B-ISDN (Broadband Integrated Services Digital Network)

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ABSTRACT

The subject of B-ISDN came into being in the late 1980s, together with the concept of Asynchronous Transfer Mode (ATM). ATM is closely tied to high-speed packet switching by means of specialized switches implemented in hardware. Due to its high speed and packet structure, ATM technology was considered attractive to unify voice, data, and video services. A unification of these services over the telephone infrastructure was attempted earlier by a standards offering known as Integrated Services Digital Network (ISDN). Consequently, this new service unification was termed Broadband ISDN (B-ISDN). Although due to its origins, B-ISDN is sometimes closely tied to ATM technology, the term independently represents the vision of packet-based high-speed integration of voice, data, and video services. It is important that in this process, guarantees to satisfy different Quality-of-Service (QoS) needs (in terms of delay, loss, etc) required by voice, data, and video services are provided. In this vision, what is important is the unification, or integration of services; and the underlying technology is of secondary importance. As of the early 2000s, the technology to be employed in realizing this vision seems to have shifted from its origins of ATM. In this article, our emphasis is on B-ISDN as a service integration vision. Nevertheless, we will describe its original emphasis as the service offering of ATM as well as the path the industry seems to be taking in implementing this vision.

Keywords: ISDN, B-ISDN, ATM, IP, QoS, ITU-T

1 Broadband ISDN

We first outline the history of the B-ISDN vision and then move on to the ATM technology that is envisioned to fulfill this vision.

1.1 History of B-ISDN

Shortly after the invention of the telephone by A. G. Bell in 1876, means to interconnect or network telephones at different locations were devised. Within only two years, the first switching center was built [4]. In the United States (US), during the 20^{th} century, a public company, the Bell System and its parent, AT&T, emerged as the national provider of telephony services. The fundamental principle, formulated by AT&T president T.Vail in 1907, was that the telephone would operate most efficiently as a monopoly providing universal service, by the nature of its technology. The US government accepted this principle in 1913. The Bell System made steady progress towards its goal of universal service, which came in the 1920s and 1930s to mean everyone should have a telephone. Percentage of American households with telephone service reached 50% in 1945, 70% in 1955, and 90% in 1969. This network was based on analog technology for transmission, signaling, and switching.

The Bell System studied digital telephony, first starting from its theoretical principles during the late 1940s. Most of the principles of digital telephony, such as theory of sampling, theory of quantization, and fundamental limits in information transfer were invented or perfected by Bell System scientists such as H. Nyquist, J. R. Price, S. P. Lloyd, and C. E. Shannon in the late 1940s. Parallel with this progress in theory was a fundamental breakthrough in device technology known as the transistor, introduced, again by the Bell System, in 1948. The transistor would make the digital telephony revolution possible, while many years later, powerful integrated circuits would spark the dream of B-ISDN.

Digitization of the telephony network was useful since it provided a number of advantages:

- Ease of multiplexing,
- Ease of signaling,
- Integration of transmission and switching,
- Increased noise tolerance,
- Signal regeneration,
- Accommodation of other services,
- Performance monitoring,
- Ease of encryption.

First deployment of digital transmission was in 1962 by the Bell System, while the first digital commercial microwave system was deployed in Japan in 1968. Research on digital switching was initiated in 1959 by Bell Labs. First deployment of a digital switch in the public network was in 1970 in France while in the US, the Bell System deployed an electronic switch known as 4ESS in 1976 [4].

CCITT (Comité Consultatif International de Télégraphique et Téléphonique, or Consultative Committee for International Telegraph and Telephone) is a committee of the International Telecommunications Union (ITU), which is a specialized agency of the United Nations. ITU was originally established after the invention of telegraphy in 1865 and became a specialized agency of the United Nations in 1947, shortly after the formation of the United Nations. Similar to ITU, CCITT was originally established as a standardization organization in the field of telegraphy, in 1925. In 1993, standardization aspects of CCITT and those of the sister radio standardization committee, CCIRR, were unified under the name ITU-T (International Telecommunications

Union – Telecommunication Standardization Sector). Members of ITU-T are governments. ITU-T is currently organized into 13 study groups that prepare standards, called Recommendations. There are 29 Series of Recommendations (A-Z). Work within ITU-T is conducted in 4-year cycles.

In 1968, CCITT established Special Study Group D to study the use of digital technology in the telephone network. This Study Group established 4-year study periods beginning with 1969. The first title of the group was "Planning of Digital Systems". By 1977, the emphasis of the Study Group was on "Overall Aspects of Integrated Digital Networks" and "Integration of Services". As of 1989, the title of the study shifted to "General Aspects of Integrated Services Digital Networks". The concept of an "Integrated Services Digital Network" was formulated in 1972 as one in which "the same digital services and digital paths are used to establish for different services such as telephony and data" [29]. The first ISDN standard was published in 1970, under the title "G.705 Integrated Services Digital Network (ISDN)". Although this first document of an ISDN standard is in the Series G Recommendations, most of the ISDN standards are in the Series I Recommendations, with some also in G, O, Q, and X Series Recommendations.

Three types of ISDN services are defined within the ISDN Recommendation I.200:

- Bearer services,
- Tele-services,
- Supplementary services.

Bearer services (I.140) provide the means to convey information in the form of speech, data, video, etc between users. There is a common transport rate for bearer services: it is the 64 kb/s rate of digital telephony. Various bearer services are defined as multiples of this basic 64 kb/s service, for example 64, 2x64, 384, 1536, and 1920 kb/s [29]. Tele-services cover user applications and are specified in I.241 as telephony, teletex, telefax, mixed mode, videotex, and telex. Supplementary services are defined in I.250. These services are related to number identification (such as calling line identification), call offering (such as call transfer, call forwarding, and call deflection), call completion (such as call waiting and call hold), multiparty (such as conference or three-party calling), community of interest (such as a closed user group), charging (such as credit card charging), and additional information transfer (such as the use of the ISDN signaling channel for user-to-user data transfer).

Towards the end of 1980s and almost two decades after the first study group on ISDN was formed at the CCITT, ISDN was still not being deployed by service providers at a commercial scale, especially in the US. It is important to review the reasons for this absence of activity. ISDN required digitization of both the telephony network and the subscriber loop (connection between a residence or a business and the central office of the telecommunications service provider). While the network was becoming digital, and doing so involved economies of scale (and thus was relatively inexpensive), making the subscriber loop digital required replacement of the subscriber front end equipment at the central office. This was a labor-intensive, expensive process. In addition, there was not a compelling push from consumers demanding ISDN. With the network becoming digital, the quality and reliability advantages of voice transmission were achieved. In addition, it was possible to offer supplementary services (such as caller ID, call waiting, etc) as defined by ISDN Recommendations without making the subscriber loop digital. At the time, modem technology enabled data transmission over the subscriber loop at rates up to about 30 kb/s and that was sufficient for most of the available residential data services available (which were text-based). Business data communications needs were restricted to large businesses. These needs were being served with dedicated digital lines (T1 lines) at speeds of 1.5 Mb/s in the US. Although these lines were very expensive, the market for them was relatively small. In addition, it was becoming clear that in order to serve any future ISDN service needs, ISDN transmission speeds would not suffice and packet switching was going to become necessary. At the time, some overestimates were made as to the needed transmission speeds. For example, it was considered that entertainment video was one of the services that service providers would offer on such an integrated network and that the required transmission speeds for these services were in excess of 100 Mb/s. ISDN was certainly insufficient to provide these speeds and its packet switching recommendations were not yet developed.

In 1988, CCITT issued a set of Recommendations for ISDN, under the general name of "Broadband Aspects of ISDN" (I.113: Vocabulary of Terms for Broadband Aspects of ISDN, and I.121: Broadband Aspects of ISDN). This was a time when packet switching was proven in the Internet (although Internet was still a research network), there was increased activity in video coding within the contexts of HDTV (High Definition Television) and MPEG (video coding specification by Moving Picture Experts Group), voice compression was beginning to achieve acceptable voice quality at rates around 8 kb/s, and first residential data access applications were appearing in the context of accessing the office computer and electronic bulletin boards. Consequently, telecommunications industry representatives came to the conclusion that a need for broadband services in the telecommunication network was imminent. Since ISDN was not capable of answering high-speed and packet-based service needs of such services, the concept of B-ISDN was deemed necessary. Aiding in this process was the availability of high-speed transmission, switching, and signal processing technologies. It became clear that even higher processing speeds were going to become available in the near future (e.g., the fact that the speed of processing doubles every 1.5 years, also known as Moore's Law). CCITT considered these signs so important that the usual 4-year cycle of a study group to issue Recommendations was considered too long and an interim set of Broadband ISDN (B-ISDN) Recommendations were first issued in 1990. It should be emphasized at this point that for the telecommunications industry, and specifically for the service providers, the vision of B-ISDN involves the integration of voice, video, and data services *end-to-end* and with Quality-of-Service guarantees.

1.2 ATM Fundamentals

The concept of ATM was first unveiled in an international meeting in 1987 by J. P. Coudreuse of CNET, France [9]. The basic goal of ATM was to define a networking technology around the basic idea of fast packet switching. In doing so, it was recognized that integration of services is desirable, but requires true packet switching in order to be effective and economical. Since new services were expected to operate at multi-megabit rates, a fast packet switching technology was desired. This implied a number of choices (made for simplification purposes):

- Fixed packet size (known as cells),
- Short packet size,
- Highly simplified headers,
- No explicit error protection,
- No link flow control.

Since ATM was an effort to define B-ISDN by telephone equipment vendors and service providers, voice was a major part of the B-ISDN effort from the onset. In fact, the decision on short cell size (53 bytes total, with a 48-byte payload) was made with considerations of echo cancellation for voice. For 64 kb/s voice, the use of echo cancellation equipment becomes necessary if packetization delay is more than 32 bytes (4 ms). Although the public telephone network in the US has echo cancellers installed, smaller European countries do not. To avoid echo cancellation equipment, European countries proposed that the payload for ATM be 32 bytes. The US proposal was 64 bytes and 48 bytes were chosen as a compromise. The maximum tolerable overhead due to the header was considered 10%, and thus the 5-byte header was chosen.

1.3 ATM Protocol Reference Model

The protocol reference model for ATM is shown in Figure 1. This model is different from that of ISDN. In this reference model, the ATM Layer is common to all services. Its function is to provide packet (cell) transfer capabilities. The ATM Adaptation Layer (AAL) is service dependent. The AAL maps higher layer information into ATM cells. The protocol reference model makes reference to three separate planes:

- *User Plane:* Information transfer and related controls (flow and error control),
- *Control Plane:* Call control and connection control,
- *Management Plane:* Management functions as a whole, coordination among all planes, and layer management.

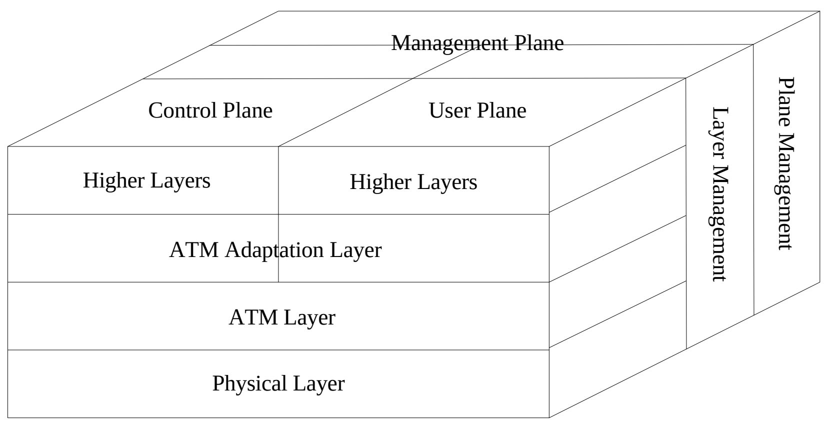

Figure 1. B-ISDN Protocol Reference Model

1.4 ATM Layer

We will first describe the ATM Layer. ATM headers are very simple by design. The cell header has a different structure at the User-to-Network Interface (UNI) and at the Network-to-Network Interface (NNI) (Figure 2 and Figure 3). Routing in ATM is achieved by an identifier field. It is the contents of this field that drives the fast hardware switching of an ATM cell. This field consists of two parts: the Virtual Circuit Identifier (VCI) and the Virtual Path Identifier (VPI). VCI is simply an index to a *connection* [14]. This "connection" is known as a Virtual Circuit (VC). A number of VCs are treated as a single entity known as a Virtual Path (VP). Thus, inside the network, cell switching can be performed based on VPI alone. The VPI field is 8 bits at the UNI and 12 bits at the NNI. The VCI field is 16 bits long at both interfaces. It should be noted that VCIs and VPIs are not addresses. They are explicitly assigned at each segment (link between ATM nodes) of a connection when a connection is established, and they remain so for the duration of the connection. Using the VCI/VPI, the ATM layer can asynchronously interleave (multiplex) cells from multiple connections. As a historical remark, we would like to note that origins of the VPI/VCI concept can be traced back to the Datakit virtual circuit switch, developed by A. Fraser of Bell Labs during the 1970s [14, 13] . Datakit was a product manufactured by AT&T for the data transmission needs of local exchange carriers.

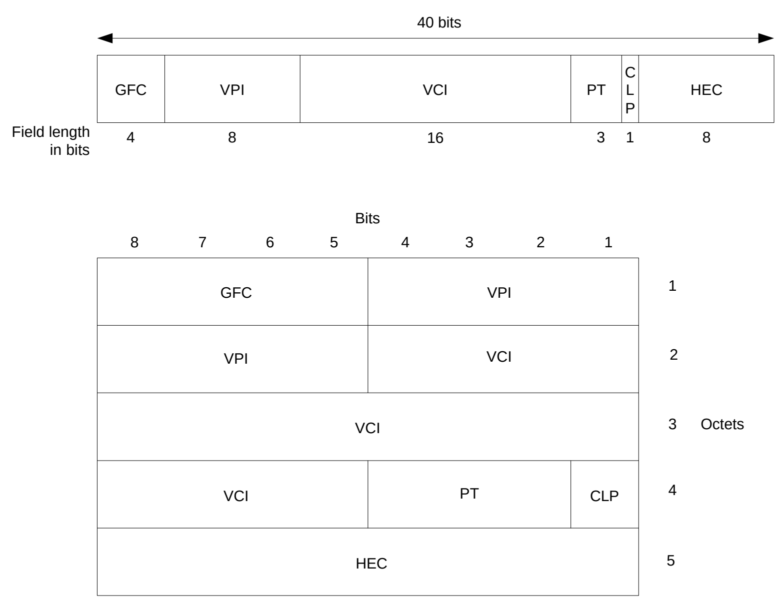

Figure 2. UNI Cell Header

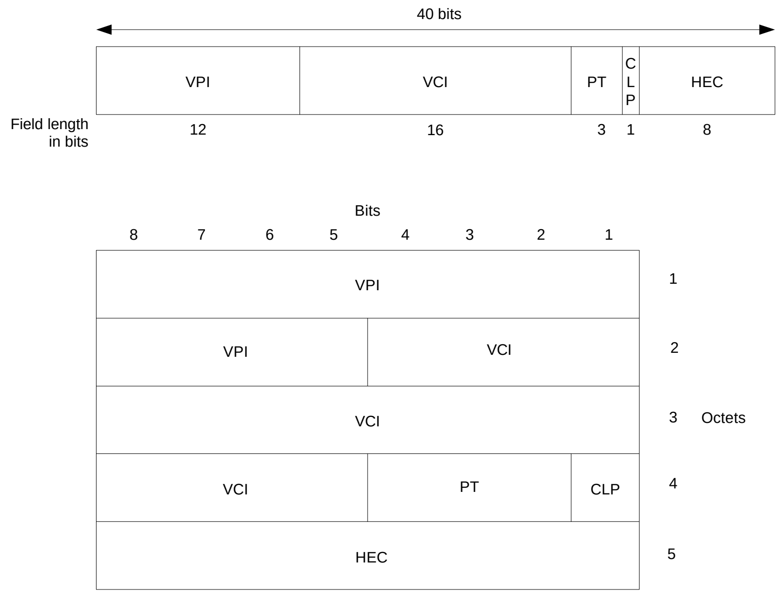

Figure 3. NNI Cell Header

HEC is an error check field, based on an 8-bit cyclic redundancy code (CRC), restricted to the cell header only. Three bits in the header (PT) are used to define the payload type. One bit (CLP) is reserved to indicate cell loss priority. This allows an ATM network to drop packets in case of congestion with the recovery mechanism provided by higher layers: Such dropped cells will be detected by the higher layers of networking protocols (such as TCP/IP) and will be retransmitted. In passing, we would like to note that some earlier design decisions for ATM were later criticized when ATM was used to carry data belonging to the TCP/IP protocol. The most common type of IP packets carried in an IP network are TCP acknowledgement packets. Those packets are 44 bytes long and constitute about half of the packets carried in an IP network. Therefore, about half of IP packets are carried in an ATM network at an efficiency loss of about 10%. This inefficiency was later criticized by service providers in deploying IP-over-ATM networks and was termed *cell tax*.

We stated above that VCI is an index to a connection. Thus, by this concept, ATM networks emulate connections between two points in a network and therefore are termed as *connection-oriented*. VP identifiers do not have to be universal in a network, they can be mapped from a set of values to another at the subnetwork boundary, albeit at a hardware cost. Virtual Connections (consisting of VPs and VCs together) can be established permanently or on a per-need basis. Permanent VCs (PVCs) are established once and all and are simple to work with. For bursty applications, Switched VCs (SVCs) are designed. At a network node, SVCs can be established (added to the VC list) and removed from the VC list on a per-need basis. Although this is a desirable property since not all connections in a network can be known in advance and the goal of developing the technology is indeed provided for bursty, unpredictable traffic, the hardware cost of incorporating this capability is high. In particular, this flexibility of being able to support bursty connections was later criticized since it limits the scalability of the ATM concept due to the difficulty of its implementation for high-speed backbone networks.

A PVC is not signaled by the end points. Both of the endpoint VC values are manually provisioned. The link-by-link route through the network is also manually provisioned. If any equipment fails, the PVC is down, unless the underlying physical network can reroute below ATM. A soft PVC also has manually provisioned end point VC values, but the route through the network can be automatically revised if there is a failure. Failure of a link causes a Soft PVC to route around the outage and remain available. A Switched VC (SVC) is established by UNI signaling methods. So an SVC is a connection initiated by user demand. If a switch in the path fails, the SVC is broken and would have to be reconnected. The difference between an SVC and a soft PVC is that an SVC is established on an “as needed” basis through user signaling. With a soft PVC, the called party cannot drop the connection.

1.5 ATM Adaptation Layer

In order for ATM to support many kinds of services with different traffic characteristics and system requirements, it is necessary to adapt the different classes of applications to the ATM layer. This function is performed by AAL, which is service-dependent. Four types of AAL were originally recommended by CCITT. Two of these have now been merged into one and a new one is added, making the total four once again. The four AALs are now described briefly.

- *AAL1* - Supports connection-oriented services that require constant bit rates and have specific timing and delay requirements. Examples are constant bit rate services such as DS1 (1.5 Mb/s) or DS3 (45 Mb/s) transport.
- *AAL2* - This is a method for carrying voice over ATM. It consists of variable size packets with a maximum of 64 bytes encapsulated within the ATM payload. AAL2 was previously known as composite ATM or AAL-CU. The ITU specification, which describes AAL2 is called ITU-T I.363.2.
- *AAL3/4* - This is intended for both connectionless and connection-oriented variable bit rate services. Two original distinct adaptation layers AAL3 and 4 have been merged into AAL3/4.
- *AAL5* - Supports connection-oriented variable bit rate data services. Compared with AAL3/4, AAL5 is substantially lean at the expense of error recovery and built-in retransmission. This tradeoff provides a smaller bandwidth overhead, simpler processing requirements, and reduced implementation complexity. AAL5 has been proposed for use with both connection-oriented and connectionless services.

There is an additional AAL, AAL0, which normally refers to the case where the payload is directly inserted into a cell. This typically requires that the payload can always be fitted into a single cell so that the AAL is not needed for upper layer PDU delineation when the upper layer PDU bridges several cells.

1.6 ATM Traffic Management

During the early 1990s, the computer networking community was looking for a replacement of the 10 Mb/s Ethernet standard at higher speeds of 100 Mb/s and beyond. ATM, as specified by CCITT, was considered a viable alternative. Various companies active in this field formed an industry consortium, known as the ATM Forum. The ATM Forum later made quick progress in specifying and modifying the ATM Specifications. ATM Forum defined the following traffic parameters for describing traffic that is injected into the ATM network at the UNI [2]:

- Peak Cell Rate (PCR): Maximum bit rate that may be transmitted from the source.
- Cell Delay Variation Tolerance (CDVT): Tolerance controlled by the network provider on how the actual peak rate deviates from the PCR.
- Sustainable Cell Rate (SCR): Upper limit for the average cell rate that may be transmitted from the source.
- Maximum Burst Size (MBS): Maximum number of cells for which the source may transmit at the PCR.
- Minimum Cell Rate (MCR): Minimum cell rate guaranteed by the network.

The ATM Forum defined different service classes to be supported by ATM. The classes are differentiated by specifying different values for the following QoS parameters defined on a per-connection basis:

- *Maximum Cell Transfer Delay (maxCTD):* CTD is the delay experienced by a cell between the transmission of the first bit by the source and the reception of the last bit of the cell by the destination. The CTD of a cell is smaller than the maxCTD QoS parameter of the connection it is carried within with a large probability, or equivalently, maxCTD is the (1 - α) quantile of CTD for a small α.

- *Peak-to-peak Cell Delay Variation (ppCDV):* The ppCDV is the difference between the (1 - α) quantile of the CTD and the fixed CTD that could be experienced by any delivered cell on a connection during the entire connection holding time.
- *Cell Loss Ratio (CLR):* The percentage of cells that are lost in the network due to error or congestion and are not received by the destination.

The QoS parameters are defined for all conforming cells of a connection, where conformance is defined with respect to a Generic Cell Rate Algorithm (GCRA) described in the ATM Forum User-Network Interface Specification 3.1 [2]. The input to this algorithm is the traffic parameters described above.

The proposed service categories by the ATM Forum are then described as follows [1]:

- *CBR (Constant Bit Rate):* The CBR service class is intended for real-time applications, i.e., those requiring tightly constrained delay and delay variation, as would be appropriate for voice and video applications. The consistent availability of a fixed quantity of bandwidth is considered appropriate for CBR service. Cells which are delayed beyond the value specified by maxCTD are assumed to be of significantly less value to the application. For the service class CBR, the attributes PCR, CDVT, maxCTD, ppCDV, and CLR are specified.
- *VBR-rt (Variable Bit Rate – Real Time):* The real-time VBR service class is intended for real-time applications, i.e., those requiring minimal loss and tightly constrained delay and delay variation, as would be appropriate for voice and video applications. Sources are

expected to transmit at a rate that varies with time. Equivalently, the source can be described as "bursty". VRB-rt expects a bound on the cell loss rate for cells conforming to the associated GCRA. Cells delayed beyond the value specified by maxCTD are assumed to be of significantly less value to the application. Real-time VBR service may support statistical multiplexing of real-time sources, or may provide a consistently guaranteed QoS. For VBR-rt, the ATM attributes PCR, CDVT, SCR, MBS, max CTD, ppCDV, and CLR are specified.

- *VBR-nrt (Variable Bit Rate – Non-Real Time):* The non-real time VBR service class is intended for non-real time applications which have "bursty" traffic characteristics and which can be characterized in terms of a Generic Cell Rate Algorithm (GCRA). VRB-nrt expects a bound on the cell loss rate for cells conforming to the associated GCRA. Bounds for cell transfer delay and cell delay variation are not provided for VBR-nrt. Similar to VBR-rt, non-real time VBR service also supports statistical multiplexing of connections. For non-real time VBR, ATM attributes PCR, CDVT, SCR, MBS, and CLR are supported.
- *UBR (Unspecified Bit Rate:)* The UBR service class is intended for delay-tolerant or non-real-time applications, i.e., those which do not require tightly constrained delay and delay variation, such as traditional computer communications applications. Sources are expected to transmit non-continuous bursts of cells. UBR service supports a high degree of statistical multiplexing among sources. UBR service includes no notion of a per-VC allocated bandwidth resource. Transport of cells in UBR service is not necessarily guaranteed by mechanisms operating at the cell level. However it is expected that resources will be provisioned for UBR service in such a way as to make it usable for some set of applications. UBR service may be considered as interpretation of the common term "best effort service." For UBR, only PCR and CDVT are specified as a traffic attribute.

- *ABR (Available Bit Rate):* Many applications have the ability to reduce their information transfer rate if the network requires them to do so. Likewise, they may wish to increase their information transfer rate if there is extra bandwidth available within the network. There may not be deterministic parameters because the users are willing to live with unreserved bandwidth. To support traffic from such sources in an ATM network will require facilities different from those for Peak Cell Rate of Sustainable Cell Rate traffic. The ABR service is designed to fill this need. The traffic attributes PCR, CDVT, MCR, and CLR are specified for the ABR service class.

There are other service categories proposed by the ITU; namely ABT (ATM Block Transfer) and CCT (Controlled Cell Transfer). However, these categories have not gained much acceptance.

2 IP Networks

In the 1990s, while ATM technology was being developed to integrate voice, data, and video, pure data services embraced the TCP/IP protocol, or the IP technology. What made the IP technology attractive is its universal adoption due mainly to the popularity of the global Internet and the unprecedented growth rates the Internet has reached. Initially, IP was not designed for the integration of voice, data, and video to the end user. Developed under the US Department of Defense (DOD) funding, IP was built around reliability and redundancy so as to allow communication to continue between nodes in case of a failure.

2.1 History of IP Networks

There was a perceived need for survivable command and control systems in the US during the 1960s. To fulfill this need, early contributors were drawn from the ranks of defense contractors, federally funded think tanks, and universities: the RAND Corporation, Lincoln Laboratories, MIT, UCLA, and Bolt Beranek and Newman (BBN), under DOD funding.

P. Baran of RAND Corporation postulated many of the key concepts of packet switching networks that were implemented in the ARPANET, the research network Advanced Research Projects Agency (ARPA) of DOD funded in 1967. Baran’s motivation was to use novel approaches to build survivable communications systems. The traditional telephone system is based on a centralized switching architecture and the concept of connection or a “circuit” that must be established between the parties of a communications session using the centralized switches. If a link or a switch is broken (or destroyed) during a connection in this architecture, the communications session will fail, which is unacceptable for survivability purposes. Baran’s work was built around the replacement of centralized switches with a larger number of distributed routers, each with multiple (potentially redundant) connections to adjacent routers. Messages then would be divided into parts (*blocks* or *packets*), and the packets would then be routed independently. This packet switching concept allows bursty data traffic to be statistically multiplexed over available communications paths, makes it possible to adapt to changing traffic demands, and to use existing resources more efficiently without a need for a-priori reservation.

ARPANET was proposed by ARPA as an ambitious program to connect many host computers at key research sites across the country, using point-to-point telephone lines and the packet switching concept. The idea of using separate switching computers, rather than the hosts, was proposed to serve as the routing elements of network, thereby offloading this function from the timesharing hosts. BBN received the contract to build the Interface Message Processors (IMP) in this newly proposed architecture. The engineers at BBN developed the necessary Host-to-IMP and IMP-to-IMP protocols, the original flow control algorithms, and the congestion control algorithms. In the BBN model, hosts communicate with each other via messages. When a host sends a message, it is broken down into packets by the source IMP (which is the IMP directly attached to the host). The IMP then routes each packet, individually through the network of IMPs, to the destination IMP. Each packet will be sent along the path which is estimated to be the shortest, and the path taken by each packet may be different. The destination IMP, upon receiving all packets for a message, will reassemble an exact replica of the original message and forward the message on to the destination host. Based on the implementation of BBN, the ARPANET started to emerge with it first four nodes at UCLA, UCSB, Stanford Research Institute (SRI), and University of Utah in 1969. The ARPANET's purpose was to provide a fast and reliable communication between heterogeneous host machines. The goal of the computer network was for each computer to make every local resource available to any computer in the network in such a way that any program available to local users can be used remotely without much degradation.

In 1969, N. Abramson, motivated by the poor telephone lines in the Hawaiian Islands, launched the Aloha Project at the University of Hawaii, a project funded by ARPA. In this project, the principles underlying a packet switched network based on fixed site radio links were investigated. The Aloha Project developed a new technology for contention-based media access, the so-called "Aloha Protocols," and applied these techniques to satellites as well as radio systems. R. Metcalfe at Xerox PARC built on this work, leading to the development of the Ethernet protocols for access to a shared wirelined medium as a local area networking technology. In 1972, L. G. Roberts and R. Kahn launched the ARPA Packet Radio Program: packet switching techniques on the mobile battlefield. ARPA also created a packet-switched experimental satellite network (SATNet), with work done by Comsat, Linkabit, and BBN. All this work motivated the need for a technology to link these independent networks together in a true "network of networks", the so-called Internet.

In 1973**,** R. Kahn and V. Cerf developed the concept of a network gateway (or a software packet switch), as well as the initial specifications for the Transmission Control Protocol (TCP). With this breakthrough concept, transmission reliability is shifted from the network to end hosts, thus allowing the protocol to operate no matter how unreliable the underlying link is. This paradigm shift was based on the "end-to-end argument" which states that the underlying network is only as strong as its weakest link and therefore improving the reliability of a single link or even an entire subnetwork may have only a marginal effect on the end-to-end reliability. With this paradigm change, the architecture internal to the network was significantly simplified. V. Cerf then joined ARPA to complete the design of the Internet Protocol (IP) Suite, overseeing the separation of the routing portions of the protocols (IP) from the transport layer issues (TCP), and the transition of the new protocols into the ARPANET.

The global Internet began around 1980 when ARPA started converting machines attached to its research networks to the new TCP/IP protocols. The ARPANET, already in place, quickly became the backbone of the new Internet and was used for many of the early experiments with TCP/IP. In 1983, the Defense Communications Agency (DCA) split the ARPANET into two separate networks, one for future research and one for military communications, with the research part retaining the name ARPANET. At around the same time, most university computer science departments were running a version of the Unix operating system available from the University of California's Berkeley software distribution, commonly called Berkeley Unix or BSD Unix. By funding BBN to implement its TCP/IP protocols for use with BSD Unix, and funding University of California Berkeley to integrate the protocols with its software distribution, ARPA was able to reach over 90% of university computer science departments in the US. A large number of hosts subsequently connected their networks to the ARPANET, thus creating the "ARPA Internet".

By 1985, the ARPANET was heavily used and congested. Based on a need for a faster network, National Science Foundation (NSF) initiated the development of NSFNET in the mid-1980s. NSF selected the TCP/IP protocol suite used in the ARPANET. However, as opposed to a single core backbone used in the ARPANET, the earliest form of NSFNET in 1986 used a three-tiered architecture which consisted of universities and other research organizations that are connected to regional networks, which are then interconnected to a major backbone network using 56 kb/s links. The link speeds were then upgraded to T1 (1.5 Mb/s) in 1988 and later in 1991 to T3 (45 Mb/s). In the early 1990s, the NSFNET was still reserved for research and educational applications. At this time, government agencies, commercial users, and the general public began demanding access to NSFNET. With the success of private networks using IP technology, NSF decided to decommission the NSFNET backbone in 1995. Commercial Internet providers then took over the role of providing Internet access. These providers have connection points called Point of Presence (POP). Customers of these service providers are connected to the Internet via these POPs. The collection of POPs and the way they are interconnected form the provider's network. Providers may be regional, national, or global, depending on the scope of their networks. Today's Internet architecture is based on a distributed architecture operated by multiple commercial providers rather than a single core network (NSFNET) that are interconnected via major network exchange points. Historically, commercial Internet providers exchange traffic at Network Access Points (NAP) and the Metropolitan-Area Exchanges (MAE) - through a free exchange relationship called bilateral public peering. Two connectivity models have recently emerged due to increasing congestion in the major exchange points a) private peering among the largest backbone providers and, b) more recently, private transit connections to multiple backbone providers, which are favored by specialized ISPs.

2.2 IP Fundamentals

The Internet provides three sets of services [8]. At the lowest level, one has a connectionless delivery service. The other two services (transport services and application services) lie on top of this connectionless delivery service. The protocol that defines the unreliable, connectionless delivery mechanism is called the Internet Protocol and is commonly referred to by its initials IP. IP defines the basic data unit of data transfer and it also performs the routing function. Therefore, IP is also referred to as the Layer 3 protocol in the Internet suite as it corresponds to the Layer 3

(Network Layer) of the OSI model. Layer 4 protocols such as TCP and UDP run on IP and provide an appropriate higher level platform that the applications depend on.

In addition to internetwork routing, IP provides error reporting and fragmentation and reassembly of information units called datagrams. Datagrams of different size are used by IP for transmission over networks with different maximum data unit sizes. IP addresses are globally unique, 32-bit numbers assigned by the Network Information Center. Globally unique addresses permit IP networks anywhere in the world to communicate with each other.

An IP address is divided into three parts. The first part designates the network address, the second part designates the subnet address, and the third part designates the host address. Originally IP addressing supported three different network classes. Class A networks were intended mainly for use with a few very large networks, because they provided only 8 bits for the network address field. Class B networks allocated 16 bits, and Class C networks allocated 24 bits for the network address field. Because Internet addresses were generally only assigned in these three sizes, there was a lot of wasted addresses. In the early 1990s only 3% of the assigned addresses were actually being used and the Internet was running out of unassigned addresses. A related problem was the size of the Internet global routing tables. As the number of networks on the Internet increased, so did the number of entries in the routing tables. By this time, Internet standards were being specified by an organization known as the Internet Engineering Task Force (IETF). IETF selected CIDR (Classless Inter Domain Routing) [15, 24] to be a much more efficient method of assigning addresses and address aggregation to address these two critical issues.

Classless Inter-Domain Routing (CIDR) is a replacement for the old process of assigning Class A, B, and C addresses with a generalized network prefix. Instead of being limited to network identifiers (or “prefixes”) of 8, 16 or 24 bits, CIDR currently uses prefixes anywhere from 13 to 27 bits. This allows for address assignments that much more closely fit an organization’s specific needs and therefore avoids address waste. The CIDR addressing scheme also enables route aggregation in which a single high-level route entry can represent many lower-level routes in the global routing tables.

In the 1990s, there have also been significant developments in IP routing. There are mainly two routing infrastructures: flat routing and hierarchical routing. In a flat routing infrastructure, each network ID is represented individually in the routing table. The network IDs have no network/subnet structure and cannot be summarized. In a hierarchical routing infrastructure, groups of network IDs can be represented as a single routing table entry through route summarization. The network IDs in a hierarchical internetwork have a network/subnet/sub-subnet structure. A routing table entry for the highest level (the network) is also the route used for the subnets and sub-subnets of the network. Hierarchical routing infrastructures simplify routing tables and lower the amount of routing information that is exchanged, but they require more planning. IP implements hierarchical network addressing, and IP internetworks can have a hierarchical routing structure.

In very large internetworks, it is necessary to divide the internetwork into separate entities known as autonomous systems. An Autonomous System (AS) is a portion of the internetwork under the same administrative authority. The AS may be further divided into regions, domains, or areas that define a hierarchy within the AS. The protocols used to distribute routing information within an AS are known as Interior Gateway Protocols (IGPs). The protocols used to distribute routing information between ASs are known as Exterior Gateway Protocols (EGPs). In today’s Internet, link state protocols like OSPF Version 2 [21] and IS-IS [23] are used as IGPs whereas the path vector protocol BGP-4 [25] is used as a exterior gateway protocol.

With the changes to IP address structure and address summarization with CIDR, and the development of efficient hierarchical routing infrastructures, IP networks have scaled up to the level of universal connectivity today. This has made the Internet a global medium in such a way that any two hosts can communicate with each other as long as they are attached to the Internet. However, currently a packet is transported in the Internet without any guarantees to its delay or loss. Due to this “best effort” forwarding paradigm, the Internet cannot provide integrated services over this infrastructure. As we described previously, the B-ISDN vision requires end-to-end QoS guarantees for different services. The IETF is working on several QoS models that may potentially realize the B-ISDN vision using IP. Using IP as opposed to ATM to realize the B-ISDN vision is a new approach made popular by the widespread use of IP.

2.3 QoS Models in IP Networks

Several QoS architectures are proposed by the IETF for IP networks to enable the support of integrated services over IP networks. We will briefly overview these models below.

2.3.1 Integrated Services (Intserv) Model

The integrated services architecture [6] defines a set of extensions to the traditional best effort model of the Internet so as to provide end-to-end (E2E) QoS commitments to certain applications with quantitative performance requirements. Two services are defined: guaranteed service [28] and controlled load [31] services. Guaranteed service provides an assured level of bandwidth, a firm end-to-end delay bound, and no loss due to queueing if the packets conform to an a-priori negotiated contract. It is intended for applications with stringent real time delivery requirements such as audio and video applications with playback buffers. A packet arriving after its playback time is simply discarded by the receiver. In the case of controlled load service, the network will commit to a flow a service equivalent to that seen by a best-effort flow on a lightly loaded network. This service is intended for adaptive real time applications that can tolerate a certain amount of loss and delay provided it is kept to a reasonable level. The integrated services architecture assumes some explicit setup mechanism such as RSVP (Resource Reservation Protocol) [7]. This setup or signaling mechanism will be used to convey QoS requirements to IP routers so that they can provide requested services to flows that request them. Upon receiving per-flow resource requirements through RSVP, the routers apply intserv admission control to signaled requests. The routers also enable traffic control mechanisms to ensure that each admitted flow receives the requested service independent of other flows. These mechanisms include the maintenance of per-flow classification and scheduling states. One of the reasons that have impeded the deployment of integrated services with RSVP is the use of per-flow state and per-flow processing which typically exceeds the flow-handling capability of today's core routers. This is known as the scalability problem in RSVP or in intserv.

The integrated services architecture is similar to the ATM SVC architecture in which ATM signaling is used to route a single call over an SVC that provides the QoS commitments of the associated call. The fundamental difference between the two architectures is that the former typically uses the traditional hop-by-hop IP routing paradigm whereas the latter uses the more sophisticated QoS source routing paradigm.

2.3.2 Aggregate RSVP Reservations Model

This QoS model attempts to address some of the scalability issues arising in the traditional intserv model. In the traditional intserv model, each E2E reservation requires a significant amount of message exchange, computation, and memory resources in each router along the way. Reducing this burden to a more manageable level via the aggregation of E2E reservations into one single aggregate reservation is addressed by the IETF [3]. Although aggregation reduces the level of isolation between individual flows belonging to the aggregate, there is evidence that it may potentially have a positive impact on delay distributions if used properly and aggregation is required for scalability purposes.

In the aggregation of E2E reservations, we have an aggregator router, an aggregation region, and a deaggregator. Aggregation is based on hiding the E2E RSVP messages from RSVP-capable routers inside the aggregation region. To achieve this, the IP protocol number in the E2E reservation's Path, PathTear, and ResvConf messages is changed by the aggregator router from RSVP to RSVP-E2E-IGNORE upon entering the aggregation region, and restored to RSVP at the deaggregator point. Such messages are treated as normal IP datagrams inside the aggregation region and no state is stored. Aggregate Path messages are sent from the aggregator to the deaggregator using RSVP's normal IP protocol number. Aggregate Resv messages are then sent back from the deaggregator to the aggregator via which an aggregate reservation with some suitable capacity will be established between the aggregator and the deaggregator to carry the E2E flows that share the reservation. Such establishment of a smaller number of aggregate reservations on behalf of a larger number of E2E flows leads to a significant reduction in the amount of state to be stored and the amount of signaling messages exchanged in the aggregation region.

Aggregation of RSVP reservations in IP networks is very similar in concept to the Virtual Path in ATM networks. In this framework, several ATM virtual circuits can be tunneled into one single ATM VP for manageability and scalability purposes. A Virtual Path Identifier (VPI) in the ATM cell header is used to classify the aggregate in the aggregation region (VP switches) and the Virtual Channel Identifier (VCI) is used for aggregation/deaggregation purposes. A VP can be resized through signaling or management.

2.3.3 *Differentiated Services (diffserv)*

In contrast with the per-flow nature of integrated services, differentiated services (diffserv) networks classify packets into one of a small number of aggregated flows or "classes" based on the Diffserv Codepoint (DSCP) written in the Differentiated Services field of the packet's IP header [22]. This is known as Behavior Aggregate (BA) classification. At each diffserv router in a Diffserv Domain (DS domain), packets receive a Per Hop Behavior (PHB), which is invoked by the DSCP. Differentiated services are extended across a DS domain boundary by establishing a Service Level Agreement (SLA) between an upstream network and a downstream DS domain. Traffic classification and conditioning functions (metering, shaping, policing, remarking) are performed at this boundary to ensure that traffic entering the DS domain conforms to the rules specified in the Traffic Conditioning Agreement (TCA) which is derived from the SLA. A PHB then refers to the packet scheduling, queueing, policing, or shaping behavior of a node on any given packet belonging to a BA, as configured by a service level agreement (SLA) or a policy decision. Four standard PHBs are defined:

- *Default PHB* [22]: provides a best-effort service in a diffserv compliant node

- *Class-Selector PHB* [22]*:* To preserve backward-compatibility with any IP Precedence scheme currently in use on the network, diffserv defines a certain DSCP for class selector code points. The PHB associated with a class selector code point is a class selector PHB. There are 8 class selector code points defined.

- *Assured Forwarding (AF) PHB* [16]*:* The AF PHB group defines four AF classes: AF1, AF2, AF3, and AF4. Each class is assigned a specific amount of buffer space and interface bandwidth, according to the SLA with the service provider or a policy decision. Within each AF class, three drop precedence values are assigned. In the case of a congestion indication or equivalently if the queue occupancy for the AF class exceeds a certain threshold, packets in that class with lower drop precedence values will be dropped. With this description, assured forwarding PHB is similar to the controlled load service available in the integrated services model.

- *Expedited Forwarding (EF) PHB* [10]*:* EF PHB provides a guaranteed bandwidth service with low loss, delay and delay jitter. EF PHB can be implemented with priority queuing and rate limiting on the behavior aggregate. EF PHB can be used to provide virtual leased line or premium services in diffserv networks similar to the guaranteed service in intserv networks and the CBR service in ATM networks.

Since diffserv eliminates the need for per-flow state and per-flow processing, it scales well to large core networks.

2.3.4 Hybrid intserv – diffserv [5]

In this QoS model, intserv and diffserv are employed together in a way that end-to-end, quantitative QoS is provided by applying the intserv model end-to-end across a network containing one or more diffserv regions. The diffserv regions of the network appear to the intserv capable routers or hosts as virtual links. Within the diffserv regions of the network, routers implement specific PHBs (aggregate traffic control) based on policy decisions. For example, one of the AF PHBs can be used to carry all traffic using E2E reservations once an appropriate amount of bandwidth and buffer space is allocated for that AF class at each node. The total amount of traffic that is admitted into the diffserv region that will receive a certain PHB may be limited by policing at the edge. The primary benefit of diffserv aggregate traffic control is its scalability. The hybrid intserv – diffserv model is closely related to the RSVP reservation aggregation model.

2.3.5 *Multiprotocol Label Switching (MPLS)*

MPLS introduces a new forwarding paradigm for IP networks in that a path is first established using a signaling protocol. A label in the IP header, rather than the destination IP address is used for making forwarding decisions throughout the MPLS domain [26]. Such paths are called Label Switched Paths (LSP) and routers that support MPLS are called Label Switched Routers (LSR). In this architecture, edge ingress LSRs place IP packets belonging to a certain Forwarding Equivalence Class (FEC) in an appropriate LSP. The core LSRs forward packets only based on the label in the header and the egress edge LSRs remove the labels and forward these packets as regular IP packets. The benefits of this architecture include but are not limited to:

- *Hierarchical Forwarding:* MPLS provides a forwarding hierarchy with arbitrary levels as opposed to the two-level hierarchy in ATM networks. Using this flexibility and the notion of nested labels, several level 1 LSPs can be aggregated into one level 2 LSP, and several level 2 LSPs can be aggregated into one level 3 LSP, and so on. One immediate benefit of this is that the transit provider need not know about the global routes which makes it very scalable [11] for transit providers.
- *Traffic Engineering:* The mapping of traffic trunks (an aggregation of traffic belonging to the same class) onto a given network topology for optimal use of network resources is called the traffic engineering problem. In MPLS networks, traffic trunks are mapped to the network topology through the selection of routes and by establishing LSPs with certain attributes using these routes. A combination of a traffic trunk and the LSP is called an LSP tunnel. In its simplest application, in the case of congestion arising from suboptimal routing, LSP tunnels may be rerouted for better performance.
- *Virtual Circuit Emulation:* Another benefit is that other connection oriented networks may be emulated by MPLS. The advantage is that a single integrated datagram network

can provide legacy services like frame relay and ATM to end customers while maintaining a single infrastructure.

2.3.6 Summary of QoS Models for IP Networks

For elastic applications that can adapt their rates to changing network conditions (e.g., data applications using TCP), a simple QoS model such as "diffserv" will be suitable. For inelastic applications such as real-time voice and video with stringent delay and loss requirements, end-to-end intserv is a better fit. The need for per-flow maintanence in RSVP capable routers is known to lead to a scalability problem especially in core networks. Therefore, several novel QoS models have recently been been introduced to attack this scalability problem. From the perspective of a network, both models rely on eliminating the per-flow maintenance requirement by either aggregating E2E reservations into one single reservation at the border nodes of this network or carrying all E2E reservations in one pre-provisioned diffserv class. However, these architectures pose a burden on the border routers and their success remains to be seen in the commercial marketplace. MPLS, on the other hand, is promising traffic-engineered backbones with routing scalability for all these QoS models.

3 B-ISDN and World Wide Web

In this section we describe the development of IP versus ATM as the underlying networking technology of B-ISDN.

The development of ATM reached full progress at ITU-T during 1989-1990. This effort was led by telecommunications service providers as well as telecommunications equipment manufacturers. The main goal was to develop the switching and networking technology for B-ISDN. As cooperation and contribution from telecommunications industry leaders were at a very substantial level, the vision of an integrated Wide Area Network (WAN) using ATM looked very likely to happen. This development in the WAN sparked interest in other networking platforms. The first affected was the computer communications industry, specifically the Local Area Networking (LAN) community. At the time, available LANs (mainly the Ethernet) had a top speed of 10 Mb/s. The technology had improved from coax to twisted pair and from shared media to switched (1991). However, as user needs increased, the top speed of 10 Mb/s became insufficient and the industry began to search for a replacement at significantly higher speeds of 100-150 Mb/s. At this time, the ATM effort at ITU-T defined a basic transport rate of 155 Mb/s. This speed was very convenient for the LAN community. In addition, adopting the same switching and networking technology with the WAN was attractive from the viewpoint of simplifying the WAN gateway. This led to an industry standardization organization known as the ATM Forum. The goal was to define a set of specifications common to the member companies, primarily for the LAN. An additional goal was to speed up the standardization process, which, at ITU-T, required long study periods and consensus from national representatives.

Another development related to ATM was the emergence of the ADSL (Asymmetric Digital Subscriber Line) technology in the 1990s [19]. At the time, invoking Shannon's capacity formula, the highest transmission rate for a voiceband modem over a subscriber loop, without changing any equipment at the central office, was considered to be about 30 kb/s. The ADSL technology replaces central office channel banks to exploit frequencies above 4 kHz. In addition, it employs sophisticated methods that limit near-end crosstalk and therefore substantially expand the transmission potential of the subscriber loop. As a result, it can operate at rates orders of magnitude higher than those of voiceband modems. The ADSL access network includes terminations both within the home and the public network (ATU-R and ATU-C respectively). The ATU-R is commonly called a "DSL modem," and the ATU-C is commonly called a "DSLAM" (DSL Access Multiplexer). ATM is used as Layer 2 in this "residential broadband" architecture. ADSL provides up to 1.5 Mb/s (downlink) rate. It may be used to extend the ATM network, and therefore QoS properties of ATM, all the way to the residential or corporate desktop. In this model [19], the ATM user-to-network interface (UNI) is tunneled through an ADSL link. By having desktop applications talk directly to the ATM network, bandwidth can be allocated end-to-end across the network which was thought to facilitate the deployment of isochronous, delay-sensitive applications such as voice, video conferencing, etc [17]. In fact, this was the intent of B-ISDN from the onset. The effort to employ ADSL to provide integrated services to the home was led by potential application service providers [20]. At this time, Personal Computer (PC) operating systems did not yet include a networking stack as part of the kernel, and beyond computer terminal emulation, there were not yet any major residential networking applications available. With the arrival of the World Wide Web (WWW) and the concept of a Web browser, the need for an IP stack in PCs became apparent. At the time the most popular PC operating system was Windows Version 3.1 from Microsoft. As this operating system did not have an IP stack, it was added to the operating system manually by the user. Later, Windows 95 became the first PC operating system to include an IP stack. With this development, IP stack became an inevitable option in residential broadband networking. Consequently, the original concept of residential ATM was later modified as IP over ATM over DSL [20]. This could have been a cosmetic change, however, and by this time, the vision of B-ISDN using ATM still looked likely to happen, with a form of IP over ATM being used mainly for best effort data transfer.

A number of developments that took place in the second part of the 1990s have changed the outlook for ATM as the underlying networking technology of B-ISDN. We list these below.

1. IEEE 802.3 Working Group made rapid progress to define a newer version of the Ethernet standard to operate at 100 Mb/s over twisted pair and switched media. This

LAN standard did not have any QoS guarantees, but the solution satisfied a much sought-after need for a LAN operating around 100 Mb/s. This solution was quickly adopted by the marketplace and the 100 Mb/s Ethernet quickly became a commodity product. The absence of a compelling need for QoS in LANs virtually stopped the Local ATM activity. With this development, the ATM Forum lost a major thrust.

2. The development of the WWW and the Web browser, as well as the commercialization of the Internet quickly made Web browsing using a PC a household activity. This development stalled or perhaps even stopped the concept of Residential ATM.

3. A possible application of ATM was in digital cable access systems. By making extensions to the coaxial or hybrid fiber/coaxial cable TV plant so that duplex transmission becomes possible (providing amplification in both directions), and using digital technology so that compression can be used to transmit 100s of TV channels, was under consideration as a potential service offering. Adding data services to this potential offering was attractive. A multi-access control algorithm was needed to share the uplink channel. A standardization activity was initiated under the IEEE 802.14 Working Group. While this group was working on an access system based on ATM and provide QoS guarantees for delivering a multitude of services, and while some progress was made, cable service providers decided to pursue their own standardization effort. They named this activity Multimedia Cable Network System (MCNS). The main reason for this secession was to make the process of standardization faster. As PC operating systems were beginning to offer IP stacks, MCNS chose IP technology as the basis of their own access system. The resulting system specification is known as the Data Over Cable Service Interface Specifications (DOCSIS). Although Version 1.0 of these specifications was for best-effort data service only, in its Version 1.1, DOCSIS supports some QoS

guarantees, specifically designed for Voice over IP (VoIP). DOCSIS is currently the de facto worldwide standard for digital cable access, while 802.14 has stopped its activities. With this development, IP, rather than ATM became the underlying technology for digital cable access systems.

4. A significant advantage of ATM was its fast switching property. ATM was designed to be a simple switching technology so that scalable switches at total throughput values approaching 100s of Gb/s could be built. This vision is by and large correct (although segmentation and reassembly at edge routers can become difficult at higher speeds). However, there was a surprising development in this period: throughput values of routers increased substantially. Today the maximum throughput values of core IP routers compete with those of core ATM switches. Implementation of algorithms for IP address lookup and memory manipulation for variable length packet switching in ASICs is largely responsible for this development.

5. The ATM Forum was founded as an industry organization with the premise of fast standardization. As we noted above, ITU-T requires long study periods and consensus among national representatives. It was thought the ATM Forum would move faster in reaching to a standard. Although that was partially achieved, the industry perception is that signaling became too complex in the ATM Forum.

6. In the 1990s, a number of developments took place in optical transport systems that altered network switching in a major way. First, invention of Erbium Doped Fiber Amplification made long distance optical transmission without intermediate electrical conversions possible. Second, development of Wavelength Division Multiplexing (WDM) or Dense WDM (DWDM) made transport of a large number of wavelengths in a

single fiber possible. The number of wavelengths approached 100s while transmission speeds on individual wavelengths approached 10 Gb/s. Third, wavelength routing or wavelength cross-connects made it possible to demultiplex individual wavelengths from a single fiber and multiplex wavelengths from different fibers into a single fiber. The result is a wavelength switch with total throughput in the 10s of Tb/s range. As a result, wavelength routing provided an alternative to electronic switching at the network core, thus making the scalability argument of ATM switching less attractive.

7. A major advantage of ATM was its QoS capabilities. However, as described in the previous section, IP community developed a set of QoS capabilities. Although there are questions and uncertainties about the realization of these capabilities, there is some established confidence in IP QoS. We would like to note that ATM actually was never deployed for the end-to-end QoS vision. The reason for this is the complexity in signaling and the needed per-flow queuing. The multiclass and aggregate IP QoS model may indeed be more scalable.

8. IP embraced ATM's VP concept. MPLS essentially implements VPs. Various tunneling mechanisms introduced into IP make switching aggregated traffic in IP possible. Furthermore, the end points of a VP implemented by MPLS do not need to be routing peers, which significantly reduces the number of peerings in the network, and therefore routing scalability.

9. Due to its scalable fast switching nature, ATM switches were used to carry and switch IP traffic. However, over time, other solutions were developed that avoid the ATM layer in between. For example, at one point, service providers deployed IP over ATM over SONET over WDM. IP was employed since applications required it, ATM was employed

for high-speed packet switching, SONET was employed because of its fast restoration capability via SONET rings, and WDM was employed for higher transmission speeds in a single fiber. The industry sought for ways to simplify this complicated hierarchy. As a result, IP extended to assume many of the functionalities of ATM and even some of those of SONET (e.g., Resilient Packet Rings).

10. We described the 10% inefficiency that results in carrying TCP/IP traffic over ATM, known as the cell tax, above. Several service providers claimed this inefficiency was too high. In reality, with IP extending to assume many of ATM's functionalities, the need for IP over ATM was alleviated and the cell tax became irrelevant.

11. In the 1980s there were several attempts made to build private networks for multiple location enterprises. These typically employed nailed up leased lines, used voice compression to reduce voice rates, and combined voice and data. Such networks, called Private Networks, were the precursors of integration of services, albeit at a small scale. As discussed above, first ISDN and then B-ISDN had the vision of integrated services. In an integrated public packet network, security, by means of proper authentication and encryption, enables construction of a Virtual Private Network (VPN). A VP is very useful in the construction of a VPN since it simplifies processing of data belonging to a particular VPN in the network. Thus ATM is a natural way to implement VPNs. However, as described above, solutions were developed to embrace the same concept in the IP world. Examples of such protocols are Layer 2 Tunneling Protocol (L2TP) [30], IPSec [18], GRE [12]. MPLS, on the other hand, makes it possible to build provider provisioned scalable VPNs also making use of BGP4 for routing and label information distribution [27]. Thus ATM is no longer a unique method to implement VPNs.

12. Another aspect of the aggregation property of VPs is the traffic engineering potential it provides. For example, one of the possibilities in integrated networks is to use different routes based on QoS properties of different flows, e.g., belonging to the same source and destination pair. There are tools, such as the concept of equivalent bandwidth, that enable traffic engineering for integrated networks. Then, VPs become very useful tools to implement the desired property. Obviously, with the development of VP-like concepts in IP networks, the superiority of ATM in this regard is no longer valid.

To summarize, from the discussion above it appears that two related events stalled the development of B-ISDN:

1. The appearance of the WWW made IP protocol instantly ubiquitous. Common PC operating systems quickly adopted an IP stack. A similar ATM stack was not needed because there was no immediate application tied to ATM in the way WWW was tied to IP.

2. IP quickly extended to assume the advantageous properties of ATM, at least in theory. As a result, ATM lost its role as the underlying technology that glues B-ISDN all together.

Therefore, it is safe to say that B-ISDN is not likely to happen as it was originally designed at the ITU-T, frequently described by the acronym B-ISDN/ATM.

Having said that, we must reiterate that integration of services is certainly useful for the consumer. Furthermore, there appears to be an increasing (albeit at a smaller rate than expected) demand for broadband services. Thus, in the near future some form of a service offering that unifies voice, broadband data, and video can be expected (in fact, it currently exists in digital cable). Whether this offering will eventually become a ubiquitous service such as expected of B-ISDN/ATM depends on many factors and it is difficult to predict today (in mid-2002). It is clear, however, that Voice over IP (VoIP) will be used to carry some voice traffic, especially in traffic engineered enterprise networks. The degree of voice compression available for VoIP (~8 kb/s versus 64 kb/s, although with VoIP overhead, this ratio of 1/8 becomes bigger), statistical multiplexing advantages, and the capability to combine with data in VPNs is an attractive value proposition. Adding the Public Switched Telephone Network and video services to this value proposition successfully in the marketplace in the short term, however, is a taller order.

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