

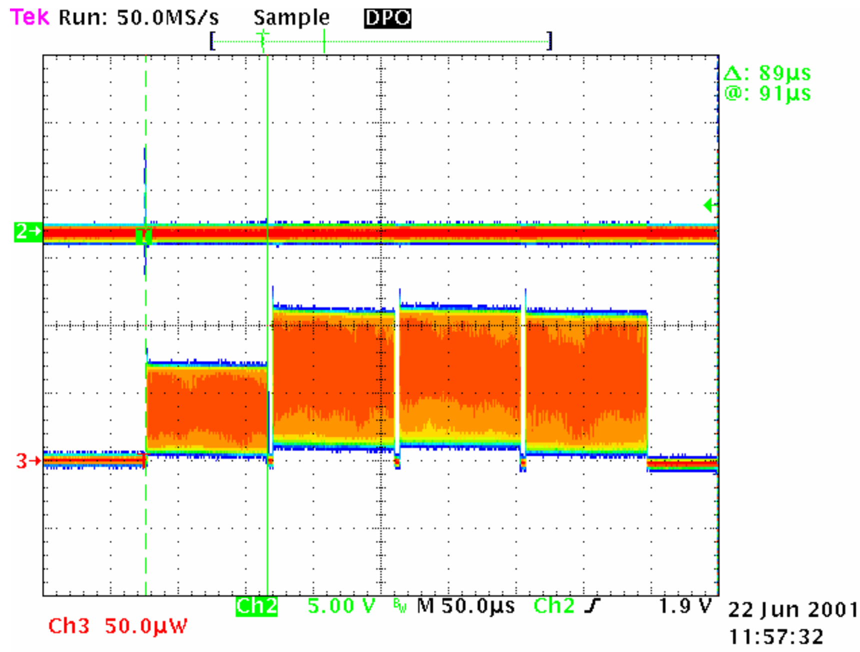


Figure 8-5. Illustration of near-far problem in a TDM PON:

a snapshot of received power level from four timeslots.

An alternative approach is to allow ONUs to adjust their transmitter powers such that power levels received by OLT from all ONUs become the same. This method is not particularly favored by transceiver designers as it makes the ONU hardware more complicated, requires special signaling protocol for feedback from the OLT to each ONU, and most importantly, may degrade the performance of all ONUs to that of a most distant unit.

Another issue is that it is not enough just to disallow ONUs to send any data. The problem is that, even in the absence of data, lasers generate spontaneous emission noise. Spontaneous emission noise from several ONUs located close to the OLT can easily obscure the signal from a distant ONU (*capture* effect). Thus, an ONU must shut down its laser between the timeslots. Because a laser cools down when it is turned off, and warms up when it is turned on, its emitted power may fluctuate at the beginning of a transmission. It is important that the laser be able to stabilize quickly after being turned on.

8.3 Ethernet PON (EPON) Access Network

Ethernet PON (EPON) is a PON-based network that carries data traffic encapsulated in Ethernet frames (defined in the IEEE 802.3 standard). It uses a standard 8b/10b line coding (8 user bits encoded as 10 line bits) and operates at standard Ethernet speed.

8.3.1 Why Ethernet?

Passive optical networking has been considered for the access network for quite some time, even well before the Internet spurred bandwidth demand. The Full Service Access Network (FSAN) recommendation (ITU G.983) defines a PON-based optical access network that uses ATM as its layer 2 protocol. In 1995, when the FSAN initiative was started, ATM had high hopes of becoming the prevalent technology in the LAN, MAN and backbone. However, since that time, Ethernet technology has leapfrogged ATM. Ethernet has become a universally accepted standard, with over 320 million port deployments worldwide, offering staggering economies of scale [9]. High-speed Gigabit Ethernet deployment is widely accelerating and 10 Gigabit Ethernet products are becoming available. Ethernet, which is easy to scale and manage, is winning new grounds in MAN and WAN. Considering the fact that 95% of LANs use Ethernet, it becomes clear that ATM PON may not be the best choice to interconnect two Ethernet networks.

One of ATM's shortcomings is the fact that a dropped or corrupted ATM cell will invalidate the entire IP datagram. However, the remaining cells carrying the portions of the same IP datagram will propagate further, thus consuming network resources unnecessarily. Also, ATM imposes a cell tax on variable-length IP packets. For example, for the tri-modal packet-size distribution reported in [10], the cell tax is approximately 13%, i.e., to send the same amount of user's data an ATM network must transmit 13% more bytes than an Ethernet network (counting 64-bit preamble and 96-bit minimum inter-frame gap (IFG) in Ethernet and 12 bytes of overhead associated with ATM adaptation layer 5 (AAL-5)). And finally, perhaps most importantly, ATM did not live up to its promise of becoming an inexpensive technology – vendors are in decline and manufacturing volumes are relatively low. ATM switches and network cards are significantly (roughly 8x) more expensive than Ethernet switches and network cards [9].

On the other hand, Ethernet looks like a logical choice for an IP data-optimized access network. Newly-adopted quality-of-service (QoS) techniques have made Ethernet networks capable of supporting voice, data and video. These techniques include full duplex transmission mode, prioritization (P802.1p), and virtual LAN (VLAN) tagging (P802.1Q). Ethernet is an inexpensive technology, which is ubiquitous and interoperable with a variety of legacy equipment. The rest of this article will focus on EPONs.

8.3.2 Principle of Operation

The IEEE 802.3 standard defines two basic configurations for an Ethernet network. In one configuration it can be deployed over a shared medium using the Carrier Sense Multiple Access with Collision Detection (CSMA/CD) protocol. In another configuration, stations may be connected through a switch using full-duplex point-to-point links. Properties of EPON are such that it cannot be considered either a shared medium or a point-to-point network; rather, it is a combination of both.

In the downstream direction, Ethernet frames transmitted by the OLT pass through a *1:N* passive splitter and reach each ONU. *N* is typically between 4 and 64. This behavior is similar to a shared-medium network. Because Ethernet is broadcast by nature, in the downstream direction (from network to user), it fits perfectly with the Ethernet PON architecture: packets are broadcast by the OLT and extracted by their destination ONU based on the media-access control (MAC) address (Figure 8-6).

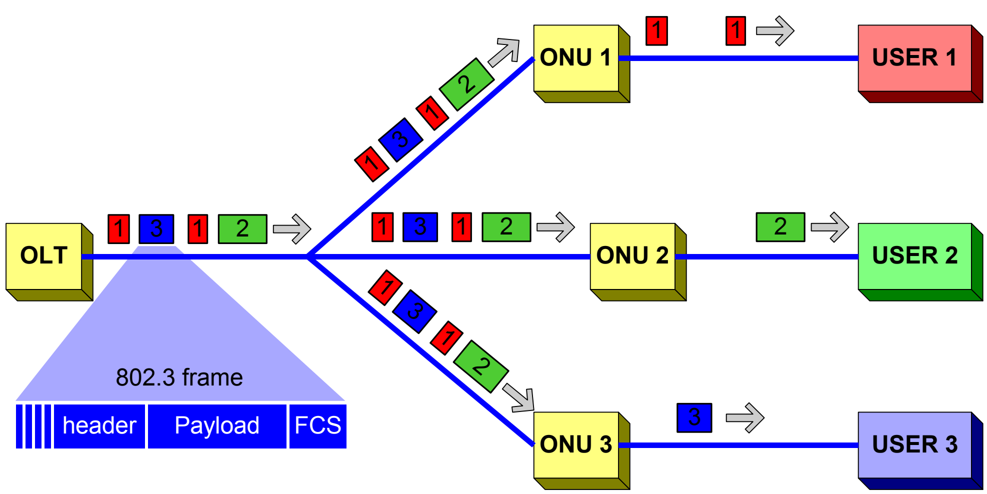


Figure 8-6. Downstream traffic in EPON.

In the upstream direction, due to the directional properties of a passive optical combiner, data frames from any ONU will only reach the OLT, and not other ONUs. In that sense, in the upstream direction, the behavior of EPON is similar to that of a point-to-point architecture. However, unlike in a true point-to-point network, in EPON data frames from different ONUs transmitted simultaneously still may collide. Thus, in the upstream direction (from users to network) the ONUs need to employ some arbitration mechanism to avoid data collisions and fairly share the fiber-channel capacity.

A contention-based media access mechanism (something similar to CSMA/CD) is difficult to implement because ONUs cannot detect a collision at the OLT (due to directional properties of optical splitter/combiner). An OLT could detect a collision and inform ONUs by sending a jam signal; however, propagation delays in PON, which can exceed 20 km in length, can greatly reduce the efficiency of such a scheme. Contention-based schemes also have a drawback of providing a non-deterministic service, i.e., node throughput and channel utilization may be described as statistical averages. There is no guarantee of a node getting access to the media in any small interval of time. It is not a problem for CSMA/CD-based enterprise networks where links are short, typically over-provisioned, and traffic predominantly consists of data. Subscriber access networks, however, in addition to data, must support voice and video services, and thus must provide some guarantees on timely delivery of these traffic types.

To introduce determinism in the frame delivery, different non-contention schemes have been proposed. Figure 8-7 illustrates an upstream time-shared data flow in an EPON.

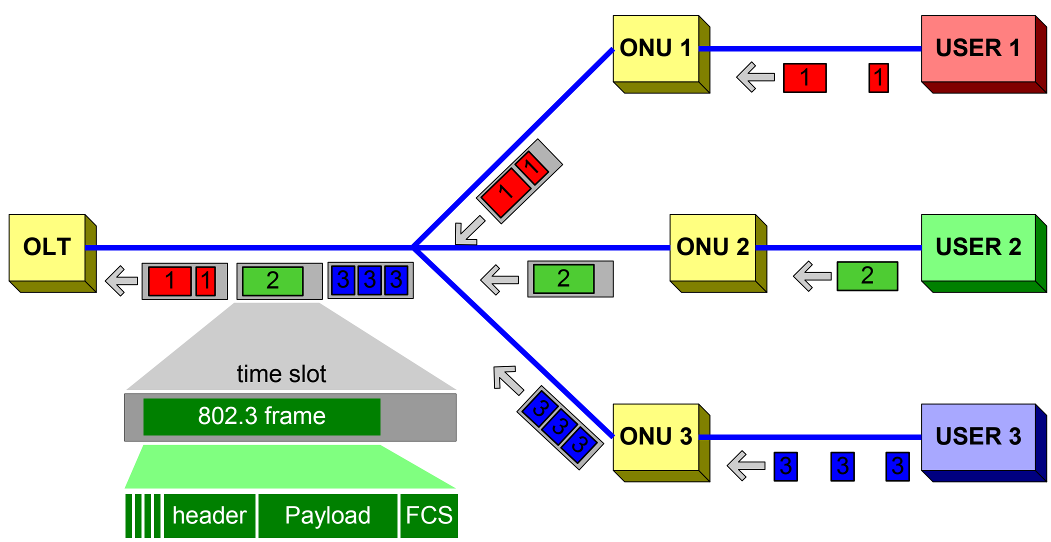


Figure 8-7. Upstream traffic in EPON.

All ONUs are synchronized to a common time reference and each ONU is allocated a timeslot. Each timeslot is capable of carrying several Ethernet frames. An ONU should buffer frames received from a subscriber until its timeslot arrives. When its timeslot arrives, the ONU would "burst" all stored frames at full channel speed which must correspond to one of standard Ethernet rates (10/100/1000/10000 Mbps). If there are no frames in the buffer to fill the entire timeslot, idles 10-bit characters are transmitted. The possible timeslot allocation schemes could range from a static allocation (fixed time-division multiple access (TDMA)) to a dynamically adapting scheme based on instantaneous queue size in every ONU (statistical multiplexing scheme). There are more allocation schemes possible, including schemes utilizing notions of traffic priority, Quality of Service (QoS), Service-Level Agreements (SLAs), over-subscription ratios, etc.

Decentralized approaches to implement a dynamic slot assignment scheme are also possible, in which ONUs decide when to send data and for how long. These schemes are somewhat similar to a token-passing approach, except that in this case it is a passive ring. In such a scheme, every ONU, before sending its data, will send a special message announcing how many bytes it is about to send. The ONU that is scheduled next (say, in round-robin fashion) will monitor the transmission of the previous ONU and will time its transmission such that it arrives to the OLT right after the transmission from the previous ONU. Thus, there will be no collision and no bandwidth will be wasted. This scheme is similar to hub polling [11]. However, this scheme has a major limitation: it requires connectivity (communicability) between ONUs. That imposes some constraints on PON topology; namely, the network should be deployed as a ring or as a broadcasting star. This requirement is not desirable as (a) it may require more fiber to be deployed, or (b) fiber plant with different topology might be already pre-deployed. In general, a preferred algorithm shall support any point-to-multipoint PON topology.

In an optical access network, we can count only on connectivity from the OLT to every ONU (downstream traffic) and every ONU to the OLT (upstream traffic). That is true for all PON topologies. Therefore, the OLT remains the only device that can arbitrate time-division access to the shared channel.

The challenge of implementing an OLT-based dynamic arbitration scheme is in the fact that the OLT does not know how many bytes of data each ONU has buffered. The burstiness of data traffic precludes a queue occupancy prediction with any reasonable accuracy. If the OLT is to make an accurate timeslot assignment, it should know the state of a given ONU exactly. One solution may be to use a polling scheme based on Grant and Request messages. Requests are sent from an ONU to report changes in an ONU's state, e.g., the amount of buffered data. The OLT processes all Requests and allocates different transmission windows (timeslots) to ONUs. Slot-assignment information is delivered to ONUs using Grant messages.

The advantage of having centralized intelligence for the slot-allocation algorithm is that the OLT knows the state of the entire network and can switch to another allocation scheme based on that information; the ONUs don't need to monitor the network state or negotiate and acknowledge new parameters. That will make ONUs simpler and cheaper and the entire network more robust.

8.3.3 Multi-Point Control Protocol (MPCP)

To support a timeslot allocation by the OLT, the multi-point control protocol (MPCP) is being developed by the IEEE 802.3ah task force. This protocol relies on two Ethernet messages: GATE and REPORT. GATE message is sent from OLT to an ONU and used to assign a transmission timeslot. REPORT message is used by an ONU to convey its local conditions (such as buffer occupancy, etc.) to the OLT to help it make intelligent allocation decision. Both GATE and REPORT messages are MAC control frames (type 88-08) and are processed by the MAC control sub-layer.

There are two modes of operation of MPCP: auto-discovery (initialization) and normal operation. Auto-discovery mode is used to detect newly connected ONUs and learn the round-trip delay and MAC address of that ONU, plus maybe some additional parameters yet to be defined. Normal mode is used to assign transmission opportunities to all initialized ONUs.

Since more than one ONU can require initialization at one time, auto-discovery is a contention-based procedure. At a high level, it works as follows:

1. OLT allocates an initialization slot, an interval of time when no previously initialized ONUs are allowed to transmit. The length of this initialization slot must be at least *<transmission size>* + *< maximum round-trip time>* - *<minimum round-trip time>*, where *<transmission size>* is the length of the transmission window which an un-initialized ONU can use.
2. OLT sends an initialization GATE message advertising the start time of the initialization slot and its length. While relaying this message from a higher layer to the MAC layer, MPCP will timestamp it with its local time.
3. Only un-initialized ONUs will respond to the initialization GATE message. Upon receiving the initialization GATE message, an ONU will set its local time to the arriving timestamp in the initialization GATE message.
4. When the local clock located in the ONU reaches the start time of the initialization slot (also delivered in the GATE message), the ONU will transmit its own message (initialization REPORT). The REPORT message will contain the ONU's source

address and a timestamp representing local ONU's time when the REPORT message was sent.

5. When the OLT receives the REPORT from an un-initialized ONU, it learns its MAC address and round-trip time. As illustrated in Figure 8-8, the round-trip time of an ONU is exactly the time difference between the time the REPORT is received at the OLT and the timestamp contained in the REPORT.

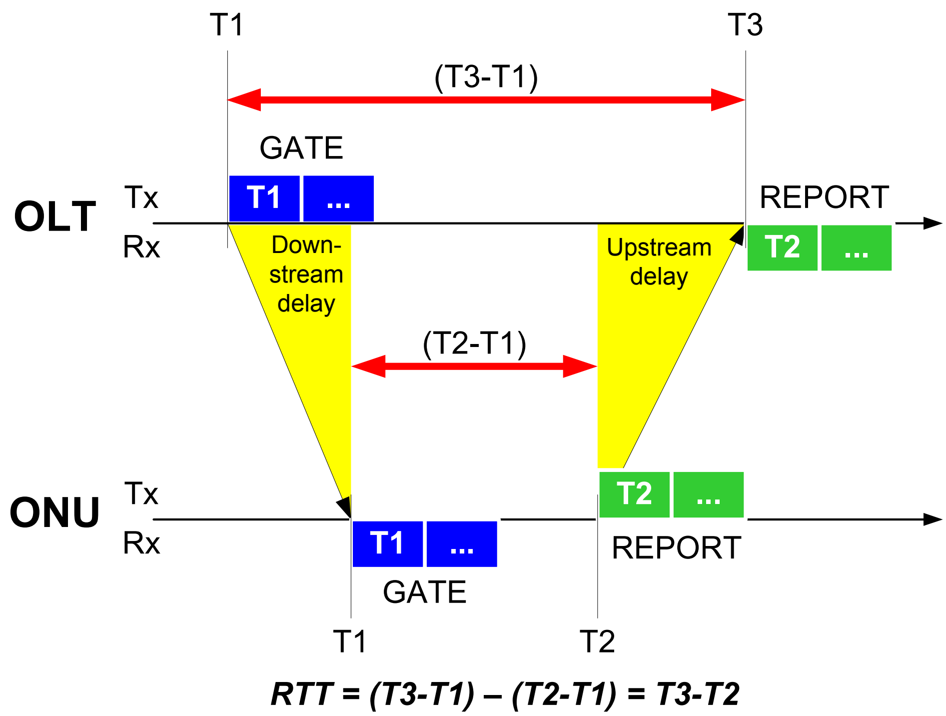


Figure 8-8. Round-trip time measurement.

Since multiple un-initialized ONUs may respond to the same initialization GATE message, the REPORT messages may collide. In that case, the ONUs whose REPORTs have collided will not get any slot assignments for their normal operation. If an ONU does not receive a slot assignment within some timeout interval, it will infer that a collision has occurred, and it will attempt to initialize again after skipping some random number of initialization GATE messages. The number of messages to skip is chosen randomly from an interval that doubles after each inferred collision, i.e., using exponential backoff.

Below we illustrate the normal operation of MPCP. It is important to notice that MPCP is not concerned with particular bandwidth-allocation schemes; rather it is a supporting protocol necessary to deliver these decisions from the OLT to the ONUs.

1. From its higher layer (MAC control client), MPCP gets a request to transmit a GATE message to a particular ONU with the following information: time when that ONU should start transmission and length of the transmission (Figure 8-9).

2. MPCP layer (in OLT and each ONU) maintains a clock. Upon passing a GATE message from its higher layer to MAC, MPCP timestamps it with its local time.

3. Upon receiving a GATE message matching that ONU's MAC address (GATE messages are unicast), the ONU will program its local registers with transmission start and transmission length times. The ONU will also verify that time when the GATE message arrived is close to the timestamp value contained within the message. If the difference in values exceeds some pre-defined threshold, the ONU will assume that it has lost its synchronization and will switch itself into un-initialized mode. In that mode, the ONU is not allowed to transmit. It will monitor its incoming traffic waiting for the next initialization GATE message to perform initialization.

4. If the time the GATE message is received is close to the timestamp value in the GATE message, the ONU will update its local clock to that of timestamp. When the local time reaches the 'start transmission' value, the ONU will start transmitting. That transmission may include multiple Ethernet frames. The ONU will ensure that no frames are fragmented. If the next frame does not fit in the remainder of the timeslot, it will be deferred till the next timeslot.

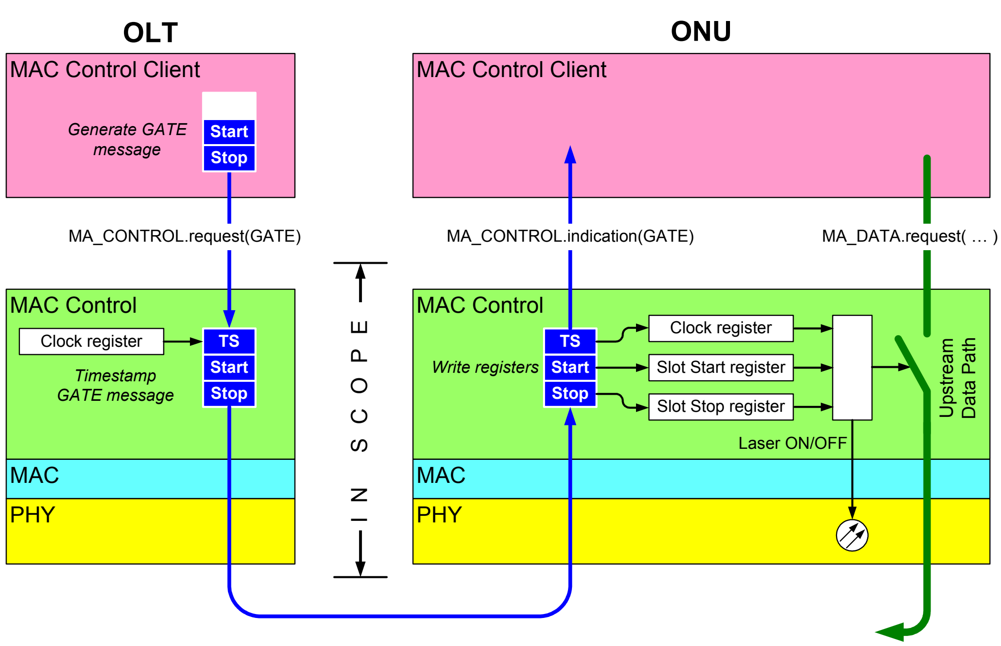


Figure 8-9. Multi-Point Control Protocol – GATE operation.

REPORT messages are sent by ONUs in the assigned transmission windows together with data frames. REPORT messages can be sent automatically or on-demand. A REPORT message is generated in the MAC control client layer and is time-stamped in the MAC control (Figure 8-10). Typically, REPORT would contain the desired size of next timeslot based on ONU's queue size. When requesting a timeslot, an ONU should account for additional overhead, namely 64-bit frame preamble and 96-bit IFG associated with every frame.

When a time-stamped REPORT message arrives at the OLT, it is passed to the MAC control client layer responsible for making the bandwidth-allocation decision. Additionally, the OLT will recalculate the round-trip time to the source ONU as shown in Figure 8-8. Some small deviation of the new RTT from the previously measured RTT may be caused by changes in fiber refractive index resulted from temperature drift. A large deviation should alarm the OLT about the ONU's potential mis-synchronization and should prevent the OLT from further granting any transmissions to that ONU until it is re-initialized.

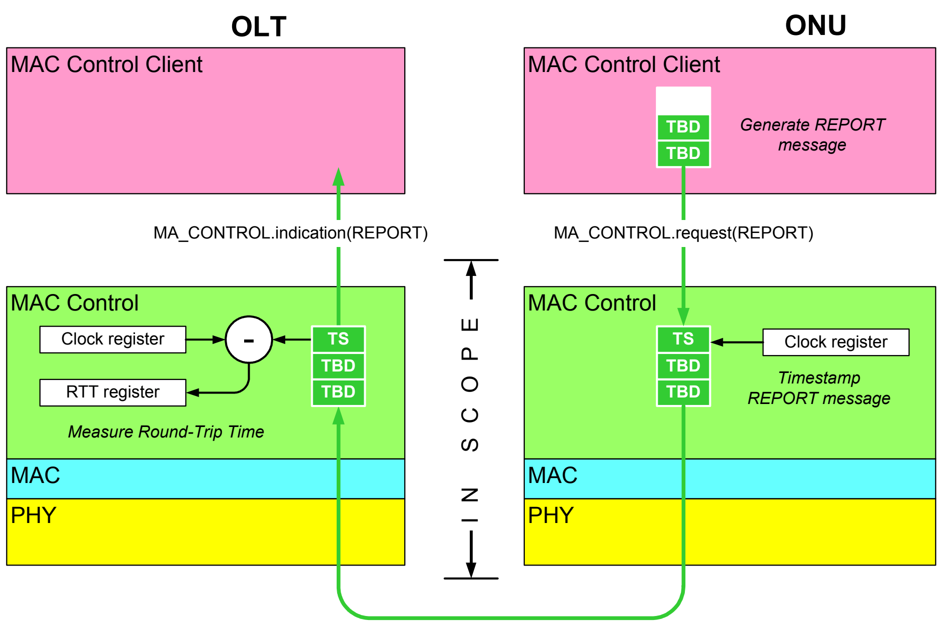


Figure 8-10. Multi-Point Control Protocol – REPORT operation.

The above description represents a framework of the protocol being developed for the EPON. There are many more details that remain to be discussed and agreed upon. This work is currently being conducted in the IEEE 802.3ah task force, a standards group charged with the development of the Ethernet solution for the subscriber access network.

8.3.4 EPON Compliance with 802 Architecture

The IEEE 802 architecture defines two types of media: shared medium and full duplex. In a shared medium, all stations are connected to a single access domain where at most one station can transmit at a time and all stations can receive all the time. The full-duplex segment is a point-to-point link connecting two stations (or a station and a bridge) such that both stations can transmit and receive simultaneously. Relying on the above definitions, bridges never forward a frame back to its ingress port. In other words, it is assumed that all the stations connected to the same port on the bridge can communicate with one another without the bridge's help. This bridge behavior has led to an interesting problem: users connected to a different ONUs on the same PON are unable to communicate with one another without data being processed at layer 3 (network layer) or above. This raises a question of compliance with IEEE 802 architecture, particularly with P802.1D bridging.

To resolve this issue and to ensure seamless integration with other Ethernet networks, devices attached to the EPON medium will have an additional sub-layer that, based on its configuration, will emulate either a shared medium or a point-to-point medium. This sub-layer is referred to as Shared-Medium Emulation (SME) or Point-to-Point Emulation (PtPE) sub-layer. This sub-layer must reside below the MAC layer to preserve the existing Ethernet MAC operation defined in the IEEE standard P802.3. Operation of the emulation layer relies on tagging of Ethernet frames with tags unique for each ONU (Figure 8-11). These tags are called "link ID" and are placed in the preamble before each frame.

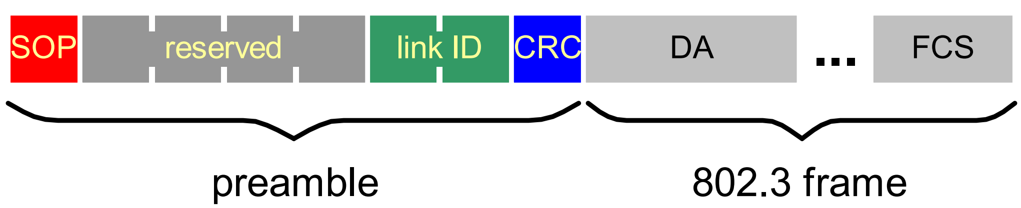


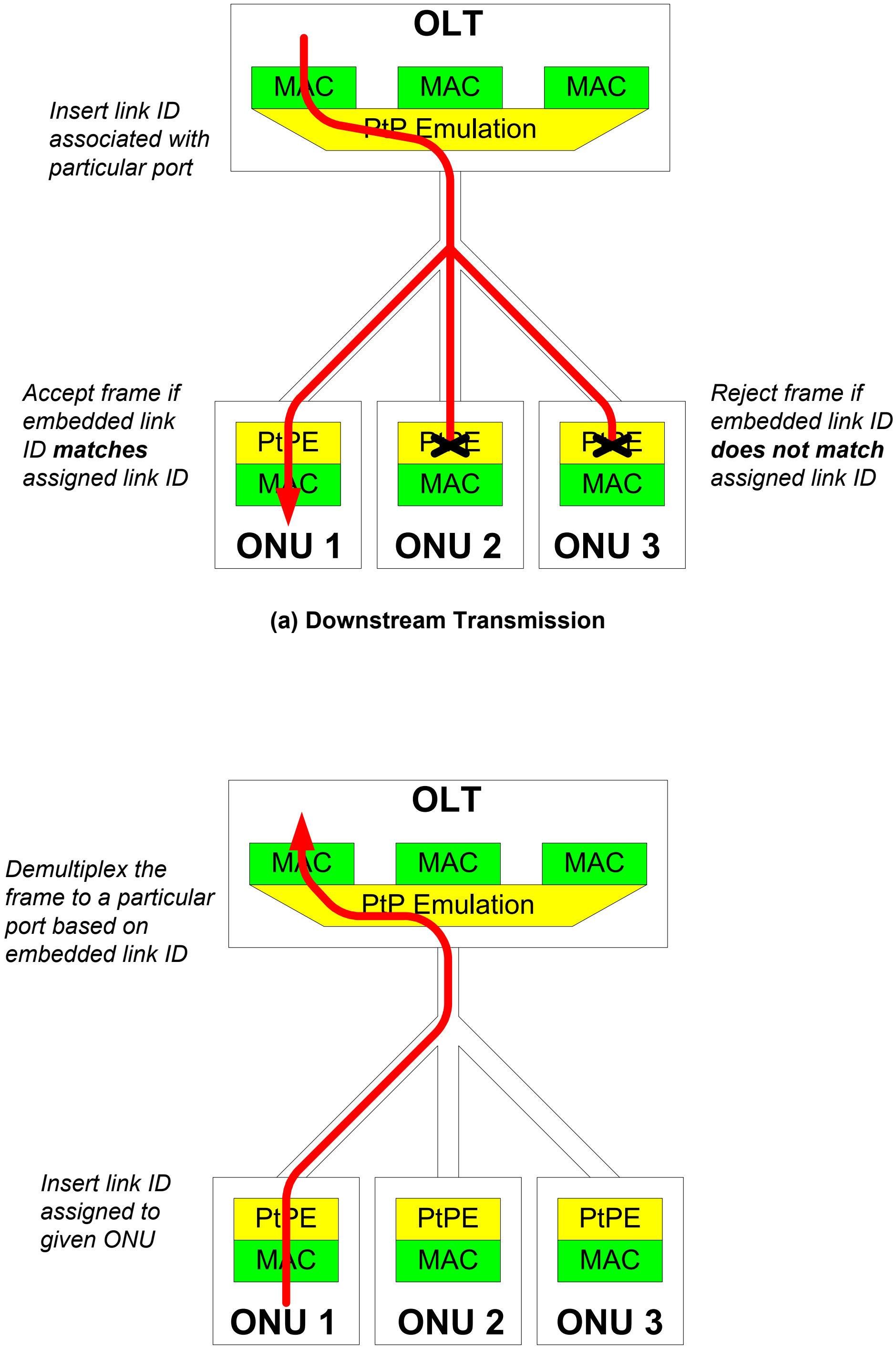
Figure 8-11. Link ID field embedded in frame preamble.

To guarantee uniqueness of link IDs, each ONU is assigned one or more tags by the OLT during initial registration phase.

8.3.4.1 Point-to-Point Emulation (PtPE)

In PtP emulation mode, the OLT must have *N* MAC ports (interfaces), one for each ONU (Figure 8-12). When sending a frame downstream (from the OLT to an ONU), the PtPE sub-layer in the OLT will insert the link ID associated with a particular MAC port that the frame arrived from (Figure 8-12.a). Even though the frame will be delivered to each ONU, only one PtPE sub-layer will match that frame's link ID with the value assigned to the ONU and will accept the frame and pass it to its MAC layer for further verification. MAC layers in all other ONUs will never see that frame. In this sense, it appears as if the frame was sent on a point-to-point link to only one ONU.

In the upstream direction, the ONU will insert its assigned link ID in the preamble of each transmitted frame. The PtPE sub-layer in the OLT will de-multiplex the frame to the proper MAC port based on the unique link ID (Figure 8-12.b).



(a) Downstream Transmission

(b) Upstream Transmission

Figure 8-12. Point-to-point emulation.

The PtPE configuration is clearly compatible with bridging as each ONU is connected to an independent bridge port. The bridge placed in the OLT (Figure 8-13) will relay inter-ONU traffic between its ports.

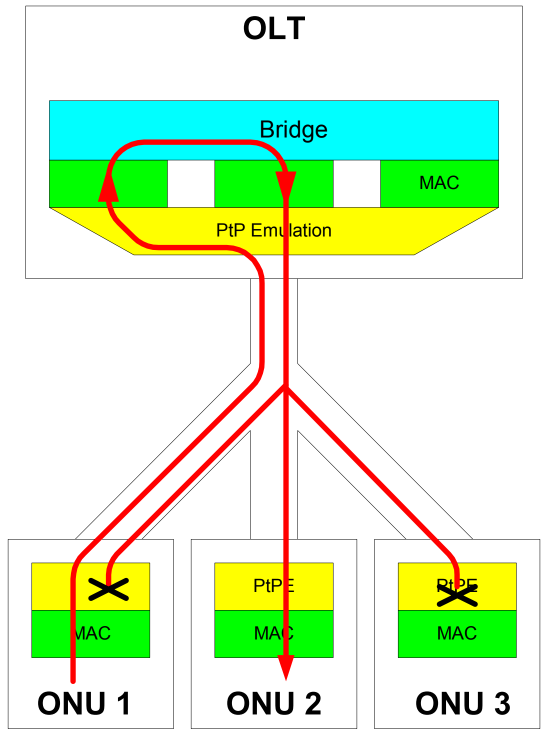


Figure 8-13. Bridging between ONUs with point-to-point emulation.

8.3.4.2 Shared-Medium Emulation (SME)

In shared-medium emulation, frames transmitted by *any* node (OLT or any ONU) should be received by *every* node (OLT and every ONU). In the downstream direction, the OLT will insert a "broadcast" link ID which will be accepted by every ONU (Figure 8-14.a). To ensure shared-medium operation for upstream data (frames sent by ONUs), the SME sub-layer in OLT must mirror all frames back downstream to be received by all other ONUs (Figure 8-14.b). To avoid frame duplication when an ONU receives its own frame, the SME sub-layer in an ONU accepts a frame only if the frame's link ID is different from the link ID assigned to that ONU.

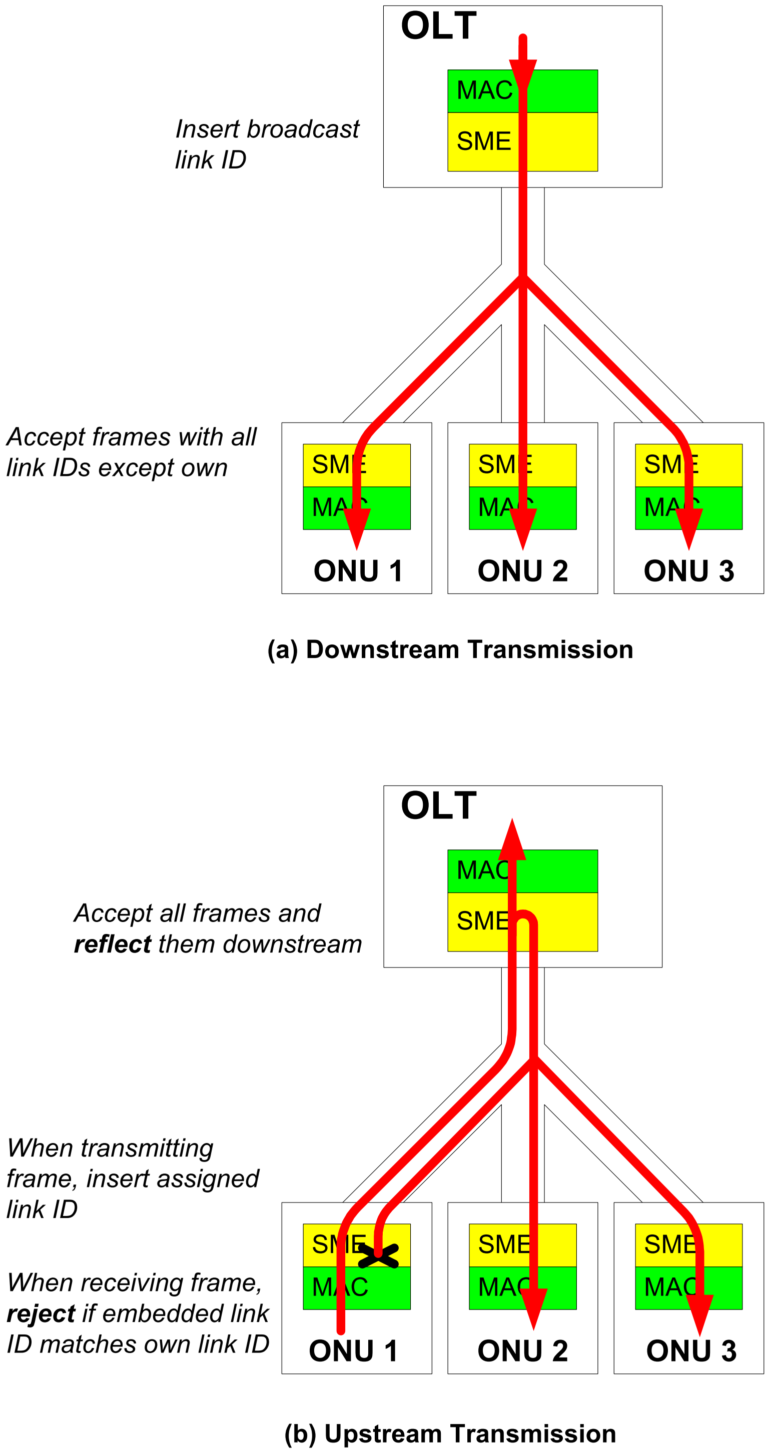


Figure 8-14. Shared-medium emulation.

The shared-medium emulation requires only one MAC port in the OLT. Physical-layer functionality (SME sub-layer) provides the ONU-to-ONU communicability, eliminating the need for a bridge.

8.3.4.3 Combined PtPE and SME Mode

While both PtPE and SME options provide solutions for P802.1 standards compliance issues, both of them also have drawbacks, specifically when considered for an application in a subscriber access networks. The PtPE mode precludes the possibility to have a single-copy multicast/broadcast when the OLT sends one frame received by several ONUs. This feature is very important for services such as video broadcast or any real-time broadcast services. To support such services, the OLT operating in the PtPE mode must duplicate broadcast packets, each time with a different link ID.

Shared-medium emulation, on the other hand, provides multicast/broadcast capabilities. However, because *every* upstream frame is reflected downstream, it wastes a lot of downstream bandwidth.

To achieve an optimal operation, it is feasible to deploy a PON with point-to-point and shared-medium emulation simultaneously. In such a configuration in an EPON with *N* ONUs, the OLT will contain *N+1* MACs: one for each ONU (PtPE) and one for broadcasting (Figure 8-15). Each ONU must have two MACs: one for shared medium and one for point-to-point emulated link. To optimally separate the traffic, higher layers (above MAC) will decide which port to send data to (e.g., by using VLANs). Only data that should be broadcast will be sent to the port connected to the emulated shared-medium segment.

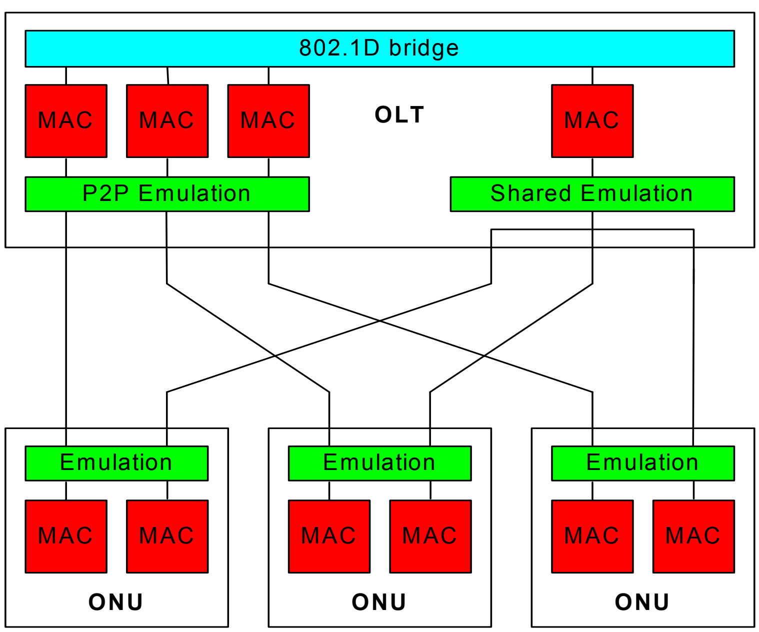


Figure 8-15. Combined point-to-point and shared-medium emulation mode.

8.3.4.4 Open Issues

The work on emulation sub-layer design is still in progress. A serious challenge that needs to be solved is that the emulation sub-layer must be able to multiplex several data flows into one flow. In PtPE mode, the emulation layer may receive data frames from multiple MAC ports simultaneously. In SME mode this happens when an ONU-to-ONU data frame competes with a network-to-ONU data frame for the downstream channel. The apparent drawback of this competition is that, now, some frames may have to be discarded below the MAC sub-layer, which may make the BER dependent on the traffic load. To drop frames intelligently, the emulation sub-layer should be aware of the sender's or the recipient's SLA and frame priority. All these features are strictly out-of-scope of the IEEE 802 standard and do not belong in PHY layer. Additionally, even if frames are not dropped in the emulation sub-layer, MAC-to-MAC delay may not be constant due to head-of-line blocking, which may have a detrimental effect on QoS.

An alternative proposal suggests putting the emulation sub-layer in the MAC control layer. In this case, the link ID information should transparently propagate through the MAC and this will require MAC modifications. Another, more subtle problem is that, since frame filtering is now performed above the MAC layer, in PtPE mode, every MAC will see all frames before they are filtered out based on link ID. This means that a corrupted and invalid frame will increment error counters in all MACs as opposed to only one MAC at the other end of its virtual PtP link. This, of course, will invalidate layer-management facilities provisioned by the standard.

Finding solutions for the above-mentioned issues, as well as converging on a best place for the emulation sub-layer, remains on the list of open issues for the IEEE 802.3ah task force.

8.4 Performance of EPON

The performance of an EPON depends on the particular bandwidth-allocation scheme. Choosing the best allocation scheme, however, is not a trivial task. If all users belong to the same administrative domain (say a corporate or campus network), full statistical multiplexing would make sense – network administrators would like to get most out of the available bandwidth. However, subscriber access networks are not private LANs and the objective is to ensure Service-Level Agreement (SLA) compliance for each individual user. Using statistical multiplexing mechanisms to give each user best effort-bandwidth may complicate billing and may potentially offset the user's drive to upgrade to a higher bandwidth. Also, subscribers may get used to and expect the performance that they get during low-activity hours when lots of best-effort bandwidth is available. Then, at peak hours, the same users would perceive the service as unsatisfactory, even though they get what is guaranteed by their SLA. An optimized bandwidth-allocation algorithm will ultimately depend on the future SLA and billing model used by the service provider.

This notion has led to a "fixed pipe" model for an access network. Fixed pipe assumes that each user will agree to and pay for a fixed bandwidth regardless of the network conditions or applications using it. Because the contracted bandwidth must be available at any time, this model does not support over-subscription. Correspondingly, network operators are not eager to give users an additional best-effort bandwidth. It is not easy to charge for and users are not willing to pay for what is hard to measure. In a sense, this model operates like a fixed circuit given to each customer.

Recently, however, there has been a shift to a new paradigm. Since bandwidth is getting cheaper, the revenues the service providers get from data traffic are decreasing. Correspondingly, many carriers complain that, to accommodate the increased traffic on their networks, they have to upgrade their networks often, and thus their capital expenses increase, but the revenue remains flat or even decreases. In recent years, it has become apparent that raw bandwidth cannot generate enough revenue. The new thinking among telecommunication operators calls for service-based billing in which users pay for the services they get, and not for the guaranteed bandwidth they are provisioned. In this model, the network operators are willing to employ statistical multiplexing to be able to support more services over the network.

Below we will compare the EPON performance operated in fixed TDMA ("fixed pipe") and statistical multiplexed modes.

8.4.1 Model Description

In this study, we consider an access network consisting of an OLT and *N* ONUs connected using a passive optical network (Figure 8-16). Every ONU is assigned a downstream propagation delay (from the OLT to the ONU) and an upstream propagation delay (from the ONU to the OLT.) While with a tree topology both downstream and upstream delays are the same, with a ring topology delays will be different. To keep the model general we assume independent delays and select them randomly (uniformly) over the interval [50 μs, 100 μs]. These values correspond to distances between the OLT and ONUs ranging from 10 to 20 km.

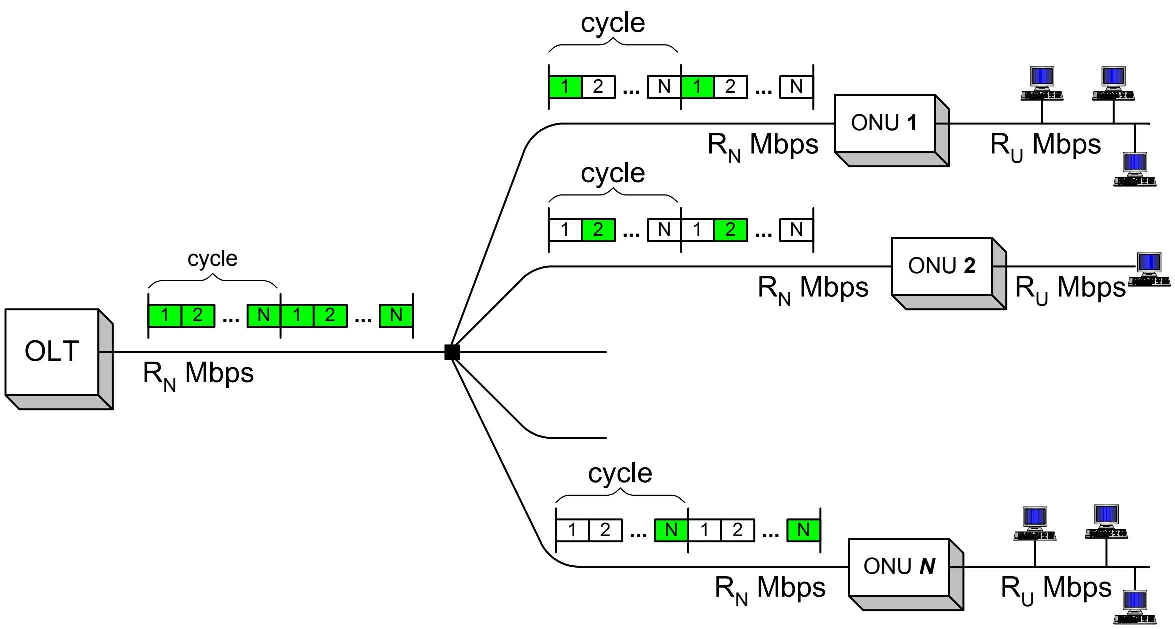


Figure 8-16. Simulation model of an EPON access network.

From the access side, traffic may arrive at an ONU from a single user or from a gateway of a LAN, i.e., traffic may be aggregated from a number of users. Ethernet frames should be buffered in the ONU until the ONU is allowed to transmit the packets. The transmission speed of the PON and the user access link may not necessarily be the same. In our model, we consider R_U Mbps to be the user data rate (rate of access link from a user to an ONU), and R_N Mbps to be the network data rate (upstream slotted link from an ONU to the OLT) (see Figure 8-16). We should mention here that, if $R_N \geq N \times R_U$, then the bandwidth utilization problem does not exist, as the system throughput is higher than the peak aggregated load from all ONUs. In this study, we consider a system with N=16 and R_U and R_N being 100 Mbps and 1000 Mbps, respectively.

A set of N timeslots together with their associated guard intervals is called a *cycle*. In other words, a cycle is a time interval between two successive timeslots assigned to one ONU (Figure 8-16). We denote cycle time by T. Making T too large will result in increased delay for all the packets, including high-priority (real-time) packets. Making T too small will result in more bandwidth being wasted by guard intervals.

To obtain an accurate and realistic performance analysis, it is important to simulate the system behavior with appropriate traffic injected into the system. There is an extensive study showing that most network traffic flows (i.e., generated by http, ftp, variable-bit-rate (VBR) video applications, etc.) can be characterized by self-similarity and long-range dependence (LRD) (see [12] for an extensive reference list). To generate self-similar traffic, we used the method described in [13], where the resulting traffic is an aggregation of multiple streams, each consisting of alternating Pareto-distributed ON/OFF periods.

Figure 8-17 illustrates the way the traffic was generated in an individual ONU. Within the ON period, every source generates packets back to back (with a 96-bit inter-frame gap and 64-bit preamble in between). Every source assigns a specific priority value to all its packets. Packets generated by n sources are aggregated (multiplexed) on a single line such that packets from different sources do not overlap. After that the packets are forwarded to the respective queues based on their priority assignments and the queues are served in order of their priorities.

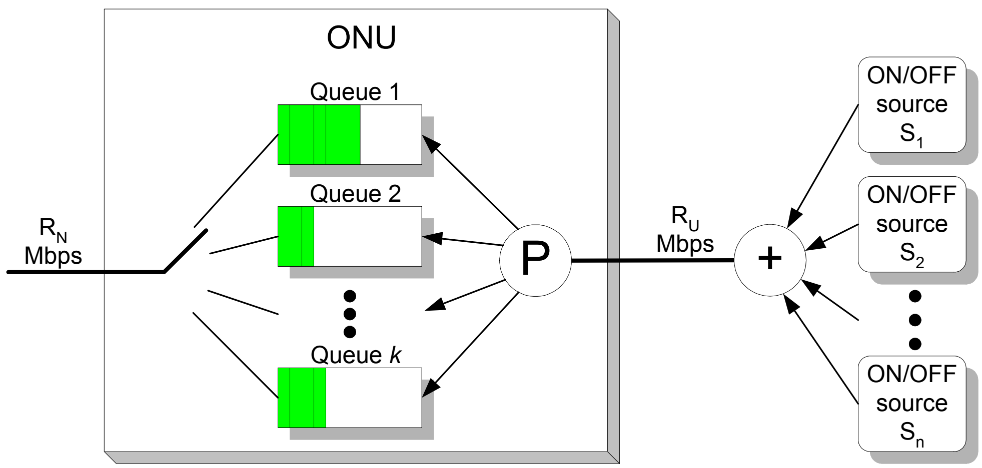


Figure 8-17. Traffic generation in the ONU.

Each ON/OFF source generates load

$$\tilde{\phi}_i = \frac{E[ON_i]}{E[ON_i] + E[OFF_i]} \qquad (1)$$

where $E[ON_i]$ and $E[OFF_i]$ are expected lengths (durations) of ON and OFF periods of source i. Load aggregated from all n sources in an ONU is called *offered ONU load* (OOL) and denoted by ϕ:

$$\phi = \sum_{i=1}^{n} \tilde{\phi}_i \qquad (2)$$

Offered network load (ONL) Φ is the sum of the loads offered by each ONU and scaled based on R_D and R_U rates. Clearly, since the network throughput is less than the aggregated peak bandwidth from all ONUs, the ONL can exceed 1:

$$\Phi = \frac{R_U}{R_N} \sum_{j=1}^{N} \phi^{[j]} \tag{3}$$

It is important to differentiate between offered load and *effective load*. The effective ONU load (EOL) is denoted ϖ and results from the data (packets) that have been sent out by the ONUs. Thus, the EOL is equal to the OOL only if the packet loss rate is zero. In general, $\varpi \le \phi$.

The EOL generated by the ONU j is denoted $\varpi^{[j]}$. *Effective network load* (ENL) Ω is just a sum of the EOLs generated by all ONUs with a corresponding scaling coefficient based on the PON and user link bit rates:

$$\Omega = \frac{R_U}{R_N} \sum_{j=1}^{N} \varpi^{[j]} \tag{4}$$

Every ONU may have k queues which are served in order of their priority (priority queuing is discussed in Section 8.5.3.1). Every ONU has a finite buffer of size Q. The memory is allocated to different queues based on demand and priority, i.e., if the entire buffer is occupied and a frame with a higher priority arrives, the lowest-priority non-empty queue will drop one or more frames, so that the higher-priority queue can store the new packet. In our simulations, buffer size Q was set to 10 Mbytes.

8.4.2 Bandwidth-Allocation Schemes

The essence of the MPCP protocol is in assigning a variable-sized slot (transmission window) to each ONU based on decisions made by some bandwidth-allocation scheme. To prevent the upstream channel being monopolized by one ONU with high data volume, there should be a maximum transmission window size limit assigned to every ONU. We denote an ONU-specific maximum transmission window size by $W_{MAX}^{[i]}$ (in bytes). The choice of specific values of $W_{MAX}^{[i]}$ determines the maximum granting cycle time T_{MAX} under heavy load conditions:

$$T_{MAX} = \sum_{i=1}^{N} \left(G + \frac{8 \times W_{MAX}^{[i]}}{R} \right) \tag{5}$$

where $W_{MAX}^{[i]}$ - maximum window size for i^{th} ONU (in bytes), G – guard interval (seconds), N – number of ONUs, and R – line rate (bps).

The guard intervals provide protection for fluctuations of round-trip time of different ONUs. Additionally, the OLT receiver needs some time to readjust its sensitivity due to the fact that signals from different ONUs may have different power levels (near-far problem).

Making T_{MAX} too large will result in increased delay for all Ethernet frames, including those carrying high-priority (real-time) IP packets. Making T_{MAX} too small will result in more bandwidth being wasted by guard intervals.

It is the ONU's responsibility to ensure that the frame it is about to send fits in the remainder of the timeslot. If the frame does not fit, it should be deferred till the next timeslot, leaving the current timeslot underutilized (not filled completely with Ethernet frames). Section 8.5.1 investigates the timeslot utilization issues in more details.

In addition to the maximum cycle time, the $W_{MAX}^{[i]}$ value also determines the guaranteed bandwidth available to ONU i. Let $\Lambda_{MIN}^{[i]}$ denote the (minimum) guaranteed bandwidth of ONU i (in bps). Obviously,

$$\Lambda_{MIN}^{[i]} = \frac{8 \times W_{MAX}^{[i]}}{T_{MAX}} \tag{6}$$

i.e., the ONU is guaranteed to be able to send at least $W_{MAX}^{[i]}$ bytes (or $8 \times W_{MAX}^{[i]}$ bits) in at most T_{MAX} time. Of course, an ONUs bandwidth will be limited to its guaranteed bandwidth only if all other ONUs in the system also use all of their available bandwidth. If at least one ONU has less data, it will be granted a shorter transmission window, thus making the granting cycle time shorter, and therefore the available bandwidth to all other ONUs will increase proportionally to their $W_{MAX}^{[i]}$. This is the mechanism behind dynamic bandwidth distribution described in [15]: by adapting the cycle time to the instantaneous network load (i.e., queue occupancy), the bandwidth is automatically distributed to ONUs based on their loads. In the extreme case, when only one ONU has data to send, the bandwidth available to that ONU will be:

$$\Lambda_{MAX}^{[i]} = \frac{8 \times W_{MAX}^{[i]}}{N \times G + \frac{8 \times W_{MAX}^{[i]}}{R}} \tag{7}$$

In our simulations, we assume that all ONUs have the same guaranteed bandwidth, i.e., $W_{MAX}^{[i]} = W_{MAX}$, $\forall i$. This results in

$$T_{MAX} = N\left(G + \frac{8 \times W_{MAX}}{R}\right) \tag{8}$$

We believe T_{MAX} = 2 ms and G = 5 μs are reasonable choices. They make W_{MAX} = 15,000 bytes. With these choices of parameters, every ONU will get a guaranteed bandwidth of 60 Mbps, and maximum (best-effort) bandwidth of 600 Mbps (see Equations 6 and 7).

The following algorithm was considered in our simulation study.

`RTT[i]`	–	table containing round-trip times for each ONU. This table is originally populated during auto-discovery phase. It is updated constantly during normal operation.
`V`	–	size of transmission window (in bytes) requested by ONU
`W`	–	size of transmission window (in bytes) granted to ONU
`ch_avail`	–	time when channel becomes available for next transmission
`local_time`	–	read-only register containing OLT's local clock values
`guard`	–	guard interval (constant)
`delta`	–	time interval to process GATE message in the ONU (minimum time between GATE arrival and beginning of timeslot)

```
ch_avail = 0;

repeat forever
{
      FOR i from 1 to N
      {
            /* wait for REPORT from ONU i */
            until(REPORT from ONU i arrived)
                  /* do nothing */;


            /* get round-trip time */
            RTT[i] = local_time – REPORT.timestamp

            /*get requested slot size */
            V = REPORT.slot_size

            /* update channel availability time
               to make sure we don't schedule slot
               for the past time */
            if(ch_avail < local_time + delta + RTT[i])
                  ch_avail = local_time + delta + RTT[i]

            /* make timeslot allocation decision
               (specific allocation schemes are
               considered below)*/
            W = f(V)

            /* create GATE message */
```

```
            GATE.slot_start = ch_avail - RTT[i]
            GATE.slot_size  = W

            /* update channel availability time for next ONU*/
            ch_avail = ch_avail + guard + time(W)

            /* send GATE message to ONU i */
            send(i, GATE)
        }
    }
```

The remaining question is how the OLT should determine the granted window size if the requested window size is less than the predefined maximum ($W^{[i]} < W_{MAX}$)? Table 8-1 defines a few approaches (services) the OLT may take in making its decision.

Service	Formula	Description
Fixed	$W^{[i]} = W_{MAX}$	This scheduling discipline ignores the requested window size and always grants the maximum window. As a result, it has a constant cycle time T_{MAX}. Essentially, this is a "fixed pipe" model and corresponds to the fixed TDMA PON system described in [14].
Limited	$W^{[i]} = MIN\begin{cases} V^{[i]} \\ W_{MAX} \end{cases}$ $V^{[i]}$ = requested window size	This discipline grants the requested number of bytes, but no more than W_{MAX}. It is the most conservative scheme and has the shortest cycle of all the schemes.

Gated	$W^{[i]} = V^{[i]}$	This service discipline does not impose the W_{MAX} limit on the granted window size, i.e., it will always authorize an ONU to send as much data as it has requested. Of course, without any limiting parameter, the cycle time may increase unboundedly if the offered load exceeds the network throughput. In this discipline, such a limiting factor is the buffer size Q, i.e., an ONU cannot store more than Q bytes, and thus, it will never request more than Q bytes.
Constant Credit	$W^{[i]} = MIN\begin{cases} V^{[i]} + Const \\ W_{MAX} \end{cases}$	This scheme adds a constant credit to the requested window size. The idea behind adding the credit is the following: assume x bytes arrived between the time when an ONU sent a REPORT message and the beginning of the timeslot assigned to the ONU as the result of processing the REPORT. If the granted window size equals requested window + x (i.e., it has a credit of size x), then these x bytes will experience smaller delay, and thus the average delay will reduce.
Linear Credit	$W^{[i]} = MIN\begin{cases} V^{[i]} \times Const \\ W_{MAX} \end{cases}$	This scheme uses a similar approach as the Constant Credit scheme. However, the size of the credit is proportional to the requested window. The reasoning here is the following: LRD traffic possesses a certain degree of predictability (see [16]), viz., if we observe a long burst of data, then this burst is likely to continue for some time into the future. Correspondingly, the arrival of more data during the last cycle time may signal that we are observing a burst of packets.

Elastic	$W^{[i]} = MIN\begin{cases} V^{[i]} \\ NW_{MAX} - \sum_{j=i-N}^{i-1} W^{[j]} \end{cases}$	Elastic service is an attempt to get rid of a fixed maximum window limit. The only limiting factor is the maximum cycle time T_{MAX}. The maximum window is granted in such a way that the accumulated size of last N grants (including the one being granted) does not exceed NW_{MAX} bytes (N = number of ONUs). Thus, if only one ONU has data to send, it may get a Grant of size up to NW_{MAX}.

Table 8-1. Grant scheduling services used in simulation.

8.4.3 Simulation results

First, let us take a look at the components of the packet delay (Figure 8-18).

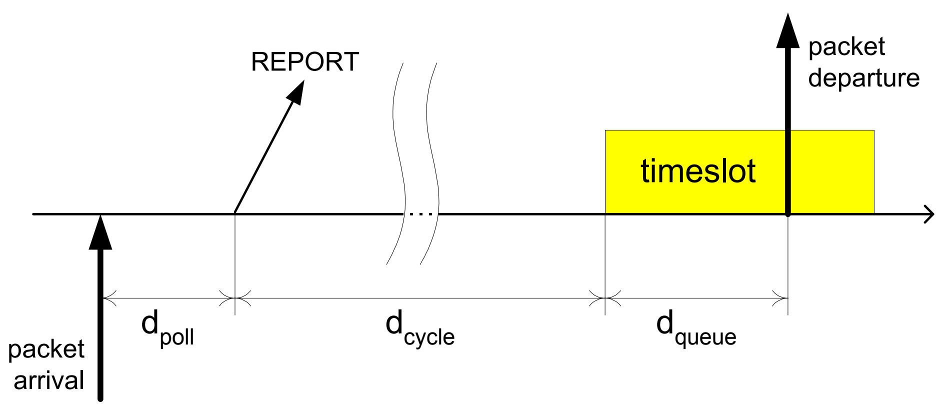


Figure 8-18. Components of packet delay.

The packet delay d is equal to:

$$d = d_{poll} + d_{cycle} + d_{queue} \quad (8)$$

where

d_{poll} = time between packet arrival and next REPORT sent by that ONU. On average this delay equals half of the cycle time.

d_{cycle} = time interval from ONU's request for a transmission window till the beginning of the timeslot in which this frame it to be transmitted. This delay may span multiple cycles (i.e., a frame may have to skip several timeslots before it reaches the head of the queue), depending on how many frames there were in the queue at the time of the new arrival.

d_{queue} = delay from the beginning of the timeslot till the beginning of frame transmission. In average, this delay is equal to half of slot time and is insignificant comparing to the previous two.

Figure 8-19 illustrates the mean packet delay for different timeslot allocation services as a function of an ONU's offered load ϕ. In this simulation, all ONUs have identical load; therefore, the offered network load Φ is equal $N\phi$.

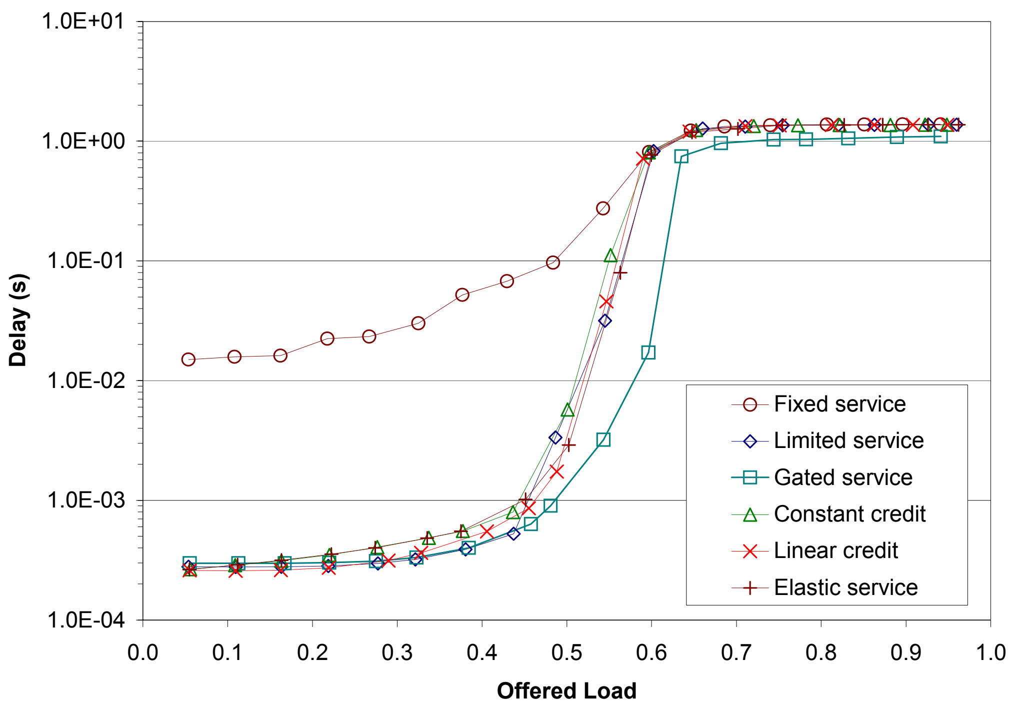


Figure 8-19. Mean packet delay.

As can be seen in the figure, all granting services except fixed and gated have almost coinciding plots. We will discuss fixed and gated service results below. As for the rest of the schemes, no other method gives a detectable improvement in packet delay. The explanation to this lies in the fact that all these methods are trying to send more data by way of increasing the granted window size. While that may result in a decrease or elimination of the d_{cycle} delay component for some packets, it will increase the cycle time, and thus result in an increase of the d_{poll} component for all the packets.

The fixed service plot is interesting as an illustration of the LRD traffic. Even at the very light load of 5%, the average packet delay is already quite high (~15ms). This is because most packets arrive in very large packet trains. In fact, the packet trains were so large that the 10-Mbyte buffers overflowed and about 0.14% of packets were dropped. Why do we observe this anomalous behavior only with fixed service? The reason is that the other services have a much shorter cycle time; there is just not enough time in a cycle to receive more bytes than W_{MAX}, thus the queue never builds up. In fixed service, on the other hand, the cycle is large (fixed) regardless of the network load and several bursts that arrived close to each other can easily overflow the buffer.

Before we continue with our discussion of gated service, we would like to present the simulation results for the average queue size (Figure 8-20). This picture is similar to the mean delay plot: fixed service has a larger queue due to larger cycle time.

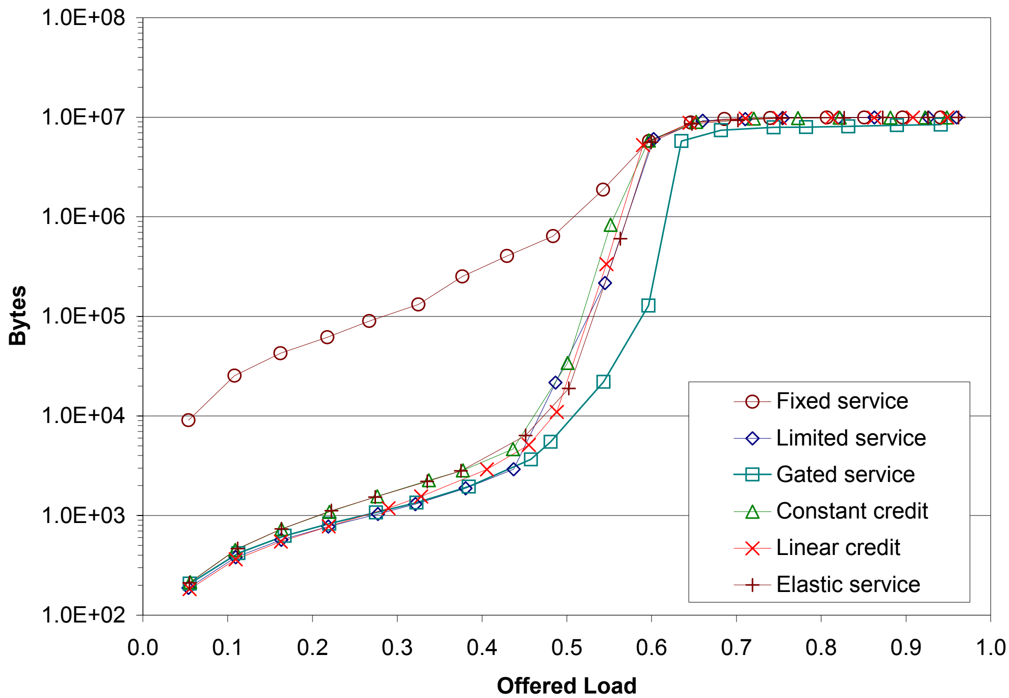


Figure 8-20. Average queue size.

Let us now turn our attention to the delay and queue size plots for gated service. It can be noticed that gated service provides a considerable improvement in the mid-range load between 45% and 65%. At 60% load, for example, the delay and average queue size are approximately 40 times less than with other services. This happens because gated service has higher channel utilization due to the fact that the cycle time is much larger, and, as a result, fewer guard intervals are used per unit of time. For the same reason, its saturation delay is a little bit lower than in other services (refer to Figure 8-19) – the entire buffer contents are being transferred in one jumbo transmission rather then in batches of W_{MAX} bytes with a guard time in front of each batch.

Next, we will show that, even though gated service has lower delay and average queue size, it is not a suitable service for an access network under consideration. The problem lies in the much longer cycle time (see Figure 8-21). As a result, the d_{poll} delay will be much larger, and therefore, the packet latency will be much higher. Clearly, large d_{cycle} and d_{queue} delay components can be avoided for high-priority packets by using priority queuing. But d_{poll} is a fundamental delay, which cannot be avoided in general. This makes gated service not feasible for access network.

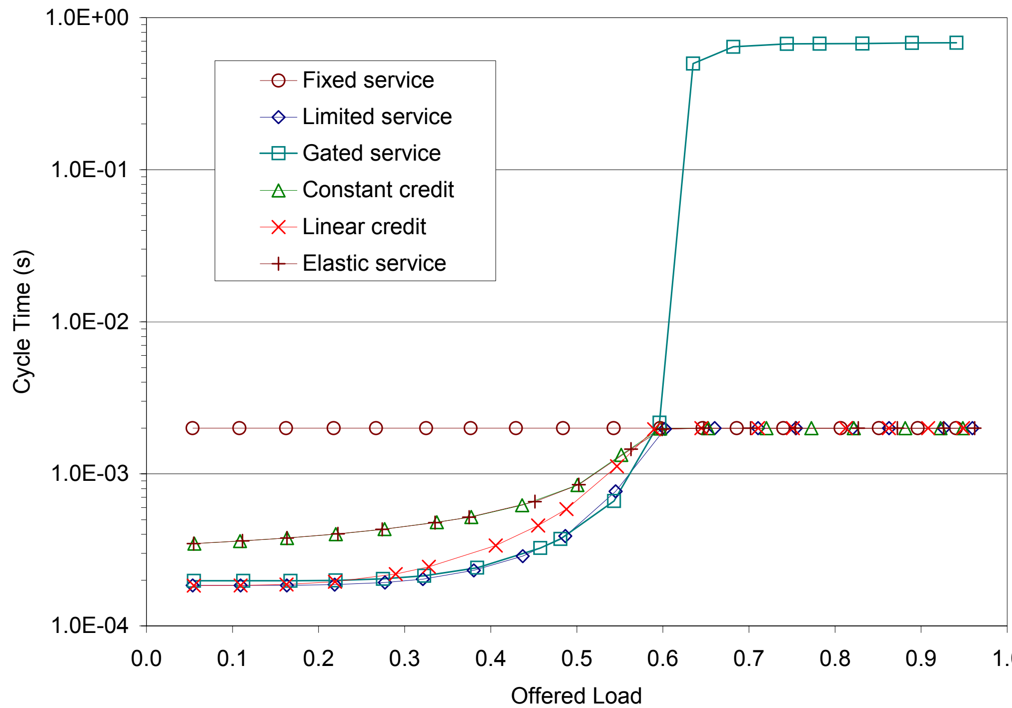


Figure 8-21. Mean cycle times for various service disciplines.

Thus, we conclude that neither of the discussed service disciplines is better than limited service, which is most conservative of all. As such, for the remainder of this study, we will focus our attention on the limited service discipline. In the next section, we will analyze the fairness and QoS characteristics of limited service.

8.4.4 Performance of Limited Service

In this section, we analyze the performance of one ONU (called tagged ONU) while varying its offered load (ϕ_i) independently of its ambient load (effective load Ω generated by the rest of the ONUs). In Figure 8-22, we present the average packet delay. All system parameters remained the same as in the previous simulation.

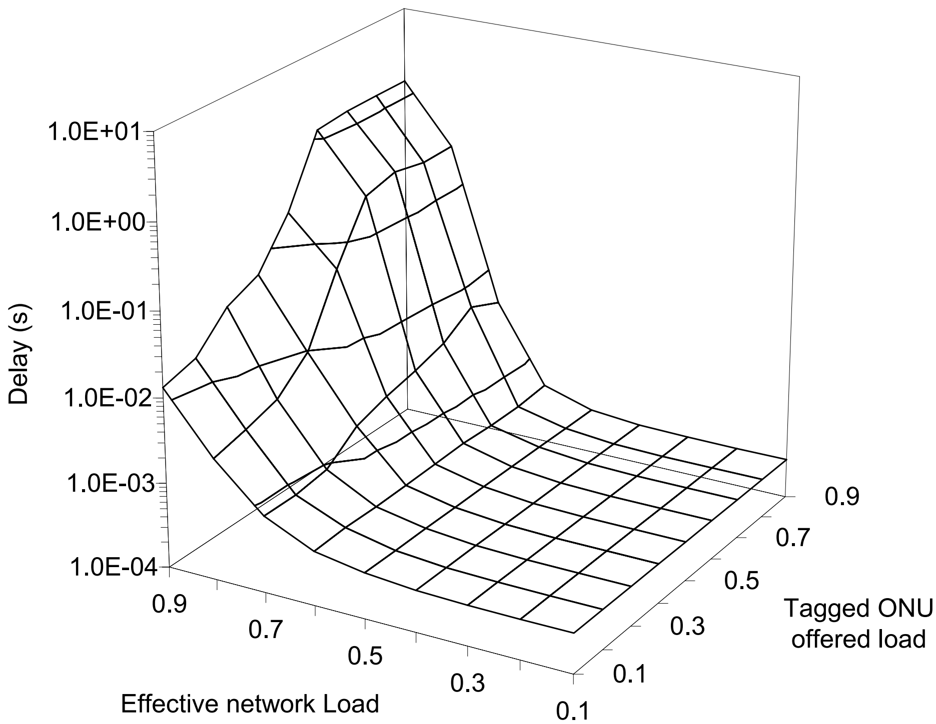


Figure 8-22. Average packet delay as a function of effective network load and ONU offered load.

When the effective network load is low, all packets in a tagged source experience very little delay, no matter what the ONU's offered load is. This is a manifestation of dynamic bandwidth allocation – when the network load is low, the tagged source gets more bandwidth.

The opposite situation – low offered load at the ONU and high effective network load – results in a higher delay. The only reason to this is the burstiness (i.e., long-range dependence) of the traffic. This is the same phenomenon observed with fixed service: high network load results in increased cycle time. This cycle time is large enough to receive more than W_{MAX} bytes of data during a burst. Hence, the d_{cycle} delay for some packets will increase beyond one cycle time. We will discuss a way to combat this phenomenon by using priority queuing in Section 8.5.3.

Figure 8-23 shows the probability of a packet loss (due to buffer overflow) in a tagged ONU *i* as a function of its offered load (ϕ_i) and the effective load of the entire network (Ω). The buffer size *Q* was set to 10 Mbytes as in the previous simulations. Once again we observe that the packet-loss ratio is zero or negligible if the effective network load is less than 80%. When the network load is above 80% and the tagged ONU offered load is above 50%, we observe considerable packet loss due to buffer overflow.

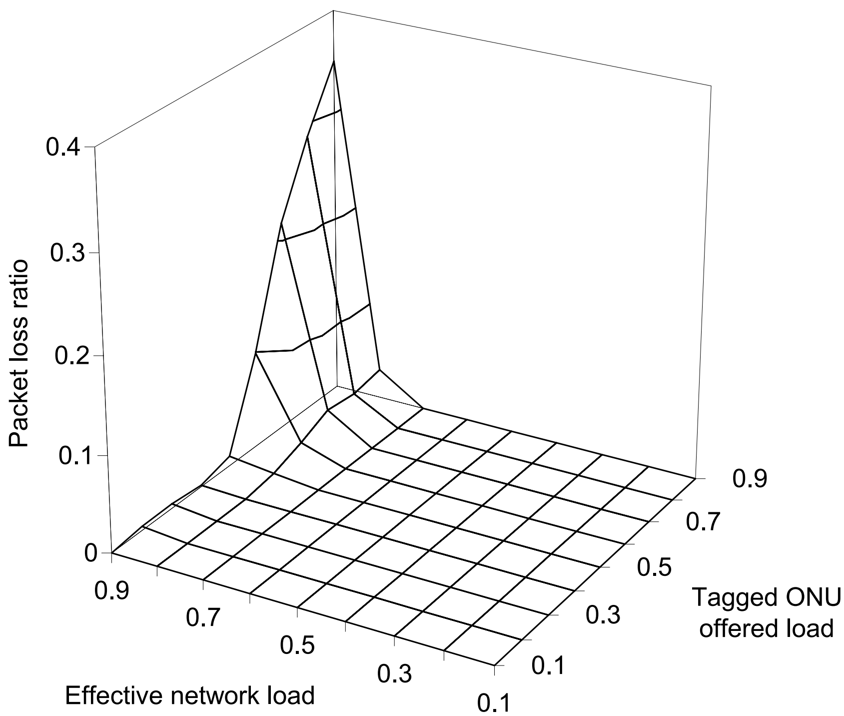


Figure 8-23. Packet-loss ratio as a function of effective network load and ONU offered load.

In the above simulations all traffic was treated as belonging to only one class (i.e., all frames having the same priority). In Section 8.5.3.1 we discuss EPON performance for multiple classes of service.

8.5 Considerations for IP-Based Services over EPON

The driving force behind extending Ethernet into the subscriber access area is Ethernet's efficiency for delivering IP packets. Data and telecom network convergence will lead to more and more telecommunication services migrating to a variable-length packet-based data networks. To ensure successful convergence, this migration should be accompanied by implementation of specific mechanisms traditionally available in telecom networks only.

Being designed with IP layer in mind, EPON is expected to seamlessly operate with IP-based traffic flows, similarly to any switched Ethernet network. One distinction with the typical switched architecture is that in an EPON, the user's throughput is slotted (gated), i.e., packets cannot be transmitted by an ONU at any time. This feature results in two issues unique to EPONs: (a) slot utilization by variable-length packets and (b) slot scheduling to support real-time and controlled-load traffic classes.

8.5.1 Slot Utilization Problem

The slot utilization problem is related to the fact that Ethernet frames cannot be fragmented and as a result variable-length packets don't fill the given slot completely. This problem manifests itself in a fixed service when slots of constant size are given to an ONU regardless of its queue occupancy. Slots may not be filled to capacity also in the case when the OLT grants to an ONU a slot smaller than the ONU requested based on its queue size. The fact that there is an unused remainder at the end of the slot means that the user's throughput is less than the bandwidth given to the user by a network operator in accordance with a particular SLA. Figure 8-24 presents slot utilization for packet traces obtained on an Ethernet LAN in Bellcore [17]. Increasing the slot size improves the utilization, i.e., the user's throughput approaches the bandwidth assigned by the operator; however it has detrimental effects on the data latency as larger slots increase the overall cycle time.

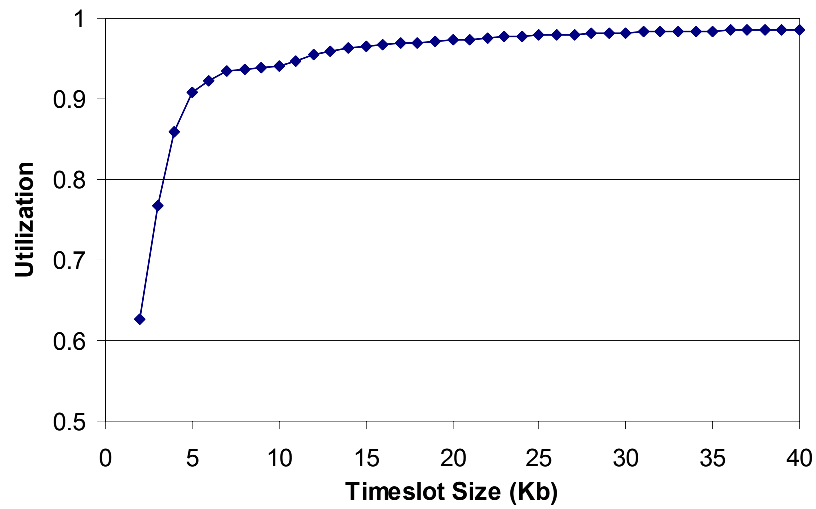


Figure 8-24. Slot utilization for various slot sizes.

Slot utilization can also be improved by employing smarter packet scheduling (e.g., the bin-packing problem). Rather than stopping the transmission when the head-of-the-queue frame exceeds the remainder of the slot, the algorithm may look ahead in the buffer and pick a smaller packet for immediate transmission (first-fit scheduling). However, as it turns out, first-fit scheduling is not such a good approach. To understand the problem, we need to look at the effects of packets reordering from the perspective of TCP/IP payload carried by Ethernet frames. Even though TCP will restore the proper sequence of packets, an excessive reordering may have the following consequences:

1. According to the fast retransmission protocol, the TCP receiver will send an immediate ACK for any out-of-order packet, whereas for in-order packets, it may generate a cumulative acknowledgement (typically for every other packet) [18]. This will lead to more unnecessary packets being placed in the network.
2. Second, and more importantly, packet reordering in ONUs may result in a situation where *n* later packets are being transmitted before an earlier packet. This would generate *n* ACKs (*n-1* duplicate ACKs) for the earlier packet. If *n* exceeds a predefined threshold, it will trigger packet retransmission and reduction of the TCP's congestion window size (the *cwnd* parameter). Currently, the threshold value in most TCP/IP protocol stacks is set to 3 (refer to the Fast Retransmission Protocol in [18] or elsewhere).

Even if special care is taken at the ONU to limit out-of-order packets to only 1 or 2, the rest of the end-to-end path may contribute additional reordering. While true reordering typically generates less than 3 duplicate ACKs and is ignored by the TCP sender, together with reordering introduced by the ONU, the number of duplicate ACKs may exceed 3, thus forcing the sender to retransmit a packet. As a result, the overall throughput of the user's data may decrease.

So, what is the solution? It is reasonable to assume that the traffic entering the ONU is an aggregate of multiple flows. In the case of business users, it would be the aggregated flows from multiple workstations. In the case of a residential network, we still may expect multiple connections at the same time. This is because, as a converged access network, PON will carry not only data, but also voice-over-IP (VoIP) and video traffic. Also, home appliances are becoming network plug-and-play devices. The conclusion is that, if we have multiple connections, we can reorder packets that belong to different connections, and never reorder them if they belong to the same connection. Connections can be distinguished by examining the source/destination address pairs and source/destination port numbers. This will require an ONU to look up layer-3 and layer-4 information in the packets. Thus, the important tradeoff decision that EPON designers have to make is whether it makes sense to considerably increase the required processing power in an ONU to improve the bandwidth utilization.

8.5.2 Circuit Emulation (TDM over IP)

The migration of TDM circuit-switched networks to IP packet-switched networks is progressing at a rapid pace. But even though the next-generation access network will be optimized for IP data traffic, legacy equipment (RF set-top boxes, analog TV sets, TDM private branch exchanges (PBXs), etc.) and legacy services (T1/E1, Integrated Services Digital Network (ISDN), Plain Old Telephone Service (POTS), etc) will remain in use in the foreseeable future. Therefore, it is critical for next-generation access networks, such as Ethernet PONs, to be able to provide both IP-based services and jitter-sensitive and time-critical legacy services that have traditionally not been the focus of Ethernet.

The issue in implementing a circuit-over-packet emulation scheme is mostly related to clock distribution. In one scheme, users provide a clock to their respective ONUs, which in turn is delivered to the OLT. But, since the ONUs cannot transmit all the time, the clock information must be delivered in packets. The OLT will regenerate the clock using this information. It is somewhat trivial to impose a constraint that the OLT should be a clock master for all downstream ONU devices. In this scenario, an ONU will recover the clock from its receive channel, use it in its transmit channel, and distribute it to all legacy devices connected to it.

8.5.3 Real-Time Video and Voice Over IP

Performance of a packet-based network can be conveniently characterized by several parameters: bandwidth, packet delay (latency) and delay variation (jitter), and packet-loss ratio. Quality of Service (QoS) refers to a network's ability to provide bounds on some or all of these parameters. It is useful to further differentiate statistical QoS from guaranteed QoS. Statistical QoS refers to a case when parameters can exceed the specified bounds with some small probability. Correspondingly, guaranteed QoS refers to a network architecture where parameters are guaranteed to stay within the specified bounds for the entire duration of a connection. A network is required to provide QoS (i.e., bounds on performance parameters) to ensure proper operation of real-time services such as video-over-packets (digital video conferencing, VoD), voice-over-IP (VoIP), real-time transactions, etc. To be able to guarantee QoS for higher-layer services, QoS must be maintained in all traversed network segments, including the access network portion of the end-to-end path. This section only focuses on QoS in the EPON access network.

The original Ethernet standard based on the CSMA/CD MAC protocol was never concerned with QoS. All connections (traffic flows) were treated equally and were given best-effort service from the network. The first major step in allowing QoS in the Ethernet was an introduction of the full-duplex mode. Full duplex MAC (otherwise called null-MAC) can transmit data frames at any time; this eliminated non-deterministic delay in accessing the medium. In a full duplex link (segment), once a packet is given to a transmitting MAC layer, its delay, jitter, and loss probability are known or predictable all the way to the receiving MAC layer. Delay and jitter may be affected by head of line blocking when the MAC port is busy transmitting the previous frame at the time when the next one becomes available. However, with 1-Gbps channel, this delay variation becomes negligible since the maximum-sized Ethernet frame is transmitted in only about 12 μs. It is important to note that the full-duplex MAC does not make the Ethernet a QoS-capable network: switches located in junction points still may provide non-deterministic, best-effort services.

The next step in enabling QoS in Ethernet was brought by introduction of two new standards extensions: P802.1p "Supplement to MAC Bridges: Traffic Class Expediting and Dynamic Multicast Filtering" (later merged with P802.1D) and P802.1Q "Virtual Bridged Local Area Networks". P802.1Q defines a frame format extension allowing Ethernet frames to carry priority information. P802.1p specifies the default bridge (switch) behavior for different priority classes; specifically, it allows a queue in a bridge to be serviced only when all higher-priority queues are empty. The standard distinguishes the following traffic classes:

1. *Network Control*—characterized by a "must get there" requirement to maintain and support the network infrastructure.
2. "*Voice*"—characterized by less than 10 ms delay, and hence maximum jitter (one way transmission through the LAN infrastructure of a single campus).
3. "*Video*"—characterized by less than 100 ms delay.
4. *Controlled Load*—important business applications subject to some form of "admission control," be that pre-planning of the network requirement at one extreme to bandwidth reservation per flow at the time the flow is started at the other.
5. *Excellent Effort*—or "CEO's best effort," the best-effort type services that an information services organization would deliver to its most important customers.
6. *Best Effort*—LAN traffic as we know it today.
7. *Backgroun*d—bulk transfers and other activities that are permitted on the network but that should not impact the use of the network by other users and applications.

If a bridge or a switch has less than 7 queues, some of the traffic classes are grouped together. Table 8-2 illustrates the standards-recommended grouping of traffic classes.

Number of queues	Groups of traffic types
1	Network Control, Voice, Video, Controlled Load, Excellent Effort, Best Effort, Background
2	Network Control, Voice, Video, Controlled Load
	Excellent Effort. Best Effort, Background
3	Network Control, Voice
	Video, Controlled Load
	Excellent Effort, Best Effort, Background
4	Network Control, Voice
	Video, Controlled Load
	Excellent Effort, Best Effort
	Background
5	Network Control, Voice
	Video
	Controlled Load
	Excellent Effort, Best Effort
	Background
6	Network Control, Voice
	Video
	Controlled Load
	Excellent Effort
	Best Effort
	Background
	Best Effort
7	Network Control
	Voice
	Video
	Controlled Load
	Excellent Effort
	Best Effort
	Background

Table 8-2: Mapping of traffic classes into priority queues (P802.1p).

Both full duplex and P802.1p/P802.1Q standards extensions are important but not sufficient QoS enablers. The remaining part is admission control. Without it, each priority class may intermittently degrade to best-effort performance. Here, EPON can provide a simple and robust method for performing admission control. In Section 8.3.3, we mentioned that multi-point control protocol (MPCP) relies on GATE messages sent from the OLT to ONUs to allocate the transmission window. A very simple protocol modification may allow a single GATE message to grant multiple windows, one for each priority class. REPORT message is also to be extended to report queue states for each priority class. Alternatively, admission control can be left to higher-layer intelligence in ONUs. In this case, the higher layer will know when the next transmission window will arrive and how large it will be, and will schedule packets for transmission accordingly.

8.5.3.1 Performance of COS-Aware EPON

In this section, we will investigate how priority queuing will allow us to provide a delay bound for some services. Below we describe a simulation setup in which data arriving from the user is classified in three priority classes and directed to different queues in the ONU. The queues then are serviced in order of their priority; a lower-priority queue is serviced only when all higher-priority queues are empty. In this experiment, the tagged ONU has a constant load. We investigate the performance of each class as the ambient network load varies.

Best Effort (BE) class has the lowest priority. This priority level is used for non-real time data transfer. There is no delivery or delay guarantees in this service. The BE queue in the ONU is served only if higher priority queues are empty. Since all queues in our system share the same buffer, the packets arriving at higher priority queues may displace the BE packets that are already in the BE queue. In our experiment, the tagged source has the BE traffic with an average load of 0.4 (40 Mbps).

Assured Forwarding (AF) class has higher priority than the BE class. The AF queue is served before the BE queue. In our experiment, the AF traffic consisted of a VBR stream with average bit rate of 16 Mbps. This corresponds to three simultaneous MPEG-2-coded video streams [19]. Since the AF 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 AF or higher priority packets.

Guaranteed Forwarding (GF) priority class was used to emulate a T1 line in the packet-based access network. The GF class has the highest priority and can displace the BE and AF data from their queues if there is not enough buffer space to store the GF packet. A new GF packet will be lost only if the entire buffer is occupied by GF packets. The GF queue is served before the AF and BE queues. The T1 data arriving from the user is packetized in the ONU 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 T1 data consumed the bandwidth equal to 4.48 Mbps. Of course, we could put 48 bytes of T1 data in one packet and send one 94-byte packet every 250 µs. This would consume only 3.008 Mbps, but will increase the packetization delay.

Figures 8-25 and 8-26 show the average and the maximum packet delay for each type of traffic. The average load of the tagged ONU was set to 40 Mbps of BE data, 16 Mbps of AF data, and 4.48 Mbps of GF data, or to a total of ~60 Mbps. The horizontal axis shows the effective network load. These figures show how the traffic parameters depend on the overall network load.

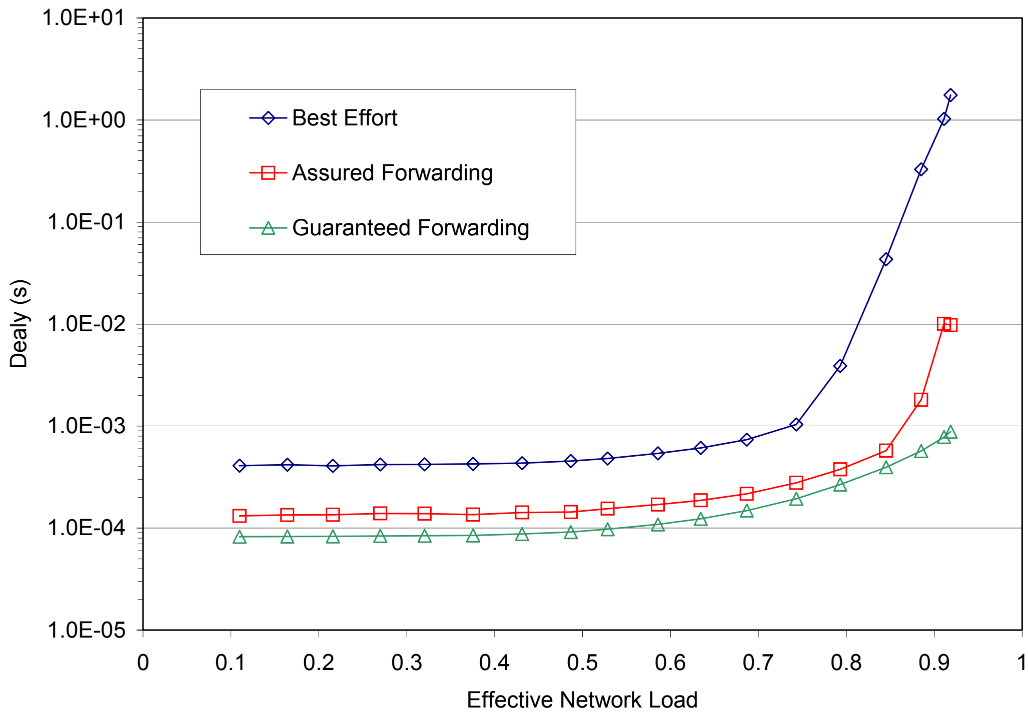


Figure 8-25. Average packet delay for various classes of traffic as a function of the effective network load.

We can see that the BE traffic suffer the most when the ambient load increase. Its delay increase and the simulation results show that some packets were discarded when the network load exceeded 80%. The AF data also experienced an increased delay, but no packet losses were observed. The increased delay in the AF 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 GF 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 T_{MAX} (2 ms with our chosen set of parameters).

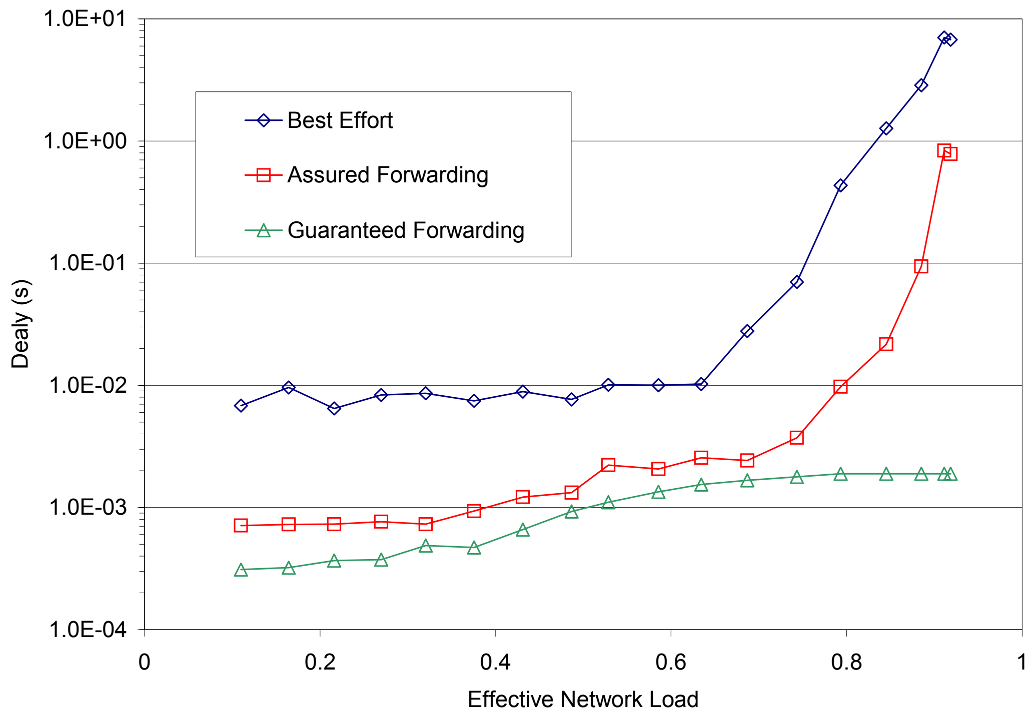


Figure 8-26. Maximum packet delay for various classes of traffic as a function of the effective network load.

Of course, to restore the proper T1 rate, a shaping buffer (queue with constant departure rate) should be employed at the receiving end (in the OLT). After receiving a packet (or a group of packets) from an ONU, the shaping buffer should have at least 2 ms worth of data, i.e., 384 bytes of T1 data. This is because the next packet from the same ONU may arrive after the maximum delay of 2 ms. When such a delayed packet arrived, it should still find the non-empty buffer. Let us say the minimum buffer occupancy is 24 bytes which is 125 µs of T1 transmission time (we call it a *buffer under-run protection time*). In this case, the overall latency experienced by T1 data will consist of:

1. 125 µs of packetization delay in ONU;
2. polling delay in ONU (up to T_{MAX});
3. up to 100 µs of propagation delay (assuming maximum distance of 20 km); and
4. wait time in the shaping buffer.

Items 2 and 4 together are equal to T_{MAX} plus buffer under-run protection time, i.e., 2.125 ms. Thus, the total latency is about 2.35 ms. If this latency is too high for a given specification, it can be reduced by decreasing T_{MAX} (see Equation 5).

8.6 Security issues

Security has never been a strong part of Ethernet networks. In point-to-point full-duplex Ethernet, security is not a critical issue because there are only two communicating stations, using a private channel. In shared half-duplex Ethernet, security concerns are minimized because users belong to a single administrative domain and are subject to same set of policies.

EPON, however, has a different set of requirements, mostly due to its intended use in subscriber access environment. EPON serves non-cooperative, private users, but on the other hand, has a broadcasting downstream channel, potentially available to any interested party capable of operating an end station in promiscuous mode. In general, to ensure EPON security, network operators must be able to guarantee subscriber privacy, and must be provided mechanisms to control subscriber's access to the infrastructure.

In a residential access environment, individual users expect their data to remain private. For the business access application, this requirement is fundamental. The two main problems associated with lack of privacy are subscriber's susceptibility to eavesdropping by neighbors (a subscriber issue), and susceptibility to theft-of-service (a service provider issue). Let us explore these two problems:

8.6.1 Eavesdropping

In EPON, eavesdropping is possible by operating an ONU is promiscuous mode: being exposed to all downstream traffic, such an ONU can listen to traffic intended to other ONUs.

Point-to-point emulation adds link IDs (see Section 8.3.4) that allow an ONU to recognize frames intended for it, and filter out the rest. However, this mechanism does not offer the required security, as an ONU might disable this filtering, and monitor all traffic.

The upstream transmission in an EPON is relatively more secure. All upstream traffic is multiplexed, and is visible only to the OLT (due to directivity of a passive combiner). Although reflections might occur in the passive combiner, sending some upstream signal downstream again, the downstream transmission is in a different wavelength than the upstream transmissions. Thus, the ONU is "blind" to reflected traffic which is not processed in the receive circuitry.

The upstream can also be intercepted at the PON splitter/combiner, as splitters and combiners are most often manufactures as symmetrical devices, i.e., even though only one coupler port is connected to the trunk fiber, more ports are available. A special device sensitive to the upstream wavelength can be connected facing downstream to one such unused port. This device will be able to intercept all upstream communications.

8.6.2 Theft-of-service

Theft of service occurs when a subscriber impersonates his neighbor, and transmits frames that are not billed under impersonator's account. OLT obtains the identity of the subscriber through link ID inserted by each ONU in the frame preambles. This link ID can be faked by the malicious ONU when transmitting in the upstream direction. Of course, to be able to transmit in the hijacked timeslot, the impersonating ONU should also be able to eavesdrop to receive GATE messages addressed to a victim.

8.6.3 Solving the Problem using Encryption

Encryption of downstream transmission prevents eavesdropping when the encryption key is not shared. Thus, a point-to-point tunnel is created that allows private communication between the OLT and the different ONUs.

Encryption of the upstream transmission prevents interception of the upstream traffic when a tap is added at the PON splitter. Upstream encryption also prevents ONU impersonation: data arriving from an ONU must be encrypted with a key available to that ONU only. There exist secure methods of key distribution, but they are outside the scope of this book.

Encryption and decryption may be implemented at the physical layer, data link layer, or in higher layers. Implementing encryption above the MAC sub-layer will encrypt the MAC frame payload only, and leave headers in plain text. In that scenario, the transmitting MAC will calculate Frame Check Sequence (FCS) for the encrypted payload, and the receiving MAC will verify the received frame integrity before passing the payload to a higher sub-layer for decryption. This scheme prevents malicious ONUs from reading the payload, but they may still learn other ONUs MAC addresses.

Alternatively, encryption can be implemented below the MAC. In that scheme, the encryption engine will encrypt the entire bit stream, including the frame headers and FCS. At the receiving end, decryptor will decrypt the data before passing it to MAC for verification. Since encryption keys are different for different ONUs, frames not destined to a given ONU will not decrypt into a properly formed frame and will be rejected. In this scheme no information may be learned by a malicious ONU. Implementing an encryption layer below MAC appears to be more secure and reliable method.

8.6.3.1 Encryption method

The downstream transmission in an EPON is a frame-based communication channel in which each frame is addressed to a different destination. As each frame is an independent piece of information, the encryption method can not be stream-based. The most appropriate solution is a block-based cipher that encrypts each frame separately.

The link ID field located in each frame's preamble is used to identify a tunnel between the OLT and an ONU (PtP emulation). This header may also be used to support encryption mechanism in the EPON. For this purpose one of the reserved bytes in the header will be used as a key index (key identifier) (Figure 8-27). Based on the value of this field it will be possible to determine whether the frame is encrypted, and what key was used.

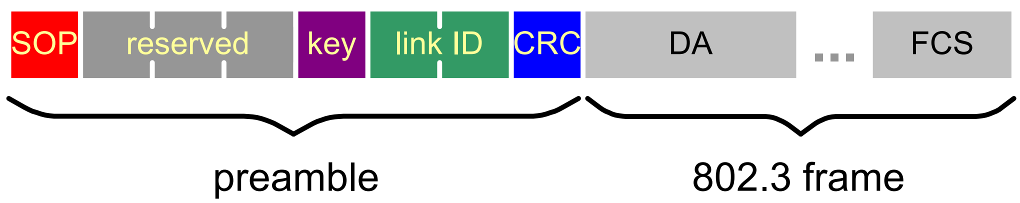


Figure 8-27. Frame preamble with embedded link ID and encryption key index.

Each ONU holds a key that is valid for the current session. The key identifier references the key in the ONU. This behavior allows for smooth transition from one valid session to the next, when re-keying the session. A default key identifier is used for frames sent unencrypted. This mechanism has a built-in expansion path as different key indexes can lead to different cipher algorithms, and allow for eventual implementation of conditional access systems at layer 2.

Periodic re-keying allows maintaining security of the established tunnels indefinitely. As block ciphers use fixed-size blocks, and the Ethernet frames are of variable length, block boundary may be dissimilar to packet boundary, and the last block will be stuffed to reach the required block size. As zero stuffing is a potential weakness in the encryption, an alternative method is used in which the last n bits ($n < 128$) will be XOR-ed with the result of a second cipher iteration of the next-to-last block.

The Advanced Encryption Standard (AES) algorithm, originally designed to replace the aging Data Encryption Standard (DES), is considered for Ethernet PONs. This algorithm allows the use of 128-bit, 192-bit or 256-bit keys.

8.7 EPON Upgrade Scenarios

By the amount of bandwidth it makes available to subscribers, EPON is a giant step forward compared to DSL and cable modem technologies. With a line rate of 1 Gbps (current target for IEEE 802.3ah efforts) and 16 to 64 subscribers per EPON, each user will get between 15 and 60 Mbps bandwidth. Still, unavoidably, as more bandwidth-intensive services become available to users, this capacity will get exhausted. It is therefore, crucial for the success of EPON technology to provide a smooth path for future upgrades. It is hard to envision what upgrade scenario will be most favorable at the time when the EPON capacity will become a limiting factor. The best scenario will not require a lift-fork upgrade (i.e., will allow incremental expenses) and will capitalize on the most matured technology. In this section we consider three possible directions: wavelength upgrade, rate upgrade, and spatial upgrade.

8.7.1 Wavelength Upgrade

If WDM technology matured enough to provide high-volume, low-cost components it may become economically feasible to migrate EPON to a multi-wavelength configuration. In this scheme, some of the EPON ONUs will migrate to new wavelengths for both upstream and downstream traffic. While the data rate on each wavelength will remain the same, there will be fewer ONUs to share that bandwidth capacity. This procedure can be repeated again in the future and eventually will lead to a WDM PON system where each subscriber is allocated its individual wavelengths. The major cost factor of such an upgrade is the necessity of tunable transmitters and receivers in ONUs. Also, the OLT must have multiple transceivers (tunable or fixed) – one for each wavelength. To allow incremental upgrade, a new spectral region should be allocated for premium ONUs. This will allow non-premium ONUs to continue using cheap 1310 nm lasers with high spectral width. Only the premium ONUs will have to be replaced by new ones operating at different wavelength or having tunable transceivers.

8.7.2 Rate Upgrade

With the finalizing of the 10 Gbps Ethernet standard by the IEEE, rate upgrade appears as an attractive solution. To allow incremental upgrade cost, only a subset of ONUs may be upgraded to operate at higher rates. Thus, the rate-upgrade scenario will call for a mixed-rate EPON where some ONUs operate at 1 Gbps, and some at 10 Gbps. This upgrade would require high-speed electronics in the OLT which is able to operate at both rates. The main challenge with this kind of upgrade is that dispersion penalties become much more severe. The narrow margin of available power budget may not allow the desired distances or concentration ratios.

8.7.3 Spatial Upgrade

It is reasonable to expect the fiber prices to reduce enough to consider a spatial-upgrade scenario in which a subset of users is migrated to a separate EPON. In this scenario, a new trunk fiber is deployed from the central office to the splitter and some branches are re-attached to a new trunk fiber (and a new splitter). To avoid the cost of additional fiber deployment, this upgrade fiber can be pre-deployed at the time of the original deployment. Alternatively, some network operators consider EPON deployment with a splitter located in the central office. This EPON configuration will require as much fiber to be deployed as in the point-to-point configuration, but it will still require only one transceiver in the OLT. This will allow having much higher-density equipment, which is very important in a limited CO space available to competitive local exchange carriers (CLECs) who have to rent CO space from incumbent LECs (ILECs). In a scenario where a splitter is located in the central office, to upgrade to higher bandwidth, some users will be reconnected to another EPON in the patch panel located in the same central office. Given the current state of the technology, this seems to be the most cost-efficient type of upgrade. Eventually, spatial upgrade may lead to a point-to-point architecture with an independent fiber running to each subscriber.

8.8 IEEE P802.3ah Status

The standards work for Ethernet in the local subscriber access network is being done in the IEEE P802.3ah Ethernet in the First Mile Task Force. This group received approval to operate as a Task Force from the IEEE Standards Association (IEEE-SA) Standards Board in September 2001.

The P802.3ah Ethernet in the First Mile Task Force is bringing Ethernet to the local subscriber loop, focusing on both residential and business access networks. While, at first glance, this appears to be a simple task, in reality the requirements of local exchange carriers are vastly different from those of enterprise managers for which Ethernet has been designed. In order to "evolve" Ethernet for local subscriber networks, P802.3ah is focused on four primary standards definitions:

1. Ethernet over copper,
2. Ethernet over point-to-point (PtP) fiber,
3. Ethernet over point-to-multipoint (PtMP) fiber, and
4. Operation, Administration and Maintenance (OAM).

Thus, the Ethernet in the First Mile (EFM) Task Force is focused both on copper and fiber standards, optimized for the first mile, and glued together by a common operation, administration and maintenance system. This is a particularly strong vision, as it allows a local network operator a choice of Ethernet flavors using a common hardware and management platform. In each of these subject areas, new physical layer specifications are being discussed to both meet the requirements of service providers, while preserving the integrity of Ethernet. Standards for Ethernet in the First Mile are anticipated by September 2003, with baseline proposals emerging as early as March 2002.

The Ethernet over Point-to-Multipoint (PtMP) track is focusing on the lower layers of an EPON network. This involves a Physical (PHY) layer specification, with possibly minimal modifications to the 802.3 MAC. The standards work for P2MP fiber-based Ethernet is in progress, while the multi-point control protocol framework is emerging. This emerging protocol uses MAC control messaging (similar to the Ethernet PAUSE message) to coordinate multipoint-to-point upstream Ethernet frame traffic. Materials concerning the P802.3ah standards effort can be found at [20] and presentations materials at [21].

8.9 Conclusion

Unlike the backbone network, which received an abundance of investment in long-haul fiber routes during the Internet boom, optical technology has not been widely deployed in the access network. It is possible that EPONs and point-to-point optical Ethernet offer the best possibility for a turnaround in the telecom sector. As service providers invest in optical access technologies, this will enable new applications, stimulating revenue growth and driving more traffic onto backbone routes. The large increase in access network bandwidth provided by EPONs and point-to-point optical Ethernet will eventually stimulate renewed investment in metro and long-haul fiber routes.

The subscriber access network is constrained by equipment and infrastructure not originally designed for high-bandwidth IP data. Whether riding on shorter copper drops or optical fiber, Ethernet is emerging as the future broadband protocol of choice, offering plug-and-play simplicity, IP efficiency, and low cost. Of particular interest are Ethernet Passive Optical Networks, which combine low-cost point-to-multipoint optical infrastructure with low-cost, high-bandwidth Ethernet. The future broadband access network is likely to be a combination of point-to-point and point-to-multipoint Ethernet, optimized for transporting IP data, as well as time-critical voice and video.

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