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**Triinet: a demand-adaptive media-access protocol for metropolitan area networks**

Sirazi, Semir, Ph.D.

ILLINOIS INSTITUTE OF TECHNOLOGY, 1987



U·M·I
300 N. Zeeb Rd.
Ann Arbor, MI 48106

# TRIINET: A DEMAND-ADAPTIVE MEDIA-ACCESS PROTOCOL FOR METROPOLITAN AREA NETWORKS

BY

SEMIR SIRAZI

Submitted in partial fulfillment of the
requirements of the degree of
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Approved Graham Campbell
Adviser



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S.S.

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## LIST OF ABBREVIATIONS AND SYMBOLS

| Abbreviation | Term |
|---|---|
| AP | Alternating Priorities |
| ATDMA | Asynchronous Time-Division Multiple Access |
| BER | Bit Error Rate |
| BRAM | Broadcast Recognizing Access Method |
| CAD | Computer Aided Design |
| CATV | Community Antenna Television |
| CDMA | Code-Division Multiple Access |
| CPODA | Contention Priority-Oriented Demand Assignment |
| CSMA | Carrier Sense Multiple Access |
| CSMA/CD | Carrier Sense Multiple Access with Collision Detection |
| FDM | Frequency-Division Multiplexing |
| FDMA | Frequency-Division Multiple Access |
| FPODA | Fixed Priority-Oriented Demand Assignment |
| FSK | Frequency Shift Keying |
| GSMA | Global Scheduling Multiple Access |
| ISDN | Integrated Services Digital Network |
| ISO | International Standards Organization |
| LAN | Local Area Network |
| LLC | Logical Link Control |
| MAN | Metropolitan Area Network |
| MLMA | Multi-Level Multiple Access |
| OSI | Open System Interconnection |
| PBX | Private Branch Exchange |

| Abbreviation | Term |
|---|---|
| P-CSMA | Prioritized Carrier Sense Multiple Access |
| P-CSMA/CD | Prioritized Carrier Sense Multiple Access with Collision Detection |
| PDN | Public Data Network |
| PDU | Protocol Data Unit |
| PSK | Phase Shift Keying |
| RF | Radio Frequency |
| RO | Random Order |
| RR | Round Robin |
| SDU | Service Data Unit |
| STDMA | Synchronous Time-Division Multiple Access |
| TDM | Time-Division Multiplexing |
| TDMA | Time-Division Multiple Access |
| TTL | Transistor-Transistor Logic |
| VLSI | Very Large System Integration |
| VT-CSMA | Virtual-Time Carrier Sense Multiple Access |

## ABSTRACT

Several media-access protocols have been proposed and implemented for packet-switching networks including Satellite, Radio, and Local Area Networks. The protocols employed in these packet-switching networks are mainly intended and tailored for certain applications such as data and multiplexed voice. With the emergence of Metropolitan Area Networks (MANs) using coaxial cable and fiber optics cable as a transport medium, a new media-access protocol is needed to offer integrated services such as data, packetized voice, and digitized video over a single shared channel.

This dissertation proposes a new media-access protocol and communication network topology for MANs. This protocol will support data, packetized voice and digitized/compressed video services. The scope of this dissertation also includes the definition and characterization of Metropolitan Area Networks encompassing network topologies, properties, and service requirements. The media-access protocols intended for Satellite, Radio, and Local Area Networks are summarized and vigorously scrutinized as applied to Metropolitan Area Networks.

# CHAPTER I

# INTRODUCTION

## 1.1 Evolution of Media-Access Protocols

The rapidly declining cost of communication links makes it more economical to access specialized resources at their source rather than making hardware and software available at each user site. Packet-switched networks have emerged as the main transport mechanism for a variety of services ranging from transmission of data and voice to the delivery of digitized images and graphics. Packet switching is a reapplication of the basic dynamic allocation techniques used for over a century by the mail and telegraph systems [KUO 81].* Packet switching can readily adapt to a wide range of user services and user demands because it permits communication resources to be used at utmost efficiency. Presently, packet switching is primarily being used in connection with computer and data communications. However, its effectiveness for digitized voice and video, and other wideband communication services has demonstrated that this technique will undoubtedly become a "de facto" standard for integrated service networks [ROSN 82].

The speed and volume requirements of the data traffic created by the advent of desk-top computers and distributed

* Numbers in parentheses refer to numbered references in the bibliography.

computing devices are beginning to overflow the conventional circuit-switching telephone network [COOK 84]. On the other hand, the transmission and processing of voice and video signals is shifting towards the digital arena [MOKH 84]. For example, digital PBXs are being designed to serve as local area networks to support voice and data services simultaneously. It is perceived that the integration of these services, data, voice and video, over a single medium shared among thousands of users would be the most economical and feasible solution [MAZU 83, MASO 83, RITC 83]. However, an arbitration mechanism, namely a media-access protocol is required to support all these services which have different requirements. Specifically, the degree of throughput-delay-stability tradeoffs for the media-access protocols may vary considerably such that one media-access protocol may not meet all application requirements. In addition, the performance of a multiple access protocol, in which conflicts are resolved through a certain method, is strongly dependent on the traffic model and network loading. In general, some traffic characteristics do favor one class of protocols more than the others.

Since the development of packet-switched radio networks in the early 70s, there have been significant advances in data communications, specifically in high speed local area networks, LANs utilizing coaxial cable as a shared medium. The majority of LANs serve a geographical area of one to two miles and mainly carry only one signal and generally provide high speed data transmission. In order to transmit signals over long distances the signal must be modulated, i.e., use a broadband medium. The broadband medium can carry multiple signals by using frequency division multiplexing on the same medium. Advantages of a broadband system include: allocation of bandwidth on demand, simultanenous transmission of data, voice and live video, high capacity of signalling rates, mass-produced cable and shared expenses. Examples range from a city, in which thousands of buildings and houses are tied into one wide area network [MCNA 83], also called Metropolitan Area Network MAN, through a university campus, in which thousands of dormitory rooms, labs, offices and up to several hundreds of buildings are connected via a single MAN [SHIP 82], to a company occupying a few buildings scattered across a large geographical area [MAZU 83]. LANs serve users in a limited geographical area, up to a couple of miles, whereas Long Haul Networks serve users that are hundreds or thousands of miles apart. Nonetheless, a long haul network, in other words a global network, can be formed by interconnecting several MANs through gateways or bridges. In the last couple of years, LANs have been extended to cover larger geographical areas up to 10 kilometers in radius. In order to span over such a distance the same media-access protocols with some performance tradeoffs are being employed over a broadband medium [ENNI 83]. However, it is unlikely that these protocols would perform as well over a broadband medium as over the baseband medium for which they were originally intended.

The permanent and virtual circuit-switching technique with store-and-forward capabilities is most commonly used in long haul networks where the flow control and routing problems are addressed rather than media-access protocols. These networks mainly support the bulk of point-to-point data traffic across the U.S.A. through the circuit-switched telephone network operating at the maximum of 56 kilobits per second. The operational range of these networks is extended around the globe via satellites. The present global network architecture does not require a media-access protocol, rather point-to-point communication links are established to provide full-interconnection among network nodes.

In satellite networks the frequency-division (FDM) and time-division multiplexing (TDM) techniques are commonly used. In general terms, a fixed-bandwidth is assigned to a single user or a certain portion of the bandwidth is allocated to multiple users on a demand basis. However, these networks generally support up to two hundred users that have heavy and continuous traffic requirements. The most typical property of Satellite Networks is that the one-way signal propagation delay is in the range of 125 milliseconds, therefore, LANs media-access protocols such as CSMA/CD, Token-Bus, etc., cannot be employed efficiently. If the traffic between two points is regular and continuous, Frequency-Division Multiple Access (FDMA) scheme performs very efficiently. In this technique network nodes communicate via carrier signals on which message signals are modulated as in radio and TV transmission. For point-to-point communications one or more channels are fixed-assigned. For multiple users sharing a frequency spectrum, each channel may be subdivided into an arbitrary number of subchannels that are dynamically allocated on a demand basis.

The Time-Division Multiple Access (TDMA) scheme performs well with user devices that have very regular information transfer requirements, in a high-duty cycle, since it can sustain high signalling rates under those conditions. However, there are some instances where TDMA may not meet information transfer requirements, therefore, further frequency-multiplexing may be needed before time-division is applied. In other words, TDMA is superimposed on the elementary FDM.

In the Reservation-TDMA scheme for satellite networks the channel is subdivided into multiple time slots. These slots are dynamically allocated on a demand basis. The reservation schemes are inevitably inefficient when network traffic load is low, and reserved slots will not be used for long periods of time between transmission bursts. However, if the time slots can be reallocated dynamically and if empty slots can be removed effectively, then the throughput can be improved. For those applications where guaranteed response time, high performance and stability are important, the reservation TDMA scheme with dynamic control would be ideal.

In packet-switched radio networks the following media-access protocols are used with limited efficiency: 1) Pure-Aloha, 2) Slotted-Aloha, 3) CSMA, and 4) Round-Robin with various priority schemes. All of these protocols are optimized specifically for bursty interactive data traffic such as terminal-to-computer, computer-to-computer, etc. In general terms, packet-switched radio networks serve a geographical area maximum of one or two hundred miles in radius, and the number of users supported on the network does not exceed a couple of hundred users.

The best media-access protocols for Metropolitan Area Networks, in which the signal propagation delay is inherently large, are not necessarily the same as for Local Area Networks which are optimized for specific applications and distances of a mile or two in cable length. The longer propagation delay of signals in MAN means that an extremely high price must be paid in data rate or efficiency if other parameters are unchanged. Several different media-access protocols have been developed for LANs by which a couple of hundred users are supported for one specific application, mainly data transmission.

In Carrier Sense Multiple Access with Collision Detection, CSMA/CD scheme, collision detection depends on the mutilated data packets existing anywhere on the medium [P803 85]. Therefore, the packet transmission time must be at least twice the maximum signal propagation time. Unless the packet size is made excessively long, an extension of the transmission system by a factor of X results in a speed reduction by a factor of X. In order to obtain the same throughput, given that the packet size is kept the same the data rate must be reduced by a factor proportional to the system size, otherwise there is considerable throughput degradation. Moreover, in wide area networks covering large geographical areas the collision detection is a much more difficult task than LANs. Generally, CSMA/CD provides very fast response under light load condition and sustains very good efficiency. Since it is a statistical method, as the load increases the average delay may exceed a threshold that is defined for proper packetized voice transmission, thus preventing the transmission of voice traffic [BRUT 83]. The stability characteristic of CSMA/CD protocol due to its non-deterministic nature should also be considered in certain applications such as digitized voice traffic, etc., where instability and delay variance cannot be tolerated. CSMA/CD provides no limit on the number of packets that can be transmitted between successive transmission rights given to a network station. That is, it has no intrinsic mechanism for guaranteeing a certain portion of the bandwidth to any station. However, if extrinsic restrictions are imposed on demands, then a certain amount of the bandwidth may be guaranteed for any station, in turn limiting the maximum packet length and requiring additional rules for higher level protocols to limit packet transmission rates. But this turns the system into a voice only network where bandwidth is allocated in small increments for long periods of time.

In the Token-Bus media-access protocol, a token is circulated to all nodes on the network, whether they have anything to send or not [P804 82]. Therefore, too much time is wasted by passing the token to all users in MAN where the propagation delay is quite large and the token must travel to the headend and then transmitted back in the forward direction. This scheme is also very sensitive to the number of users sharing the medium. As the number of users increases the average delay will also increase proportionally. In addition, the recovery procedures needed for lost-token conditions may be very complicated in MAN environment. Nonetheless, token-passing schemes, in general, behave well under load but introduce longer delays than CSMA/CD under light load [BRUT 83]. Since the token-passing schemes are deterministic, the average delay can be predicted, allowing easy handling of voice traffic, and the stability of the protocol is not an issue to be concerned with.

Token-Bus protocol also intrinsically provides a guaranteed bandwidth. That is, the worst case access time delay between transmissions for a given network station is determined by the time it takes all other stations to send maximum-size packets. On the other hand, the best access time delay is defined as the time between successive transmissions by a given station plus the time for the token to pass through all stations. Nonetheless, propagation on the medium adds at least 2T/N per station, where T is the one-way end-to-end signal propagation time for the medium and N represents the number of stations. Since the signal propagation delay of wide area networks is inherently large, the token-bus protocol performance would degrade drastically.

In the token-passing ring protocol, a station can start sending a token almost immediatelly after it starts receiving it. In a ring configuration, the delay for a station to decode the incoming physical signals, recognize a token, decide whether to change it to a start of frame, and re-code an outgoing signal stream can be kept minimal and this delay is relatively independent of the bit rate of the ring [P805 85]. Since this scheme allows much more time spent for data transmission rather than propagating the token, it can be considered for the integration of voice and data on the same shared channel. However, as the number of network users increases, the average delay will increase proportionally. For those networks covering a large geographical area, a break between any stations or any station failure on the ring will disrupt the network operation. Due to its topological ramifications, the token passing ring protocol may not be applicable to MANs.

A major difference between MANs and LANs is that LANs are owned by the same private organizations, therefore, no billing and control issues exist. On the other hand, MAN is considered a Public Network, in turn requiring centralized control of maintenance and billing. These functions are not provided by the peer-to-peer protocols used for LANs.

## 1.2 Emergence of Metropolitan Area Networks

A digital communication network serving a metropolitan area with a radius of one to twenty miles is called a Metropolitan Area Network, MAN. It is a compromise between Long Haul Networks like ARPANET, TYMNET, SATNET, TELENET, etc., and Local Area Networks like ETHERNET, CHEAPERNET, TOKEN-RING, STARLAN, NET/ONE, LOCALNET/20, MITRENET, etc. Communication media for MANs include broadband coaxial cable, fiber optics and radio.

By providing the capability of transmitting signals over long distances via fiber optics cable, coaxial cable, or radio transport media, the distribution, retrieval and exchange of digital information in any form in large geographical areas is made possible. The salient motivation for MANs is the desire to reduce the average costs to many distributed users by using intelligent components in the communication network systems to share transmission facilities and improve the overall performance [PHIL 84]. Fiber optic cable in its present technological state is an excellent transmission medium for point-to-point high-speed information exchange. Fiber has very low signal attenuation and does not radiate any electromagnetic energy, therefore, no eavesdropping is possible. However, fiber tapping, splicing and distribution is still very cumbersome and expensive. Fiber optics cable is ideal for a star topology, but not for a bus topology. Alternatively, radio networks generally require the stations to be light-of-sight, and multi-path problems exist in those areas which contain large and/or tall infrastructures. Furthermore, the station cost is prohibitive because of the hardware requirements and the transmission speed is limited due to confined bandwidth capacity. Coaxial-based Metropolitan Area Networks, by virtue of flexibility, easy distribution, and large area of coverage, are ideal candidates for the delivery of data, packetized voice and digitized/compressed video in a geographical area with a radius of up to twenty miles. However, it is expected that fiber optics cable will play a major role in the next generation of communication networks.

Today, CATV systems, an acronym for Community Antenna Television, are available in almost every city in the U.S.A., mainly for the delivery of entertainment services. Of the 80 million households in the U.S., about 55 million are currently passed by a coaxial cable, and about 35 million subscribe to cable services. It is projected that cable will pass about 80 million houses by the end of the year 1987, and serve 50 million of them [TJAD 83]. In residential and institutional CATV systems, several thousands of users in a metropolitan area share a common antenna by tapping onto a common coaxial cable carrying radio-frequency modulated signals. Only a certain portion of the bandwidth is being used for entertainment services, thus there is abundance of bandwidth available for value-added integrated services. With the advancement of data communications and the availability of ultra high speed transmission capabilities over Institutional Networks and

CATV facilities, delivery of data, voice and video to the business and home markets has become economically feasible.

The broadband medium has become the prefered choice for those applications being introduced in large geographical areas where a large population of users are connected to a communication network [GIBS 82, DINE 80]. Cities, university campuses, large office buildings, industrial facilities, and even small offices are being wired with broadband cable, because it provides a more flexible environment and allows the addition of forthcoming services such as integrated data/voice/video. The concept of combining data, digitized voice and video on a single shared medium will lead to the integration of these diverse information entities. Such integration would potentially yield a much more flexible information exchange milieu. Not only transportation of the information is important, but it is also crucial that the real-time structure and integrity of the information be preserved. The availability of off-the-shelf hardware coupled with reliable system operation in an extremely flexible environment has brought the broadband medium to the point where it is considered to be the backbone of the next generation of certain communication systems providing virtually error-free transmission [TOUG 83, MASO 83, TJAD 83, STAH 82].

## 1.3 Summary of Results

Several media-access protocols for packet switched networks are examined as applied to Metropolitan Area

Networks. Some of these protocols are intended for LANs and tailored for specific applications. The emergence of Metropolitan Area Networks using coaxial cable and fibre optics cable as a transport medium means that a new media-access protocol is needed to provide integrated services such as data, packetized voice and digitized video over a single shared channel.

The most significant properties of media-access protocols are determined by the services provided on the communication network. Metropolitan Area Networks by nature are intended for integrated services, namely the provision of conventional data communications, interactive transactional services, bulk data, multiplexed voice, compressed video, LAN interconnection, gateways, etc., therefore, development of a new media-access protocol for MANs became necessary. The development of such a protocol offered considerable challenge with its diversity of applications, topologies and load fluctuations.

The basic goal of this dissertation is to develop a new media access protocol for MANs over which thousands of network users can share the same physical channel, and diverse applications ranging from transmission of conventional data to compressed video images are supported. Various media-access protocols developed for local area networks, radio terrestrial and satellite networks are analyzed based upon the parameters that are important in MANs.

It has been realized that some of these aforementioned protocols may work well for certain services while performing poorly for other applications. Moreover, none of these protocols can accommodate all the services required on MANs [STUC 83]. The nature of MAN traffic demonstrates two extreme load conditions, namely bursty and/or continuous. It has been shown that contention schemes perform well with high bursty traffic, whereas the throughput, in other words channel utilization, may be degraded largely due to heavy traffic load. Besides the degradation of the throughput, variance in access time delay does not permit real-time voice communications in which the traffic load is also continuous [MAXE 82]. On the other hand, a TDMA technique in which delay variance is virtually eliminated performs effectively under high duty-cycle traffic [CAPE 79]. Therefore, it is the best media-access scheme for voice traffic but it does introduce longer access delays with a burst traffic load. Nonetheless, the tradeoff mechanisms between stability, access delay and throughput should be taken into consideration in the selection of media-access protocol for MANs. In light of these observations and presentations, it has been concluded that a load-adaptive, multi-mode media-access protocol (constructed as a combination of reservation and contention protocols) will be ideal for MANs supporting integrated services.

In Chapter II, the typical properties of Metropolitan Area Networks are summarized and various network topologies applicable to MANs are examined. Since the most significant properties of media-access protocols are in the main determined by the service types supported on the communication network, the salient service characteristics of MANs are described and a Service Matrix grouping data rate, required network performance, and traffic burstiness is presented.

In Chapter III, the salient functional specifications of a media-access protocol supporting data, digitized real-time voice and compressed video over MANs are described. Having defined the functional requirements for a MAN protocol provided the guidance and goals for the development of a new Demand-Assignment Media-Access Protocol.

In Chapter IV, a vigorous analysis of several media-access protocols developed for satellite, radio and local area networks is performed as applied to Metropolitan Area Networks. The analysis is based upon the following criteria:

1) Signal propagation delay impact on performance,
2) Large number of users affected,
3) Handling of integrated data/voice/video services,
4) Protocol parameters' effect on traffic handling characteristics,
5) Access delay vs. throughput vs. stability tradeoffs,
6) Performance sensitivity to fluctuating traffic in the channel,
7) Effect of overhead on channel throughput,

8) Deterministic and non-deterministic nature,

9) Fairness and guaranteed access,

10) Network topology applicability,

11) Robustness,

12) Network manageability and reliability.

It was concluded that none of these protocols can likely support the basic integrated services on a shared medium. As a matter of fact, a summary of comparative protocol analyses is given for various traffic models. By virtue of our discussions and presentations, it is concluded that a media-access protocol that is a combination of contention and reservation techniques can effectively and efficiently be utilized for Metropolitan Area Networks.

In Chapter V, the basic characteristics of the proposed media-access protocol are described in detail. A hierarchical network topology that is applicable to the proposed media-access protocol is developed and described. The implications of this network topology in conjunction with the proposed media-access protocol and intersegment bridging are studied. A protocol architecture similar to the OSI Reference Model is defined and their relationship is investigated. In addition to frame and subframe structures along with their components and the packet types, the establishment of a global time reference as well as frame synchronization and subframe slotting in time domain are described. The inherent signal propagation delay is an important performance factor in MANs covering geographical areas up to twenty miles in radius, therefore, propagation delay compensation and node organization across the network is considered and solutions to this problem are proposed. In a broadband transmission medium where frequency-division multiplexing is possible, the allocation and utilization of physical channels may play an important role in ensuring a reliable and maintainable network operation. A unique addressing scheme coupled with channel segmentation is proposed to allow stations to operate in a multi-channel system. A communication network with multi-service capabilities requires some form of priority assignment. In this proposed protocol, eight priority levels are defined and differentiation among priority assignments is accomplished without additional overhead in the channel. Static and dynamic channel-time reservation algorithms are developed and described in the context of the media-access protocol, and datagram transmission without reservation is explained. Two new dynamic retransmission backoff algorithms are proposed and their applicability to the demand-adaptive media-access protocol is demonstrated. Finally, a complete state diagram of the multi-mode channel access mechanism is provided.

In Chapter VI, we present our findings regarding the media-access schemes that are applicable to MANs and conclude with the contributions made with this research. In addition, we also recommend what future research work would be needed to verify the correctness of the proposed media-access scheme through computer simulation and/or real-time reconfigurable network testbed facility.

In summary, the major contributions of this dissertation are:

1) Definition and characterization of Metropolitan Area Networks including network properties, topologies and service requirements are described.
2) Several media-access protocols intended for satellite, radio and local area networks are analyzed and scrutinized as applied to MANs. The reasons why these protocols may not all be applicable to MANs are stated and the type of a media-access protocol that can be used in MANs is concluded to be a load-adaptive, multi-mode access scheme.
3) MAN protocol functional specifications are defined and a new hierarchical communication network topology and protocol architecture for a MAN is proposed.
4) A new media-access protocol that meets MAN service requirements is proposed, developed and described in detail.

In conclusion, we believe that the proposed media-access protocol is an optimum scheme for MANs supporting integrated services ranging from data packetized voice communications to digitized/compressed video communications. This research also emphasizes the fact that integrated services require a multi-mode media-access protocol rather than a single-mode protocol like CSMA/CD. This research could lead to several protocol implementations for Metropolitan Area Networks, not to exclude satellite, radio and local area networks, providing integrated services as a public network.

# CHAPTER II

# METROPOLITAN AREA NETWORK CHARACTERISTICS

## 2.1 Metropolitan Area Network Properties

The rapid proliferation of data communications has created several types of Metropolitan Area Networks that have ultra high-speed transmission capabilities and span a geographical area of several miles in radius. A Metropolitan Area Network, MAN serves a community in its proximity as a public network and has capabilities of delivering data, voice and video services via a single conduit cable, be it a coaxial or a fiber optics cable.

Broadband transmission of digitally encoded signals represents the only logical method to distribute a broad range of services to a large population of users; this is accomplished by averaging costs across many users sharing transmission facilities, thus providing overall improved utilization. MANs, in general, provide the following communication services in a unified manner: computer-to-computer, terminal-to-host computer, facsimile, digitized real-time voice communications, digitized/compressed video, etc. Examples of MANs include, i.e., a city wide network in which thousands of houses, buildings, etc., are tied into a single broadband cable; a true campus-wide network, in which several hunderds of buildings, labs, and dormitory rooms are interconnected via a ubiquitous medium, essentially via coaxial or fiber optics cable; a company network scattered over several buildings across a large geographical area [SMIT 79, TJAD 83]. These examples illustrate several different communication network requirements such as data, voice and video which may exist in analog or digital form [PHIL 84]. MANs are perceived to be one of the alternatives to provide integrated services within intercity limits, eventually interconnecting LANs and forming an extended segment of a nation-wide network meeting the requirements of a long-haul public communication network [COOK 84].

The salient characteristics of a MAN are summarized as follows:

- Span a geographical area of one to 20 miles in radius,
- Provide very high-speed transmission with low bit error rate,
- Support integrated services such as data/voice as well as video,
- Permit backbone network architecture for LAN interconnects,
- Provide very high bandwidth capacity, allowing frequency-division multiplexing,
- Introduce large signal propagation delay into the transmission,
- Allow easy distribution to a large population of users,
- Support several network topologies including bus, star, ring and tree-and-branch,

- Provide public network service environment,
- Allow centralized maintenance, control and billing.

Communication media for MANs include broadband coaxial cable, fiber optics cable and radio. A broadband medium, in particular, provides the transport mechanism to carry information longer distances than does a baseband medium. This is accomplished by modulating the baseband signal at radio frequencies. This modulation permits the signals to travel greater distances by use of signal amplification devices resident in the transport medium. Baseband media cannot be used for MANs because of their limited bandwidth capacity, and their area of coverage which is confined to a maximum one or two miles radius [KRUT 81].

Today, LANs are defined to serve a geographical area of a maximum of five miles in diameter. Although there is an overlap in the area covered by LANs and MANs, LANs are designed for data, perhaps packetized voice, services in a limited geographical area, whereas MANs may provide integrated services including interconnection of LANs in a relatively large geographical area over a single shared medium. Alternatively, telephone companies are planning to install digital transmission systems to support the Integrated Services Digital Network (ISDN) which will allow simultaneous use of the network for both switched voice and switched digital data applications [KOST 84, BHUS 84]. A MAN supporting integrated services should be capable of acting as an extension of the ISDN and also be capable of meeting all the requirements.

Integrated fiber optics distribution systems are being installed mainly in high-density metropolitan urban areas to provide point-to-point and point-to-multipoint transmission capabilities. Fiber optic cable in conjunction with wavelength multiplexing can support multiple integrated services in a wide geographical area. Moreover, fiber optics offers broad bandwidth, low signal attenuation and many other unique features. These features, anticipating its low cost in the near future, make it a very promising transmission medium for applications ranging from long-haul transmission and inter-office trunking to local end-user distribution. The present state of fiber optics technology presents local distribution with an economic challenge. This is due to difficult and costly tapping, splicing and distribution properties of the fiber optics cable [TOUG 83]. Today, fiber optics is being considered as the infrastructure for a Metropolitan Area Network. In other words, the fiber optics cable is deemed an interconnect medium between MANs. Fiber optic cable is being used for trunking purposes because of its aforementioned properties. However, broadband coaxial cable is utilized in end-user distribution systems. Fiber optic cable is the preferred medium for point-to-point high-speed transmissions and for those applications where electromagnetic radiation, which could facilitate eavesdropping, is not allowed. The local telephone companies are contemplating replacing the conventional twisted-pair wire with fiber optics cable in the near future, allowing ISDN services supported on the same medium in conjunction with voice communications. In most cases fiber optics cable lends itself to star network topology to fully interconnect network devices including computer mainframes, terminals, teleconferencing devices, voice equipment and interactive video devices. It is conjectured that it will take at least three to five years before fiber optics technology reaches the state at which it becomes more economical and advantageous than broadband coaxial cable for the construction of MAN physical transport media.

The radio transmission medium is generally ideal for the distribution and retrieval of digital information in a line-of-sight environment. Radio-based communication networks can easily serve a wide geographical area because radio signals can travel longer distances and can readily be received by line-of-sight stations and those scattered across a large geographical area. However, due to multi-path problems in areas containing large and/or tall infrastructures, the quality of signal reception and the reliability of the information received are of concern. Radio medium offers broad bandwidth, low attenuation and other unique advantageous such as allowing a network station to move freely within its area of coverage. Since the signals are transmitted over the air, no additional physical medium is required to carry the signals over long distances. Even though the transmission medium is free, the use of the radio spectrum is regulated and, to a certain degree, restricted by the FCC. Due to radio medium's low signal-to-noise ratio and multi-path problems, the bit error rate (BER) would not be expected to be better than one in one thousand bits. Therefore, vigorous error correction is necessary in radio networks, in turn increasing the overhead and consequently decreasing the throughput. Under the best of conditions, including error correction, the radio medium may not provide reliable information exchange because of line-of-sight requirements and a high bit error rate. The BER required for some integrated services supported by a MAN may not be maintained, and the cost of a network station for a small user with also bursty and minimal data requirements may be prohibitive. Thus, the use of radio medium for MANs would be both restrictive and expensive as compared to using coaxial and fiber optics cable.

The availability of off-the-shelf CATV hardware coupled with reliable system operation in a variety of environments has made coaxial cable the preferred prime physical transport medium for Metropolitan Area Networks. It is also shown that a broadband medium has economic advantages over the conventional twisted-pair wire-based switching approach and baseband coaxial cable [COOP 83, BRAN 83, COOK 84, KRUT 81, MOKH 84, SELL 82].

Today, CATV systems, acronym for Community Antenna Television, are being made available almost in every city in U.S.A., mainly for the delivery of video entertainment services. In addition to residential two-way cable systems (FCC mandated that all cable plants built after 1975 be two-way capable), institutional plants are being constructed as part of franchise requirements. In residential and institutional systems several thousands of users in a metropolitan area share a common antenna by tapping onto a coaxial cable carrying radio-frequency modulated signals. Broadband coaxial cable is being used in Local Area Networks where the area of coverage is limited to five kilometers [ENNI 83, DINE 80]. After exploiting the revenue potential of video entertainment services, cable operators are preparing to deliver data services, commencing with interactive video applications and then moving into more sophisticated value-added applications such as PBX and LAN interconnect, computer-to-computer and computer-to-terminal communications, etc., [GERR 80, TJAD 83, SIRA 84, MCGA 83, MCNA 83]. These systems represent the forefront of the coaxial-based Metropolitan Area Networks that could eventually be the catalyst for dramatic growth in integrated communications services.

Coaxial-based MANs using CATV facilities generally utilize a tree-and-branch topology. Signals emanating from the central distribution point (headend) are referred to as "Downstream" (in the parlance of CATV industry other names used include "Outbound" and "Forward"), and are of broadcast type, that is, all stations on the network listen to the same ubiquitous information. Therefore, no media-access arbitration mechanism is required in the forward direction. Conversely, signals transmitted by the network stations are designated as "Upstream" (in the parlance of CATV industry other names used include "Inbound" and "Reverse"), and are of multi-access type. Therefore, a multi-access protocol is required to arbitrate the channel access among network stations in the reverse direction [COOP 83].

The uni-directional system architecture of the coaxial-based networks is similar to that of radio-based networks. Namely, receive and transmit frequencies generally are different, hence, a frequency translator is required either at the cable headend or at the radio transponder to establish a virtual-bus type network medium. The frequency translator unit translates upstream signal frequencies to downstream signal frequencies. It should also be remembered that the network topology plays a dominant role in determining the media-access scheme that can effectively and efficiently be applied to MANs.

In summary, the reasons broadband coaxial cable is preferred over other media for MAN implementations are given below:

- Shared medium, supporting video and voice as well as data services simultaneously,
- Multiple distinct services,
- Large bandwidth,
- Long distance,
- Dense area coverage,
- Prewired facilities possible,
- Flexibility for future adaptation to new user needs,
- Low-cost implementation.

Table 2.1. Comparison of Transmission Media for Metropolitan Area Networks

| Media | Broadband Coaxial Cable | Fiber Optics Cable | Radio And Microwave |
|---|---|---|---|
| Bandwidth | High | High | High |
| Ability to Handle Many Nodes | High | Low | Low |
| Distance | High | Very High | Very High |
| Topological Versatility | High | Moderately Low | Low |
| Installation Ease | High | Moderate | Moderate |
| Noise Immunity | High | Very High | Low |
| Cost | Low | Moderate | High |

A survey of relative comparisons of MANs by transmission media is given in Table 2.1 [PARK 83]. By virtue of the attributes stated above, the coaxial-based broadband systems have a great potential to emerge as the backbone of a MAN providing transmission of information of any kind, namely data, voice and video, in any form.

## 2.2 Metropolitan Area Network Topologies

When a Metropolitan Area Network architecture is under consideration one of the several design parameters of interest is the network topology. The traffic distribution, load balancing, delays, stability, throughput, reliability, maintainability and other communication efficiencies are influenced by the communication network topology. The topology is described as the pattern of interconnections used between the various nodes of the network. This interconnection process is highly dependent upon the medium in question. For example, in local telephone switching systems customer premises equipment is connected to the central switch via a twisted-pair wire, thus implementing a true star topology. In another case where coaxial cable is the physical transport medium, a bus topology can be created with the use of a frequency translator at the headend. These examples illustrate that the topology is determined by the characteristics of the media, distribution needs, economy of scale, bandwidth capacity, and area of coverage.

This disseration is only concerned with coaxial and fiber optics media and related topologies. As stated before the network topology plays a major role in determining the media-access protocol, therefore, we will only explain those topologies that are being used or being considered for Metropolitan Area Networks.

### 2.2.1 Tree-and-Branch Topology

Because of ease of distribution, flexibility, expandable capacity, capability of connecting several thousands of devices, and operability over greater distances, broadband coaxial cable lends itself to a tree-and-branch topology, the cable plant fans out from the main distribution point (headend) in a tree fashion, splitting and resplitting until the maximum area of coverage is reached. To allow signals to travel greater distances amplification devices are required in both directions on the distribution network. Tree-and-branch topology is characterized by the fact that both the upstream and downstream signals travel over the main distribution cable (trunk) and its branches (line extenders), thus allowing all network stations to be connected across the medium. The bandwidth of a network is shared among all network users, allowing everyone to receive and transmit signals on all channels. All network nodes are fully interconnected, failure of any node on the network does not interrupt the network operation. Moreover, it also permits expansion without interrupting network services. The result is that network users may connect or disconnect their network interface units to/from the network while others are operating.

In a tree-and-branch topology, the downstream (forward) signal path originating from the headend is broadcast type and all network stations receive the same signals at different times depending upon their distance from the headend. Conversely, the signals generated by the network stations merge through the branches and trunk cables at the headend, forming a multiple access medium. The information exchange between stations is accomplished by transmitting on the upstream channels. Signals traveling to the headend are retransmitted in the forward direction, and received on the downstream channels. The upstream and downstream channels are paired so that if a node transmits its data on an upstream channel, then it should also know the downstream channel on which its data would be heard. This topology is similar to that of satellite networks where the signals are transmitted to the satellite transponder and are echoed back to all listening network stations on the earth. A network station transmits data to another station connected to the same network, by transmitting the data in the reverse direction to the headend; the signals are then echoed back in the forward direction, allowing all stations to listen to any data addressed to them.

As shown in Figure 2.1, the signals flow in both directions: the downstream channels are filtered and amplified in the forward direction whereas the upstream signals are filtered and amplified only in reverse direction. This system architecture transforms the coaxial cable into a uni-directional medium [STAH 82].

The frequency spectrum of a coaxial cable is partitioned into two main subchannels: 1) downstream channels, and 2) upstream channels whose bandwidth is determined by the density of the expected traffic in both

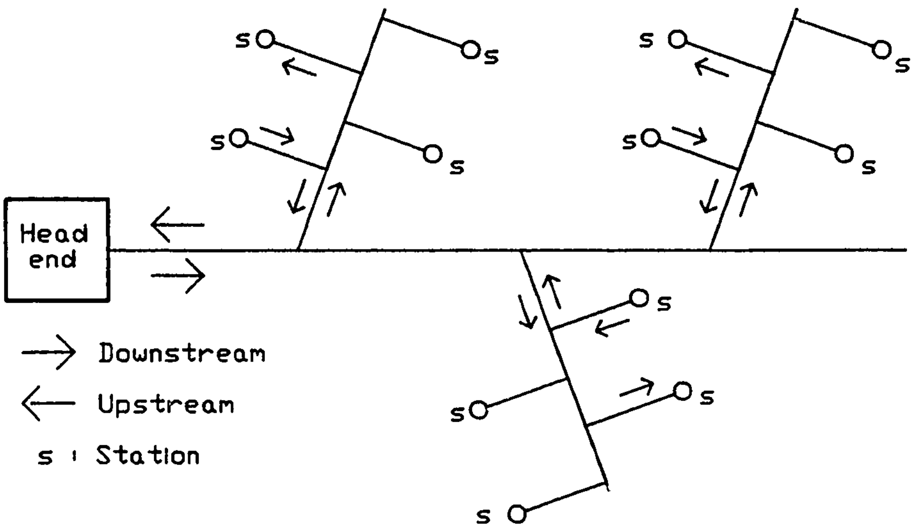



Figure 2.1. Tree-and-Branch Topology

Table 2.2. Frequency Spectrum Allocation of Coaxial-Based Broadband Networks

| System | Downstream Channels | Upstream Channels |
|---|---|---|
| Sub-split | 54 to 440 Mhz | 5 to 32 Mhz |
| Mid-split | 168 to 440 Mhz | 5 to 116 Mhz |
| High-split | 234 to 440 Mhz | 5 to 174 Mhz |

directions. Coaxial-based broadband networks are built in three different configurations depending on the services they deliver. The symmetry of the downstream and upstream traffic determines the network configuration. These network configurations and the associated frequency allocation chart is given in Table 2.2, [SIRA 82a, GIBS 82]. Residential CATV systems are constructed using the sub-split system configuration because they are mainly intended for video entertainment services. Alternatively, institutional CATV systems and the majority of coaxial-based MANs being built with mid-split and high-split configurations, are intended for the delivery of data, voice and video services which possess more symmetric traffic characteristics.

Today, almost all coaxial-based MANs use the tree-and-branch topology because it provides the most economical solution for proper signal distribution and greater area coverage for a large number of users [BRAN 83]. In this dissertation we propose a new media-access protocol that can easily be applied to tree-and-branch topology. The emphasis is primarily on tree-and-branch and similar topologies because of the abundance of bandwidth available in CATV systems already built in almost every city and metropolitan area in North America.

### 2.2.2 Segment-Switched Tree-and-Branch Topology

The segment-switched tree-and-branch topology differs slightly from the tree-and-branch topology by having a centrally controlled segment or hub switching capability over the upstream channels. The downstream channels still have broadcast type transmission characteristics whereas the upstream channels are segmented across the network. This topology has exactly the same characteristics of the tree-and-branch topology including frequency spectrum allocation and network configuration. It should be noted that no routing is required with this topology, all messages pass through the headend before they are transmitted over the broadcast downstream channels. Similar to the tree-and-branch topology, this network topology also provides full connectivity among all network nodes, and any network node failure does not pose any threat of service interruption. Moreover, it still allows network expansion without any impact on the network architecture.

As is illustrated in Figure 2.2, the network segments are controlled by an intelligent controller residing at the headend. These network segment switches can be in one of the following three states: 1) OPEN, no upstream signals from that segment are allowed to flow into the trunk cable and travel to the headend location, and 2) CLOSED, all upstream signals are permitted to merge into the trunk cable and flow toward the headend location in the reverse direction, and 3) ISOLATION, 6 dB impedance is placed in circuits not selected, thus reducing reverse channel noise [MAXE 83]. Basically, these switches are addressable bandpass filters, passing only the frequency spectrum determined by the filters, and they are also used to locate and isolate faulty segments of the reverse path. In some cases, these switches are used to provide a means for an arbitration mechanism in the reverse path, namely spatial multiplexing. The coaxial-based MANs spanning a large geographical area may also act as a perfect antenna for any undesirable signals [SIRA 84].

This is because of the merging tree type of network architecture of the reverse path. Therefore, all CATV plants are equipped with bridger switches (segment switches) to maintain smooth communications network operation even if one or more segments are mutilated by some noise sources or any other intruding signals. These faulty segments are addressed by the headend to switch to OPEN state as such the upstream signals may not get into the trunk cable. The network stations located in these faulty segments can still listen to the broadcast signals transmitted on the downstream channels, but no messages would pass through the segment switches in the reverse direction.

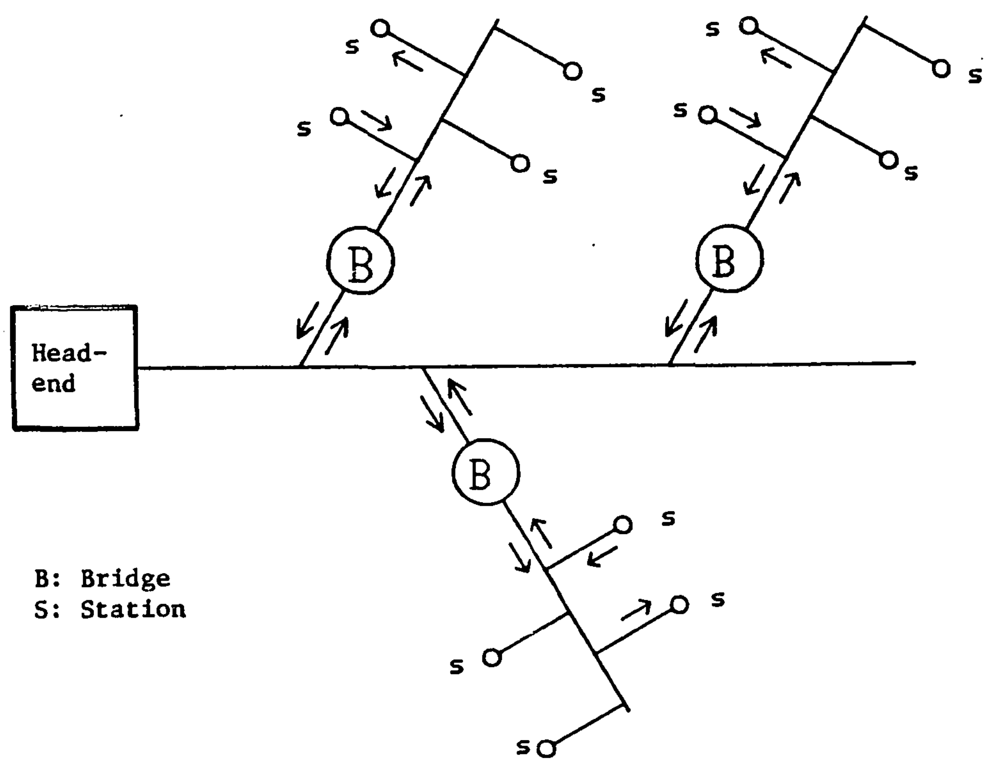



Figure 2.2. Segment-Switched Tree-and-Branch Topology

One disadvantage of this topology is the requirement for a centralized control to manage the network segment switch enclosures. However, an understanding of the practical problems of this topology indicates that it is still good practice to partition the network's upstream channels into segments so that the faulty ones can be located and isolated. From the network management stand point, this topology is highly recommended, however, the implications of a single headend segment controller that is vulnerable to single-point failure must be considered.

### 2.2.3 Bus Topology

The bus topology typically attempts to eliminate the central distribution node on the network, without sacrificing the simplicity of the other nodes. The elimination of the central node does imply a certain complexity at the other nodes of the network, but a decentralized network can be constructed by allowing users to tap onto the trunk cable rather than onto the drop cable. Broadband coaxial cable lends itself to bus topology. Bus topology provides full connectivity among network nodes, and any network node failure does not interrupt network services. Moreover, network expansion is possible while the network is operating [KINN 82].

As shown in Figure 2.3, a true bus architecture is obtained by installing a non-intelligent frequency translator at the headend. The frequency translator shifts the upstream channels up to the downstream channels, permitting the same information transmitted on the upstream channels to be translated on to the downstream channels without any data manipulation, and maintaining the integrity of the information in a unified manner.

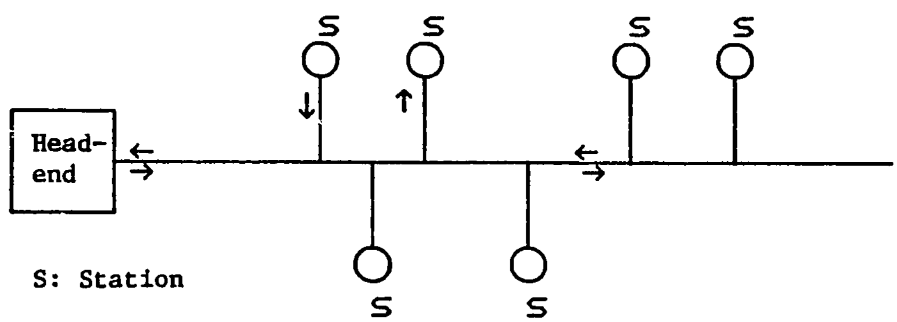



Figure 2.3. Bus Topology

No routing decisions need to be made since any node on the network may communicate with others by transmitting on an upstream channel which is frequency translated at the headend, and then listening to the downstream channel [CLAR 78]. In order to implement a virtual bus architecture, the upstream channels are paired with the downstream channels. Network stations can only transmit in the reverse direction and receive in the forward direction. This topology is used frequently in Institutional CATV systems delivering integrated services to schools, banks, government agencies, etc. When the network traffic is more or less symmetric in both reverse and forward directions and a small number of users are to be served, the bus topology is preferred over others [SIRA 82a]. However, as the number of users to be served increases, signal distribution using bus topology becomes more cumbersome and expensive. Therefore, the tree-and-branch topology should be considered for interconnecting a large number of users scattered across a large geographical area. Directly tapping onto the trunk cable requires no splitters, this reduces cable losses and also generates low bit error rates because of the lower probability of intrusion from the noise sources and low degree of dentricity (branching like a tree). In view of these considerations, the bus topology is recommended for those networks providing LAN interconnects, moving large amount of data between stations, and serving a small number of users.

**2.2.4 Local-Switch Star Topology**

The star topology typically eliminates the need for each network node to make routing decisions, and localizes message routing in one central node. This leads to a particularly simple structure for connecting a small number of network nodes. This topology is an obvious choice if the typical traffic pattern in the network conforms to its physical topology, as the case when all network nodes are required to communicate with the central node. For example, in time-sharing systems where the central node is the time-sharing machine itself, the star topology is a very likely choice. However, when the traffic pattern is not between one primary node and several secondary nodes, but rather involves communications among all of the nodes, then the star topology appears at a disadvantage. Clearly, network operation depends on the proper operation of the central node, which performs the routing functions, and must have capacity to cope with simultaneous conversations on all nodes. The telephone switching system uses switched-star topology, simply the equipment from each customer premises is connected to a central switch through a twisted-pair wire, and the circuit switching function is performed by the central switch itself. As illustrated in Figure 2.4(b), the topology may represent a central telephone office system in which subscriber connections are implemented in star fashion. In this type of topology failure of the central node may cause service interruption across the network whereas failure of a segment node may only impede the communications on the associated segment. Moreover, full connectivity and message broadcasting are more difficult to achieve with this topology. Contrary to the previously discussed network topologies, network expansion is limited and in some cases it is not economically and technically feasible.

Even though broadband coaxial cable is well suited for a star topology, fiber optics cable obviously is the preferred alternative because of the ease of distributing fibers in star fashion. Today, the main trunking for the majority of long-haul communication networks is implemented with fiber optics cable, while coaxial cable is still utilized for local distribution. It has been shown that selection of broadband star topology over tree-and-branch topology is highly dependent on the number of stations to be supported by the network [BRAN 83].

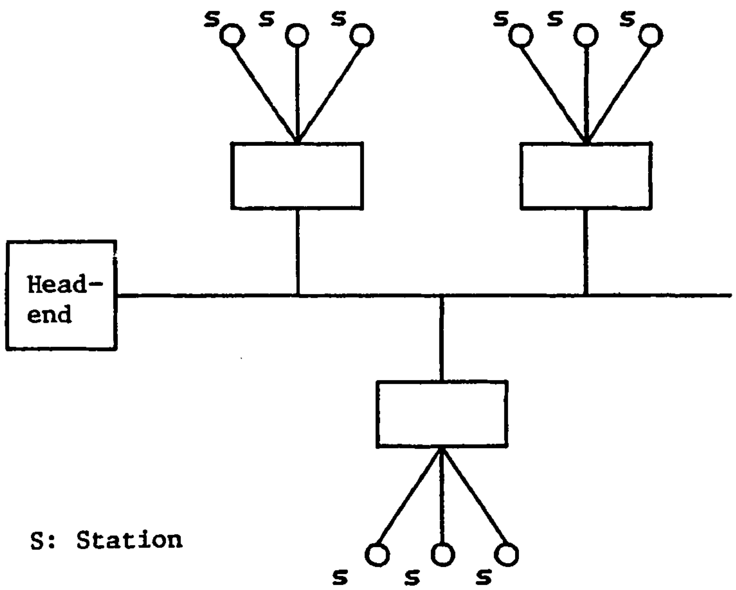



Figure 2.4(a). Local-Switch Star Topology: Bus-Star Configuration

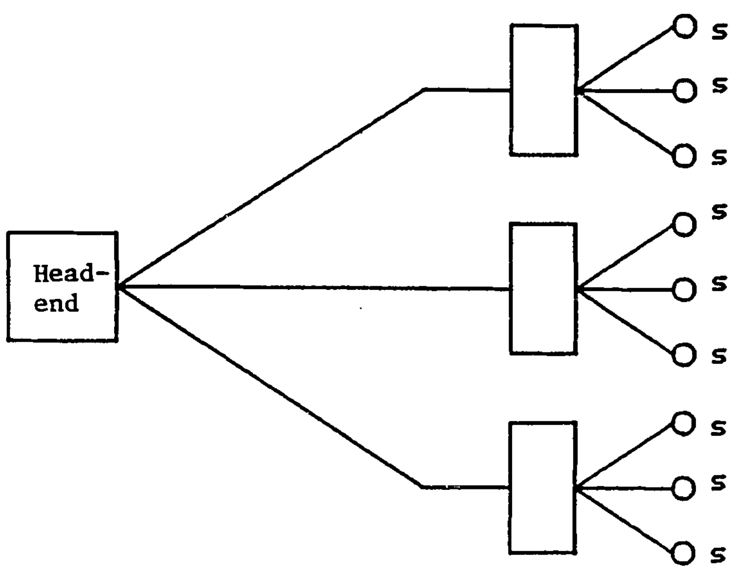



Figure 2.4(b). Local-Switch Star Topology: Star-Star Configuration

As illustrated in Figure 2.4(a&b), the star topology intrinsically has a two-level architecture. It is also referred to as a local-switch topology in which each central node is an intelligent device, and switches messages generated in the sub-network and is responsible for routing and gatewaying over the backbone network. The backbone network may take two different forms: 1) bus topology, and 2) star topology. In the local-switch star topology using bus architecture as a backbone, refer to Figure 2.4(a), the headend could be an intelligent root node or simply a frequency translator forming a virtual bus topology for the rest of the local-switch star sub-networks. In order to simplify the network management functions and to maximize the system throughput, the local-switch star topology provides an ideal architecture, but it may be costly in terms of network implementation, requiring distributed intelligence and long strings of cable. Due to the difficulty of installing and maintaining sophisticated and intelligent devices in a hostile environment, as in the case of a Metropolitan Area Network spanning several miles, this topology, technically and economically, may not be a sound selection. However, with the recent revolutionary developments in fiber optics technology, this topology could be superior to other topologies within three to five years.

### 2.2.5 Ring Topology

The ring topology also attempts to eliminate the central node on the network without sacrificing the simplicity of the other nodes. In this topology, as illustrated in Figure 2.5, a message is passed from node to node along unidirectional links. There are no routing decisions to be made in this topology; the sending node simply transmits its message to the next node in the ring, and the message passes around the ring one node at a time until it reaches the node for which it is intended. The only routing requirement placed on each node is that it be capable of recognizing the messages addressed to it.

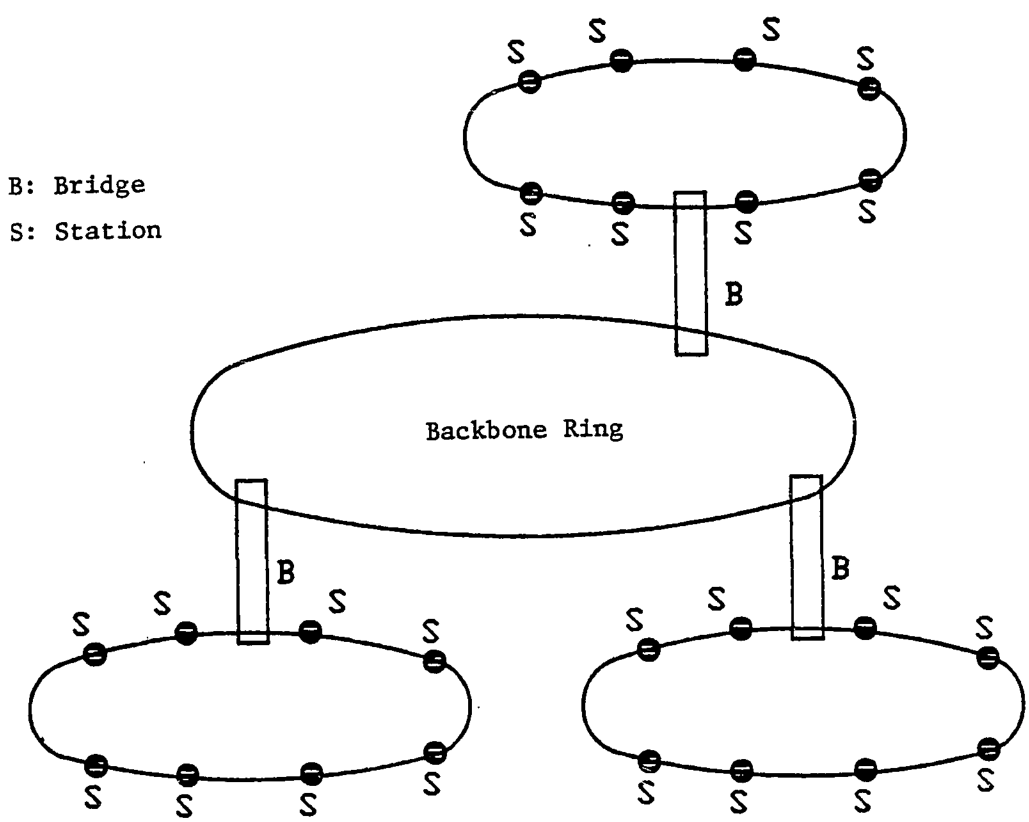



Figure 2.5. Ring Topology

Ring networks basically consist of a circular chain of signal repeaters with cable links between each repeater. Full connectivity and message broadcasting are easy to achieve with ring topology, however, any failure of the network nodes will disrupt the total network operation unless a bypass mechanism is provided. In this topology, no central node is required, but any one of the network nodes should be capable of acting as a network monitor. Twisted-pair wire and fiber optics cable lend themselves to ring topologies. In particular, fiber optics cable is very effective in connecting network stations in a point-to-point fashion. Metropolitan Area Networks can easily be constructed with the use of multiple ring networks connected via a backbone ring network. Fiber optics cable is generally the medium used for this type of network configuration. As it is shown in Figure 2.5, the ring sub-networks are interconnected through bridges to extend the area of coverage. It is obvious that since a single point of failure on the network may shut down the total network, this topology may not be the best approach for implementing a reliable Metropolitan Area Network to support a large number of uesrs for time-constrained applications.

## 2.3 Service Characteristics and Requirements

The throughput of a media-access protocol is highly dependent on the load characteristics, message length, the number of users being supported by the network, and other factors. However, the most significant properties of media-access protocols, in general, are determined by the types of service that are to be provided on the communication network.

If we assume that broadband LANs and MANs are to provide integrated services, then several types of devices such as dump terminals, peripherals, computers, and voice and video equipment, are likely to be connected to the communication network [ARTH 82].

Considering the various services to be provided and the number of users to be served by a MAN, the load conditions inevitably would fluctuate between low-duty and high-duty cycles. As it is illustrated in Table 2.3, an interactive user terminal requires a small amount of bandwidth, but introduces very bursty traffic to the network, whereas a file transfer requires a large amount of bandwidth for a short period of time. Alternatively, voice traffic requires a fixed amount of the bandwidth for an extended period of time, thus introducing a periodic traffic load to the network. For the transmission of real-time or compressed video signals or even single photographic image frame transmission, bandwidth requirements vary over a couple of orders of magnitude. As it is summarized in Table 2.3 the ratio of highest to lowest peak data rate is over four orders of magnitude; it is, therefore, obvious that no single media-access method is appropriate for all the services listed in Table 2.3.

Table 2.3. Workload Generated from each Source Type

| Type of Source | Peak Signalling Rate(KBPS) |
|---|---|
| Heat/Ventilation/Alarm/Security | 0.1 |
| Line Printer | 19.2 |
| File Server/Block Transfer | 20,000.0 |
| File Server/File Transfer | 100.0 |
| Mail Server | 100.0 |
| Information Server/Calender | 9.6 |
| Information Server/Decision Support | 56.0 |
| Word Processor | 9.6 |
| Data Entry Terminal | 9.6 |
| Data Enquiry Terminal | 64.0 |
| Program Development | 9.6 |
| Laser Printer | 256.0 |
| Facsimile | 9.6 |
| Voice/Immediate | 64.0 |
| Voice/Store and Forward | 32.0 |
| Video/Noncompressed | 85,000.0 |
| Video/Freeze Frame | 64.0 |
| Video/Compressed | 400.0 |
| Graphics/Noncompressed | 256.0 |
| Graphics/Compressed | 64.0 |
| Optical Character Reader | 2.4 |
| Gateway | 1,000.0 |
| Host/0.5 MIPS | 128.0 |
| Host/5 MIPS | 1,000.0 |

The prowess of MANs lies with high bandwidth, low average access delay, and high reliability. In some type of communication systems these factors are qualitative rather than performance measures. The tradeoffs between delay, throughput, and stability cannot be neglected in determining the total qualitative and quantitative performance of the communication network being considered. The following paragraphs dwell on the typical requirements for data, packetized voice and digitized/compressed video in terms of delay variance, tolerance of error, nature of traffic and access demand.

In any media-access protocol, the average access delay has always been of prime interest. The average access delay is highly dependent on the throughput and the stability of the media-access protocol. In other words, we may sacrifice the throughput in favor of delay and stability or vice versa. This means that decreasing the throughput will improve the average access delay and stability. The close relation between throughput, average access delay and stability allows us to optimize the network performance in accordance with network services. We can determine and predict the average access delay if other network performance parameters such as the maximum expected throughput and the degree of stability are given. The access delay for a packet may vary from one media-acess protocol to another. Hence, a variance of delay may be observed on a communication network depending on various factors such as expected throughput, traffic load, desired degree of protocol stability, number of stations being supported on the same channel, etc.

From a transmission point of view, the transfer of digital speech is less sensitive than analog data to noise, cross-talk and distortion, and faded signals that can readily be regenerated without introducing cumulative degradation [SPIL 80]. Typically, the packetized voice is based on sampling an analog waveform 8000 times per second, once every 125 microseconds, and converting each sample into an eight-bit word, hence, the bandwidth required is 64 KBPS. This bandwidth requirement has been reduced to 32 KBPS through use of standard compression techniques and is expected to be lowered further within a couple of years [MAXE 82]. Once the samples are digitized they are buffered and packetized for transmission.

The more voice samples included in a packet, the greater the delay between the time a sample is generated and the time it is delivered to the receiving destination. To achieve interactive speech and smooth playback operation, it is important to keep the end-to-end delay for most bits of voice information within tight bounds [TOBA 82a]. Moreover, to insure proper packetized voice transmission, channel access should be granted within 750 microseconds, (maximum six samples may be buffered before transmission in a single packet), otherwise the integrity of voice packet transmission cannot be guaranteed. It has been indicated that 99 percent of voice data should be delivered within the 5 milliseconds permitted delay, otherwise, noticeable degradation could be observed in voice quality [BRUT 83]. It is also important that the time stamp be used to order the voice packets (packets could arrive out of order) so that it could be played out in the proper time relationship to other packets. It is argued that if a small percentage of voice packets are discarded at random, the resultant distortion is tolerable [MAXE 82]. It has been shown that satisfactory packet speech transmission may be obtained if the probability of late or lost packets be kept in the order of 1% or less [FORG 80].

As it is shown in Table 2.4, the delay variance is not critical for data and digitized video communications, whereas in packetized voice communications the delay variance cannot be tolerated or should be constrained within certain bounds. The maximum delay allowed in current telephone connections is in the order of a few hunderd milliseconds. However, in a Metropolitan Area Network environment, the maximum access delay must be significantly less than this, since the connection probably would use an outside facility [MAXE 82].

Errors are state discrepancies between the transmitted and received bits of digitized signals. Different services have disparate bit error rate (BER) requirements. Even though MANs are assumed to provide a transport environment with a very low bit error rate, better than one in a billion, some kind of error protection may still be required. The quality and degree of this error protection is dependent on the service types that are being offered on the network. For example, as it is shown in Table 2.4, digitized video and packetized voice contain more redundant information and is more tolerant of bit errors than data transmission where low bit error rates are required. In the case of packetized voice, some packets may still be lost and the speech can be replayed without any perceivable distortion. It is recommended that a bit error rate of better than one in a million at 90 percent of one minute periods would be sufficient for good quality of interactive, real-time voice communications [GRUB 83].

Table 2.4. Data, Voice, and Video Service Characteristics

| Transmission | Delay Variance Tolerant | Error Tolerant | Periodic Traffic | Guaranteed Access Required |
|---|---|---|---|---|
| Data | YES | NO | NO | NO |
| Packetized Voice | NO | YES | YES | YES |
| Digitized/ Compressed Video | YES | YES | NO | NO |

In digitized video communications, since the analog waveform of video signals is sampled and digitized, quantizing error would be of greater concern than the transmission error rate. In straightforward video digitization, a pixel, the smallest entity of video information not including color, is represented with an eight-bit word. However, several compression techniques have been developed and are already in use to reduce the bandwidth required for the transmission of either live video or a single video frame. With certain types of compression techniques a pixel may be represented by one single bit with a high quality of video reproduction, thus allowing more video frames to be transmitted in the same channel. During the digitization and compression process a certain amount of redundant information is also included. If a certain number of digitized/compressed video packets are lost due to a noisy transmission environment, the end result, reproduction of video signals, may not show any perceivable degradation because of redundant video information included in the transmission. Therefore, in packetized video transmission, a bit error rate of one in a million would be adequate for acceptable reproduction of video signals. Conversely, data transmission is not tolerant of errors and requires a minimum one in a billion BER in any transmission environment.

There are basic differences between the traffic load requirements for data, packetized voice and digitized/compressed video. The voice packets are generated periodically and are considered to be fixed-size, whereas data and digitized video packets demand channel capacity sporadically or aperiodically. The media-access protocol optimized for aperiodic sources may not perform as well for periodic sources or vice versa. Therefore, some type of priority scheme may be required to accommodate a periodic traffic load while not introducing larger delays and unfair channel access to other sources generating aperiodic traffic load. The requirements for allocating network resources may vary from one service type to another. For example, in packetized voice communications, a certain amount of the channel capacity is required periodically, and network access should be guaranteed under all circumstances. Conversely, in data and digitized video communications, channel access need not be guaranteed. However, the average access delay should not exceed users' expectations because of periodic sources existing on the same shared channel. This implicitly proves that periodic sources should have higher priority than other sources such as data and digitized video, etc., to insure guaranteed access over the shared channel.

With the given diversity and volume requirements of applications, the typical service characteristics of Metropolitan Area Networks may be grouped under overall data rate and the burstiness nature of traffic. As it is shown in Table 2.5, the data rate, in essence, represents the required network performance. The quantitative measure of network performance is defined to be "Low" for data rates below 1 MBPS, and "High" for data rates in the range of several megabits, i.e., 10 MBPS. On the other hand, the burstiness nature of the network traffic is characterized as "Low" for continuous traffic load and "High" for low-duty cycle bursty traffic load. The Service Matrix for Metropolitan Area Networks is given in Table 2.5.

Table 2.5. Service Matrix for Metropolitan Area Networks

| Traffic Burstiness \ Data Rate | Low | High |
|---|---|---|
| Low | Conventional Data Communications | Bulk Data Multiplexed Voice Compressed Video |
| High | Interactive Transactional Services | LAN Interconnect Gateways |

In conventional data communications such as point-to-point transmission, facsimile, wire services, electronic mail, etc., low data rates would be sufficient and the traffic load introduced by these sources is expected to be periodic or continuous. Applications such as file server, information server, graphics, data entry and inquiry are categorized as interactive transactional services which require a small amount of bandwidth and introduce bursty traffic into the channel. Alternatively, applications such as bulk data transfer, multiplexed or packetized real-time voice, digitized and compressed video, interconnecting LANs, gateways and other similar services require high data rate, namely high network performance, though the network traffic wavers between two extreme bursty conditions. Gateways and bridges would need the channel capacity sporadically, however, a large amount of bandwidth capacity would only be required for a short period of time, in turn, total channel capacity would need be allocated for the duration of data transfer.

The service matrix defined for MANs, refer to Table 2.5, exemplifies the fact that the data rate requirement and network traffic characteristics for integrated services such as data, packetized voice and digitized video vary between two extreme boundaries. Therefore, a single media-access protocol would not be effective under these conditions. Therefore, we conclude that a media-access protocol that is dynamically adaptive to the burstiness nature of the offered traffic load and varying bandwidth requirements, is required to support the integrated services outlined in the service matrix.

# CHAPTER III

## FUNCTIONAL REQUIREMENTS OF A MEDIA-ACCESS PROTOCOL INTENDED FOR METROPOLITAN AREA NETWORKS

When a communication channel is to be shared among many users, an arbitration mechanism known as a media-access protocol is used to allocate channel capacity for every user of the network on a demand basis. The rules common to all stations should be followed strictly by all network users so that no one station may hog the network, thus providing equal access to all users. In a packet-switched network, the delay, throughput and stability measures of a media-access protocol are highly dependent upon the traffic load characteristics, message length, signal propagation delay, number of users to be supported, etc. In this chapter, we will define typical functional requirements of a media-access protocol intended for Metropolitan Area Networks. It should be noted that these requirements stem from the need to support integrated services, namely data/voice/video, on the same shared medium. In the following paragraphs we will dwell on the salient functional specifications of a media-access protocol intended for Metropolitan Area Networks supporting integrated services.

### 3.1 Data, Digitized Voice and Video Services

An integrated digital communication network, by definition, is perceived as supporting a wide range of services such as data, packetized voice, and digitized and/or compressed video. These services inherently have disparate characteristics such that a media-access protocol that works well for one type of service may not perform as well for others. Given the requirement that a Metropolitan Area Network should provide all integrated services, the media-access protocol intended to support this wide range of applications should allow the network stations to share the communication channel with the mix of traffic generated by the gamut of devices. These devices include data communication stations, facsimile stations, voice vocoders, video frame storage equipment, PBXs, etc. In other words, the traffic introduced into the channel consisting of conventional data, multiplexed and packetized voice, compressed video frames, etc., shall be carried on the same frequency spectrum allotted for the communication network. Briefly, the media-access protocol envisioned for Metropolitan Area Networks should be capable of handling data, packetized voice and digitized video simultaneously on the ubiquitous communication channel.

### 3.2 Large Population of Users

In a Metropolitan Area Network the medium is shared among several thousands of users rather than several hundreds as the case for a Local Area Network. Considering a large user base in a Metropolitan Area Network environment, there may exist distinct types of users generating varying degree of traffic loads. Basically, there are three types of users: small, average, and large. The small users are assumed to be generating traffic lower than the average user, and require buffer space for only one packet. The large users are expected to generate large amounts of traffic exceeding that of the average user and they require large buffer capacity. In a Metropolitan Area Network environment there would be a considerable number of large users, but their overall impact on the average traffic still would be negligible according to the "Large User Law"; that is, the average traffic introduced into the network, with approximate probability, is equal to the sum of the average traffic generated by a large population of users [LAM 74]. If we apply the "Large User Law" to a MAN environment we can state that the average traffic offered to the network by a large population of users may represent a continuous stream of information and, therefore, the MAN media-access protocol should be capable of processing this kind of traffic. Moreover, supporting various types of users and a diversity of services should not degrade the overall network performance. It may be necessary to divide the network users into multiple groups so that the diversity of services and interconnectivity between sub-networks can be provided by the MAN media-access protocol in a uniform fashion. Most of the media-access protocols proposed for Satellite, Radio and Local Area Networks practically can support up to a couple of hundred users. On the other hand, MAN protocols are required to support several thousands of users scattered across a large geographical area. Therefore, a media-access protocol that can be considered for MANs should be capable of handling a large population of users while providing high channel throughput and low-access delay.

**3.3 Applicability to Metropolitan Area Network Topologies**

Generally speaking, the network topology, to some extent, plays a major role in determining viable media-access protocols. For example, star network topologies cannot effectively be used for contention protocols because of the complexity of the central node to resolve the collisions caused by a large number of stations. In the case of ring network topologies, the contention protocols would not work at all, however, token-passing and reservation protocols would be ideal. Concerning bus network topologies, almost all media-access protocols such as token-bus, contention, polling, and reservation protocols can be implemented effectively.

When selecting a media-access protocol for MANs we should take into consideration the network topology and its ramifications on the inner workings of the protocol. At the present time the majority of MANs are constructed using a tree-and-branch or bus topology based on coaxial cable. However, within the forseeable future more and more networks may be built with fiber optics cable. We emphasize that the media-access protocol being considered for MANs will be first applied to coaxial-based MANs because they are already in place and an abundance of bandwidth is available. Frequency-division multiplexing in conjunction with time-division multiplexing may be required to support a very large number of users in a MAN environment. This will allow partitioning the entire network into multiple sub-networks each utilizing different channels in the frequency spectrum. Assuming that the same media-access protocol is used in the entire network, the communications between sub-networks will be provided through bridges between networks using the same protocol, otherwise through gateways between networks using different protocols. Therefore, the MAN media-access protocol should be capable of operating in all network topologies including the tree-and-branch, bus, switched-star, and ring topologies.

### 3.4 Layered Architecture and Internetworking

A layered architecture conforming with the Open Systems Interconnection (OSI) Reference Model should be implemented so that transparent and media independent services can be provided in a Metropolitan Area Network, and so that internetworking between LANs and MANs can be simplified. One of the main functions of a MAN is to provide the transport mechanism between local area networks and global networks. For example, users attached to a MAN may need to access public networks such as TELENET or TYMNET, through gateways using the X.25 network protocol. A typical network user may need to access other similar networks using the same media-access protocol, (requiring bridges for internetworking) or dissimilar networks using different media-access protocols, (requiring gateways for internetworking). Therefore, the seven-layer OSI Reference Model and their standards implementations would make internetworking a much easier task. As is illustrated in Figure 3.1, if the networks including MANs and LANs use the same media-access protocol then the communication links between networks can readily be established via simple bridges, thus the end-to-end message delay is minimized and transparency between networks is provided. In other words, if the user Sc in LAN X wants to send a message to the user Sf in LAN Y, the message travels first through the bridge between LAN X and MAN X then the bridge between MAN X and MAN Y, and finally the bridge between MAN Y and LAN Y, thus the message is transferred to the destination user Sf. In the case of internetworking for dissimilar networks, as shown in Figure 3.2, the interconnection between networks is established through more complicated gateways. If the user Sd in LAN Z wants to access the database D in Public Data Network (PDN) X, then the message must travel through three gateways, thus increasing the end-to-end message delay. However, we can better understand the importance of standard protocols used by all networks in the example given above. If the user Sd uses OSI standards for high level protocols, specifically the OSI Class 4 Transport protocol, then the internetworking between dissimilar networks is simplified and made transparent. Here, we also see the advantages of conforming to standard protocols and layer architecture.

It is assumed that almost all network vendors will support OSI standards including transport, session, presentation and application layers in the near future. It

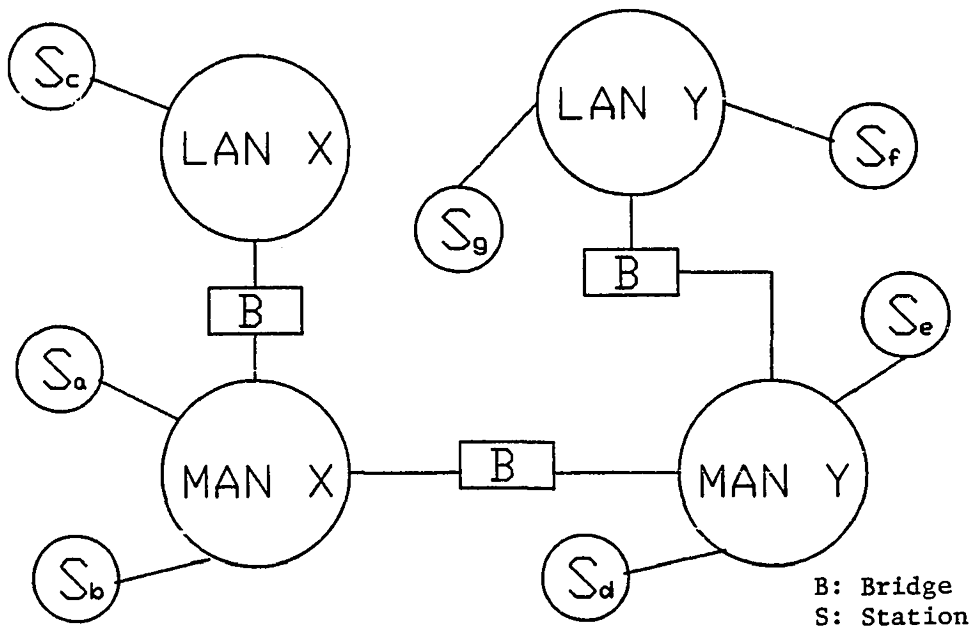



Figure 3.1. Internetworking of Similar Networks

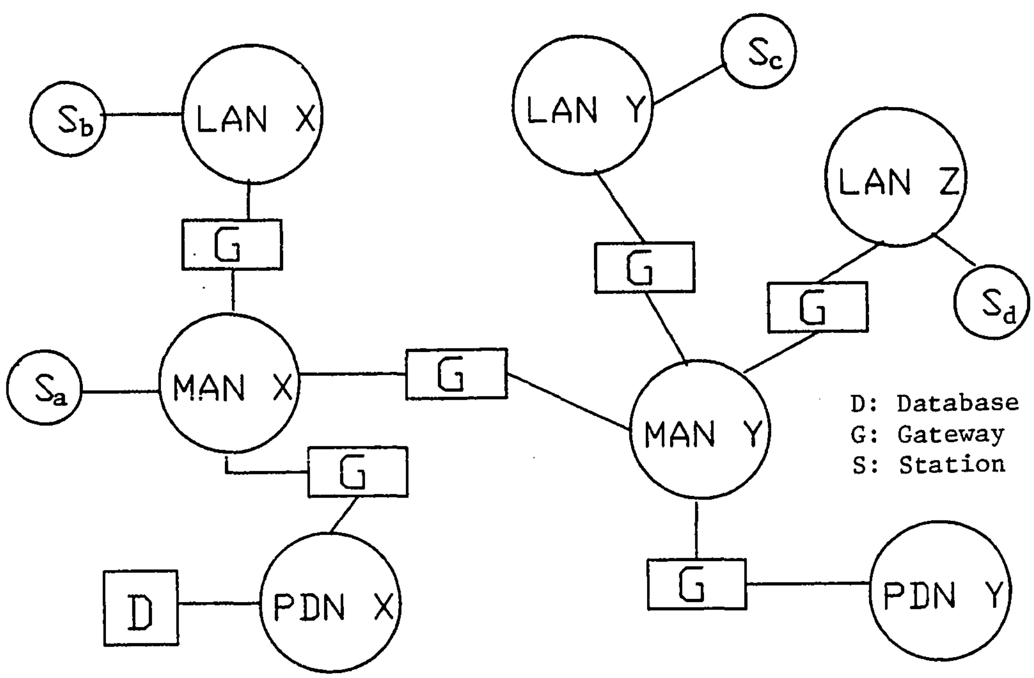



Figure 3.2. Internetworking of Dissimilar Networks

is also believed that MANs will provide the means for interconnecting various vendors' LANs and global networks. In order to accomplish this, the new media-access protocol should conform to the seven-layer OSI model and provide the mechanisms for supporting higher layer standard protocols such as the IEEE 802.2 Logical Link Control (LLC) Protocol, the ISO class 4 transport layer protocol, the ISO session layer protocol, etc. Since MANs are literally dispersed across a large geographical area, the issues related to network management should be considered carefully so that reliable and robust network operation can be maintained under all circumstances encountered in MANs. Regarding network management architecture, the layer and system level network management approach proposed by IEEE 802.1 Committee is taken as an appropriate basis. While defining the media-access protocol requirements for MANs we shall also try to conform to the standards being developed by IEEE 802, CCITT, and OSI Standards Committees.

### 3.5 Robust and Reliable Operation

Robust and reliable operation of a communication network is of prime importance to network users. In the evaluation of the media-access protocols, reliability and the ability to operate in spite of station failures should be studied closely. This can also be described as the "robustness" of the protocol, that is, its ability to survive errors, channel noise, and misinformation. The robustness of a communication network can be enhanced by enforcing the common rules pertaining to the media-access protocol despite channel errors. This implies that the network protocols should be capable of providing error-free and transparent message transmission even if some of the data packets are lost; this is accomplished by packet retransmissions coupled with packet acknowledgments. In the case of the CSMA/CD protocol, packets can be lost due to two reasons: 1) collisions due to simultaneous packet transmissions, and 2) errors caused by channel noise. The CSMA/CD protocol inherently assumes the fact that there exist packet losses and, therefore, it implements a retransmission backoff algorithm, and in conjunction with the error protection mechanism it provides error-free message transmission. Thus, the CSMA/CD protocol can be assumed to have the property of robustness. In the case of Token-Passing protocols, the robustness is enhanced by simply providing rugged token recovering techniques in conjunction with a vigorous error protection mechanism.

In order to provide reliable network operation, first of all, the media-access protocol should be extremely robust, thus network failure due to single-point failures cannot be tolerated. By virtue of this requirement, reliable and robust network operation is of prime importance. Therefore, the media-access protocol in question should be capable of providing robust and reliable message transmission under all network conditions. In order to accomplish this, rugged error protection mechanisms, proper collision handling techniques, and procedures are required for reliable network operation. This is particularly important in MANs because they may be constructed as public networks.

### 3.6 Low Bit Error Rate

Since MANs span a wide geographical area and may have to operate under extreme environmental conditions such as sub-zero and high temperatures, lightning, arcing, strong electromagnetic field, there exists the possibility of erroneous packet transmissions due to noise sources. These undesirable sources include white gaussian noise, ingress noise, corona, common-mode distortion, and impulse noise [SIRA 82b]. Contrary to conventional telephone switching systems, Metropolitan Area Networks offer high-speed transmission capacity with wide bandwidth. The availability of wide bandwidth permits radio-frequency based modulation schemes that provide transmission with very low bit error rates in the vicinity of one bit error in one billion bits. We can assume that MANs can offer very reliable message transmission facilities with very low bit error rates (BER). One of the most common techniques to decrease the bit error rate is to use wider bandwidth in the frequency spectrum. In particular, certain modulation schemes such as the phase shift keying (PSK) modulation technique will allow relatively low bit error rates. With the abundance of bandwidth available in a MAN, we can obtain very low bit error rates by increasing the carrier bandwidth.

As previously discussed data communications are not tolerant of bit error rates whereas packetized voice and digitized video communications tolerate a certain degree of packet loss. The media-access protocol being considered for MANs should be capable of providing data/voice/video services on the ubiquitous channel in a uniform manner since low bit error rates are highly desirable. Since the minimum acceptable bit error rate for data communications is in the range of one in a billion, high-performance modulation techniques with wide bandwidth allocation and rugged error detection and/or correction mechanisms may be required to meet the BER requirements for MANs.

### 3.7 Stable Operation

The stability of a media-access protocol or its ability to operate despite varying traffic demands and short-term overloading of the communication channel is of concern to network designers and users. Some of the contention-based protocols have exhibited bi-stable behavior, that is, the protocol initially operates at a point characterized by high-throughput and low average message delay; however, statistical fluctuations in the message traffic load eventually would force the protocol to operate at a point where the channel throughput is low and average access delay is high. In a given interval, assumed to be long enough, the traffic load conditions will again push the network into its initial operating point [LAM 75a, KLEI 75a].

Generally speaking, the stability of a communication network is highly dependent upon the throughput and access delay characteristics. The tradeoffs between stability, throughput and access delay should be determined in favor of the criteria selected for the communication network. Examples include tradeoffs such as high stability over low throughput and long access delay, and higher throughput over lower stability and longer access delay. Since the number of users to be supported in a MAN environment is high, the network stability issue gains importance. Unstable network operation may cause drastic degradation in the network performance and recovering from unstable states may take unacceptable periods of time because of the nondeterministic nature of certain media-access protocols. Considering the services to be offered in a MAN, these long access delays, low throughput and unstable channel conditions are not acceptable. It is, therefore, the media-access protocol in question is required to provide stable network operation under all circumstances. It must have the means to cope with unstable channel conditions or to render an unstable channel stable immediately. In order to guarantee a stable channel operation, we may have to sacrifice a certain portion of the channel capacity and/or have to tolerate longer access delays [LAM 74].

### 3.8 High Throughput

The throughput of a distributed multi-access protocol has traditionally been characterized by the maximum number of messages that it can successfully deliver per unit time and its access delay and throughput tradeoffs [KURO 84]. In high-speed communication systems, the portion of the communication channel that is used for arbitrating the channel access should be minimized so that high throughput levels can be obtained. Actually, the throughput of a shared channel is mainly determined by the efficiency of the control mechanism. The throughput of a shared channel can also be defined as a number representing the probability of successful packet transmissions in a finite period of time. From the above discussion we can state that high throughput can only be obtained by reducing the number of erroneous packets due to collisions or a noisy channel and the overhead of the media-access protocol. If high throughput is of prime concern then longer delays must be tolerated in exchange for higher performance. In other words, maximizing channel throughput will cause proportional increases in access delay. Specifically, in time-constrained applications the access delay and its variance should be optimized while the throughput is maintained at high levels. For those applications where the access delay is not so critical the throughput can be maximized by simply tolerating longer access delays. Considering the diversity and the volume of services to be offered in a MAN, high throughput levels are necessary even under fluctuating and varying load conditions.

### 3.9 Permanent Circuit Switching

In conventional telephone communication networks circuit switching is the access method used to interconnect two end-to-end points. When the connection is made on a permanent basis then it is called "Permanent Circuit Switching" in which a line or a time slot(s) is allocated physically. For the duration of the permanent circuit connection, access to the channel is guaranteed at all times.

In telephony, two possible modes of circuits are made available: i) permanent virtual circuits, and ii) switched virtual circuits. The only difference between these modes is in the interface function of session set-up and session termination [ROSN 82]. A permanent virtual circuit functionally replaces a permanent point-to-point circuit and guarantees connection, on demand, between a fixed pair of network stations. Since all data flow is between the same two end stations, there is no need for the station to tell the network the destination of each message. This type of interface is somewhat more efficient since the connection, essentially guaranteed, does not suffer a session set-up delay. The switched virtual circuit uses the maximum flexibility of the packet-switched network, which is, in effect, its main distinguishing feature.

In view of the services to be provided by a MAN, permanent circuit switching will be required to allow guaranteed access to the shared channel on a permanent basis. In the case of multiplexed voice communications, similar to T1 carrier requirements, a certain portion of the channel must be allocated for the duration of voice sessions. If a MAN is required to interconnect Private Branch Exchanges (PBXs) this type of permanent circuit switching would be ideal. In the case of packetized voice communications, permanent virtual circuit switching may be preferred. When a voice session is to established between two stations, a connection can be established with datagrams, utilizing a permanent time slot until the voice session is over. However, conventional data and digitized video communications do not require permanent circuits because there are no permanent sessions, rather sporadic transmissions of messages would be observed. From the above discussion it is obvious that permanent circuit switching would definitely be very useful for those applications in which a fraction of the channel capacity would be needed on a permanent basis.

### 3.10 Virtual Circuits and Datagrams

Typically, in circuit switching, a network resource is dedicated to a session or transaction on an exclusive basis. Physical resources in time, space, or the frequency spectrum are dedicated to the exclusive use of a single session for the duration of the session. The concept of packet switching is based on the ability of modern, high-speed digital computers to act on transmitted information so as to divide the sessions, calls, messages or transactions into pieces called "packets". Packet-switched networks can provide format, code and speed conversions between unlike terminal devices. Since most long messages or transactions will fit into a packet, long and short messages do not interfere with one other.

In packet-switched networks a channel may also be time-multiplexed among all stations that have been assigned time slots. This type of approach is known as a "Virtual Circuit", and the time during which a station holds the channel is known as a "Virtual Connection" [KURO 84]. Virtual circuits can best be defined by their properties. These properties include sequenced data transfer, data transparency, a full-duplex path, in-band and out-of-band signalling, flow control, error control, interface independence, and switchable form of operation. In sequenced data transfer, all data bits delivered to the destination station must be in the same order they were delivered to the network by the source station. This implies that the messages are to be reassembled sequentially. Data transparency means that all data bits in the user data fields must be accepted in any sequence of ones and zeroes. This implies the need for special handling of the data stream to protect against inadvertent packet delimiters. In full duplex operation, data has to be able to flow in both directions between stations simultaneously. Therefore, the initiation of a virtual connection and buffering for a message in one direction is also required in the opposite direction. In virtual circuits, the network station should be capable of reducing the input rate of information, namely flow control. One of the most important characteristics of virtual circuits is that the network operation is made independent of the physical and electrical properties of the station interface.

Virtual circuits, in general, are established when the connections between the network stations need not be made on a permanent basis. The channel capacity is allocated when only a packet is to be transmitted, otherwise time slots may be used by other stations. In essence, virtual ciruits are used by the protocols of the higher levels to provide end-to-end virtual connections that are served by datagrams. The salient characteristics of virtual circuits are:

1) The messages are long enough so that more than one packet transmission may be required,
2) Initial circuit set-up and formal circuit termination is required,
3) Absolutely reliable error-free transmission,
4) Sequenced message transfer,
5) Highly controlled operation.

In applications where the message to be transmitted cannot fit into a single packet, virtual circuits should provide a error-free, end-to-end transparent transport mechanism.

The datagrams are basically characterized with the following properties:

1) Every packet is self-contained including source and destination addresses,
2) Every packet can fully be identified because of its explicit addressing,
3) Transmission is reliable but may not be error-free,
4) Sequenced packet transmission is not required,
5) Network operation is totally uncontrolled.

In those applications where the user's data may fit into a single packet, the datagram type of network operation is preferred.

Given the diversity and requirements of the applications to be supported by a MAN, virtual circuits may be preferred for those applications where the connection is requested on a session basis, as in the case of accessing a remote database or host-to-terminal communications. Alternatively, the datagram type of service should meet the requirements of conventional data communications, electronic mail, facsimile, LAN interconnects, gateways, bulk data transfer, etc.

### 3.11 Multiple Channel Allocation

In a typical communication medium the bandwidth allocated for the shared communication channel has a limited capacity in terms of users that can be served and the services that can be supported. If the user population is large, as in the case of MANs, then a single communication channel may not have the capacity to support all network users. Basically, there are two reasons why multi-channel communication systems are required: 1) the present state-of-art technology coupled with the economy of scale under consideration limits the actual signalling rate to that of integrated circuits available today, for example the Ethernet chip set with a signalling rate of 10 MBPS, and 2) the need to partition the total number of users into smaller groups in order to minimize access delay and maximize channel throughput in addition to providing redundant and reliable network services. For example, in the case of broken communication links in a given channel due to noise, reliable network operation can be maintained by providing redundant communication channels operating at different frequencies.

A communication channel in a MAN is uniquely identified by its transmit and receive frequencies, and may be selected by the network station itself or determined by a centralized network controller. A MAN may consist of multiple physical channels operating over the same medium. A communication network to which a pair of uniquely identified transmit and receive frequencies are allocated may be referred to as a "Sub-Network". This implies that a MAN is comprised of multiple sub-networks each operating at different frequencies. The question that arises from the operation of multiple channels is related to how the interconnection between sub-networks can be established. It should be noted that a mechanism for interconnecting similar networks in a MAN environment must be developed to address this problem.

Given the requirement of multiple channel operation, the media-access protocol in question should posses a mechanism to assign the physical channels to network users so that multiple sub-network operation can be established. The physical channel number could as well be the part of station address or the sub-network designation. A MAN with multiple channel assignment can also be configured in the following forms: 1) different priority services may be offered in disparate channels, 2) coherent services may be offered in the same channel (i.e., data communications in one channel and packetized voice communications in another), and 3) a mix of services may be provided in the same channel.

### 3.12 Capacity Assignment on Demand

Any communication network that has a limited capacity can only serve a certain number of users as determined by the users' channel capacity need. The channel capacity, in general, is shared among two activities: 1) a control channel to arbitrate channel access among users, and 2) transmission channel. Fixed assignment and demand assignment are the most common channel allocation schemes used in communication networks. Fixed assignment techniques such as time-division multiple access (TDMA) or frequency-division multiple access (FDMA), provide a static allocation of capacity, which can grossly waste capacity if some of the assigned users have little or nothing to transmit over a period of time. Demand assignment techniques, such as polling or reservation methods using a slotted broadcast channel, are highly dynamic and serve a mix of high and low capacity users. However, for low demand homogenous users, the demand assignment techniques employ too much complexity for the benefits gained.

The major advantage of a demand assignment technique lies with the fact that the channel utilization can be improved greatly in comparison to fixed assignment. Considering the diversity of applications such as data, packetized voice, and digitized video communications and the varying volume requirements of MANs with a mix of changing traffic load, the channel capacity should be allocated dynamically on a demand basis.

### 3.13 Guaranteed Access with Fairness

Certain network services such as packetized voice communications require guaranteed access to the channel at periodic intervals. In order to allocate channel capacity on a guaranteed basis, reservation or token-passing protocols, namely deterministic media-access protocols in which the access delay is time-constrained, are commonly preferred. Guaranteed access to a communication channel can also be defined as each user having a certain portion of the channel at all times. This type of service implies that all network stations, with approximately equal probability, may access the shared channel under all circumstances, with a fraction of the channel capacity exclusively reserved for each network user.

Fairness in a communication network is described as the probability of each message having equal access for the next transmission. Specifically, in MANs where the network is accessed by a large population of users, the media-access protocol should allow fair arbitration among users with equal priority level so that every station is given equal opportunity to access the channel. Moreover, the media-access protocol in question should also provide equal access without any polarization toward a specific user group unless priority levels are different. In another example where a CSMA/CD access protocol is used for channel access resolution, all network stations should be capable of detecting collisions with equal probability, meaning that a strong station (station with higher receive level) and a weak station (station with weaker receive level) should both see a collision in the same manner. This is called the "Capture" effect, and even though it may help to improve overall channel utilization it is not desired if fairness among stations is going to be maintained across the network. It should be noted that if fairness in a communication network is not instituted then some stations may not be able to access the channel; this is particularly unacceptable for time-constrained applications.

Guaranteed access with fairness is a requirement in MANs because of the variety of services being offered to large numbers of users. Therefore, the media-access protocol intended for MANs should provide the means for guaranteed channel access with fairness.

### 3.14 Adaptivity to Load Fluctuations

The most significant properties of a media-access protocol for MANs are determined by the services supported simultaneously over the shared channel. It has been demonstrated that the nature of traffic load, in general, wavers between two extreme load conditions: 1) bursty, low-duty cycle, and 2) continuous, high-duty cycle [SIRA 82a]. In some instances the offered traffic load may show impulse characteristics rather than Poisson characteristics, namely very many users demanding channel access within a short span of the time domain. An abrupt surge in data or voice traffic may introduce impulse traffic whereas packetized voice traffic in a multiplexed form generally exhibits a continuous traffic pattern. In conventional data communications the offered traffic load appears to be bursty and intermittent. Even though traffic load conditions in MANs may vary within extreme boundaries, we can assume that the large-user population law can also be applied to MANs, that is, the average load in a network, with approximately equal probability, is the sum of the average loads presented by large users. Thus, as the number of network users increases the total average offered traffic load also increases proportionally. However, load fluctuations will occur because of the diversity and volume requirements of integrated services offered by MANs. Contention protocols generally exhibit high channel utilization with low average access delay under bursty traffic but exhibit poor performance and longer delay under heavy traffic. On the other hand, deterministic protocols such as polling, token-passing, etc., perform well in terms of high channel utilization and time-bounded access delay under heavy traffic conditions. Clearly, while one media-access protocol is performing well under bursty traffic conditions, it may have poor handling characteristic under heavy traffic conditions, or vice versa.

What is needed is a media-access protocol that adapts to varying traffic load conditions. However, adaptation to fluctuating traffic load may introduce more overhead, which in turn lowers channel utilization, thus introducing longer delays. The protocols using dynamic schemes generally adapt in some way to changing loadings, i.e., demand for channel use. Some media-access protocols can assess message loading levels so as to dynamically adjust to changing values in those levels by using information readily available on the channel at little or no overhead. Given the fact that the traffic load in a MAN varies within extreme boundaries, the media-access protocol suitable for MANs should have the capability to handle traffic load fluctuations without introducing overhead and excessive delays.

### 3.15 Priority Handling

The priority is defined as a function of the message to be transmitted and not of the device transmitting the message [ROM 81]. The requirements for a priority scheme are as follows:

1) Hierarchical independence of performance, that is, the performance of the scheme as seen by messages of a given priority class should be not affected by the load exercised on the channel by lower priority classes,
2) Fairness within each priority class, that is, several messages of the same priority class should be able to contend equally on the communication channel,
3) Robustness, that is, a priority scheme must be robust enough so that the network operation and performance is not affected by errors.

The diversity of services to be supported in a MAN emphasizes the need for a priority assignment scheme. Given the requirements of integrated services the media-access protocol for a MAN should have multiple priority levels corresponding to the network parameters such as average delay, guaranteed access, delay variance, etc. Prioritized services demand varying degrees of access time and channel capacity. Therefore, the priority levels should be defined such that specific service requirements can be met.

A priority-based system can be created by allocating greater transmitter power to those users with higher need to communicate, packets with greater signal strength have a higher probability of being received correctly even in the presence of interfering packets [ROSN 82]. This type of priority scheme is not appropriate for large area networks such as MANs supporting a large population of users.

A multi-level priority scheme should be available in MANs because of the diversity of applications with different priority requirements. Some media-access control parameters are determined by the priority levels associated with services offered on the communication network, and there exists a close relationship between priority level and media-access persistency. For example, real-time voice circuits require periodically scattered time slots as well as the highest priority on the shared channel. In contrast to voice traffic, terminal data traffic is tolerant of delay variance which leads to longer delays, therefore, a lower priority level is usually assigned to conventional data traffic. When two network users with equal priority level demand channel access in communication networks providing prioritized services, they access the channel with equal probability. Other users with lower priority, if demanding channel access at that time, are denied access to the channel until users with higher priority send their messages successfully. It has also been established that any priority rule will give rise to the same average packet delay [KLEI 76]. We point out that the overhead required to handle prioritized services should be minimal so that overall channel utilization is maximized.

In order tc accommodate a variety of services including data, packetized voice, and digitized/compressed video services in a MAN environment, a priority scheme with multiple and varying degree of priority levels must be incorporated into the media-access protocol.

### 3.16 Low and Predictable Delay

The media-access delay is described as the time elapsed from the transmission of a message to the reception of an acknowlegment or the reception of the message itself free of errors. There are several factors that may contribute to the total access delay, among those with the greatest impact are errors caused by noise sources in the channel and collisions due to multiple simultaneous transmissions. The offered traffic load into the channel directly impacts the average access delay, which in turn adversely affects channel throughput and stability. The average delay can be reduced at the expense of throughput and stability or vice versa.

Considering the service requirements of MANs low average delay and fast channel access without sacrificing channel throughput is desired under the light traffic load conditions generated by data communications devices. Alternatively, predictable channel access delay with bounded variance is required for time-constrained services such as real-time voice communications, etc. Since the average delay is highly dependent on the throughput of the channel, the tradeoffs between these two critical network parameters (access delay vs. throughput) should be balanced such that a low and predictable delay can be obtained for time-constrained services while maintaining high throughput and moderately acceptable delays for services that are not time-critical. Therefore, we emphasize that the media-access protocol to be considered for MANs should have the properties of contention protocols for lightly loaded systems, low delay with high channel utilization, and the properties of reservation and token-passing protocols for heavily loaded systems, predictable delay with good channel utilization.

### 3.17 Low Cost VLSI Implementation

A communication network entails a number of major design considerations for both user and supplier. The network supplier must balance flexibility with economy. The user must balance costs of service with network capabilities and reliability. From the user's perspective, the cost and average delay are the main factors to be minimized. Alternatively, from the supplier's point of view, the revenue and utilization are the two major elements of a network to be maximized. These considerations lead us to conclude that the low cost implementation of a network including associated network interface devices is crucial to providing technically and economically sound solutions to the problem of sharing network resources.

As integrated digital circuits reach their theoretical process limit, and with the increasing availability of computer aided design (CAD) systems for very large system integration (VLSI), a media-access protocol suitable for MANs could easily be implemented in a single VLSI chip. Thus, the low cost implementation of a MAN interface unit including the media-access controller is possible.

# CHAPTER IV

## ANALYSIS OF MEDIA-ACCESS PROTOCOLS FOR SATELLITE, RADIO AND LOCAL AREA NETWORKS AS APPLIED TO METROPOLITAN AREA NETWORKS

In a communication network where the transmission medium is shared among multiple network stations, there must exist a means for allocating channel capacity in an equitable manner. These channel resource-allocation schemes differ widely in their approach to multiple-access management. The most pervasive approach is to time-share the transmission medium among network stations. In the communication industry parlance, this technique is known as "time-division multiple-access" (TDMA) in which each network station is allowed access to the transmission medium at certain times under rules enforced across the network. This is analogous to controlling access to a highway via traffic lights. These rules constitute a media-access protocol by which all network stations time-share the transmission channel.

As the number of network stations and/or the message traffic load increases, the communication channel capacity will be totally utilized, and thereafter some performance degradation due to longer delays would be observed on the network. Under these circumstances, frequency-division multiplexing is utilized to extend network handling capacity by dividing the network stations into smaller sub-groups, and then assigning a unique frequency to each sub-group. Stations located in the same frequency band would have no difficulty in communicating. However, for communicating with a station located in another sub-group, bridging would be required between two sub-groups operating at different frequencies. The technique of operating a communication network using more than one frequency, each frequency comprising a sub-network, is called Frequency-Division Multiple Access (FDMA). This frequency-division multiplexing technique still permits time-division multiplexing in the same frequency band, i.e., TDMA is superimposed on the elementary FDMA. Another technique to share the frequency spectrum is Code-Division Multiple Access (CDMA). This method, also called spread-spectrum multiple access, is by far the least efficient in terms of bandwidth, and has been used mainly for military type of applications where antijam and security requirements are different [LAM 83].

The FDMA scheme is preferred when the traffic load between two communicating nodes is regular and continuous. In this technique, network stations communicate via carrier signals on which message signals are modulated as in radio and TV transmission. Hence, a multitude of network stations may share the frequency spectrum without interference. Generally, the frequency spectrum allocated for the communication system is sub-divided into channel-pairs; as such they may be assigned dynamically by a centralized control on a demand basis. One disadvantage of the frequency-division multiplexing technique is the poor utilization of the bandwidth because of guard-bands required between channels to prevent cross-interference. It has been shown that the fixed allocation of a scarce communication resource is extremely wasteful when the number of users to be served is large and the traffic they generate is bursty [TOBA 76]. Moreover, sharing a communication channel among a large number of users permits us to take advantage of the "Large Number Law" which states that the demand on the network at any instance, with very high probability, will be approximately equal to the sum of the average demands of that population [TOBA 76]. "It has also been established that rather than furnish individual low-speed channels to users, it is more efficient to provide the available communication bandwidth as a single high-speed channel to be shared by the contending users" [ROM 81].

It is obvious that the TDMA scheme would perform well under the conditions generated by a communications network with a large population of users introducing time-varying traffic load into the network. Therefore, we will concentrate on TDMA techniques developed for satellite, radio, and local area networks. We will analyze these media-access protocols as applied to Metropolitan Area Networks supporting integrated services. In this analysis, the following issues will be considered:

- Large signal propagation delay effect,
- Large population of users effect,
- Handling of integrated data/voice/video services,

- Protocol parameters' effect on traffic handling characteristics,
- Access-time delay vs. throughput vs. stability tradeoffs,
- Performance sensitivity to fluctuating traffic
- Overhead impact on channel throughput,
- Deterministic and non-deterministic nature,
- Fairness and guaranteed access,
- Topology applicability,
- Robustness,
- Network manageability and reliability.

First, we will delineate typical channel sharing characteristics based on the FDMA and TDMA techniques. Then, we will categorize all the media-access protocols intended for satellite, radio, and local area networks under four distinct basic techniques: 1) polling, 2) token passing, 3) contention, and 4) reservation. With the functional requirements of a media-access protocol intended for MANs under consideration, the media-access protocols falling under each one of the basic techniques shall be analyzed with respect to the comparison criteria given above. At the end of this chapter, we shall conclude with; 1) a comparative summary of a traffic model versus media-access protocol, and 2) the type of media-access protocol required to meet the functional requirements of a Metropolitan Area Network.

### 4.1 Channel Sharing Characteristics

A communication channel has a certain amount of bandwidth which network stations utilize to transmit and receive messages. This bandwidth can either be used for a single communication channel or be sub-divided into multiple channels. As it has been stated before, a single high-speed communication channel operates more efficiently than contiguous multiple sub-channels. The channel bandwidth is allocated in the following ways: 1) "Dedicated" channels, permanently assigned to pairs of communicating stations, 2) "Switched" channels, assigned to network station pairs on a demand basis via a centralized control, and 3) "Shared" channels, alloted by TDMA schemes.

In the case of a "Dedicated" channel, the total bandwidth of the channel is allocated to a pair of network stations, thus no arbitration mechanism is required to arbitrate access to the channel. If the traffic load between two communicating channels is continuous, then this type of allocation serves best. If the channel traffic is sporadic, then a certain portion of the bandwidth is wasted. This type of static channel assignment does not carry any overhead and no arbitration control is required, therefore, the channel throughput can be maintained at its maximum possible level. However, due to static allocation of channel capacity, some portion of the bandwidth can be wasted if some of the assigned users have little or nothing to transmit over a period of time [ROSN 82].

In communication networks where centralized control is available, the channel bandwidth can dynamically be switched from one pair of network stations to another with the use of a centrally located controller. As in the case of "Dedicated" channel assignment, no channel arbitration mechanism would be required for the "Switched" channel allocation. In order to allow channel switching, a small portion of the switched-channel capacity would be used as channel-control overhead. If the traffic load between two communicating network stations is sporadic rather than continuous, but a large amount of information is exchanged, then switched-channel allocation is ideal. Since the channel capacity is not permanently assigned, the channel throughput can be maintained at fairly high levels even though there is some overhead due to dynamic nature of the channel allocation scheme. If the traffic load is bursty, this channel allocation scheme cannot be used efficiently if the channel throughput is to be maintained at reasonable levels.

When the communication channel is shared among network users based on some sort of time-division multiple access scheme such as polling, contention or reservation, a certain amount of the channel bandwidth is wasted due to the overhead associated with the media-access protocol. The channel throughput is very much dependent upon the amount of the overhead consisting of control bits of the media-access scheme, mutilated packets because of collisions, addressing, error protection and packetization. Regardless of what the traffic load is, this shared channel allocation is the soundest technique available for sharing network resources among large numbers of users, thus reducing the average cost of communications. The media-access protocol needs to be enforced across the entire network, therefore, all network stations should have the intelligence to be able to conform to the rules. This makes the network stations more expensive and more sophisticated. Table 4.1 summarizes various characteristics of channel allocation techniques and provides comparative guidelines for communication network designers.

### 4.2 Polling-Based Media-Access Protocols

Polling is the most straightforward technique to allocate channel capacity to geographically dispersed users that are connected to a central facility via a common communication medium. This media-access technique is simple, stable, predictable, and easy to implement. In a polling technique, typically, the central control processor has ultimate authority and juristiction over the communication network. What distinguishes a polling protocol from a token passing protocol is the need for centralized control. The central processor transmits the addresses of each of the stations in sequence to inquire whether the station has any message to send. If the station has data to send then it responds to the poll by sending its message, otherwise, it may send a "no data" message back to the central control processor or it just may not respond at all; in turn the central control processor times out and polls the next

Table 4.1. Comparison of Channel Allocation Techniques

| Channel Allocation | Dedicated | Switched | Shared |
|---|---|---|---|
| Allocation Technique | Static | Dynamic | Dynamic |
| Overhead | None | Low | High |
| Centralized Control | No | Yes | Yes/No |
| Media Access Control | No | No | Yes |
| Throughput Under Continuous Traffic | Excellent | Good | Good |
| Throughput Under Bursty Traffic | Poor | Fair | Excellent |

station. This process continues in a cyclical fashion [MART 72]. This type of polling technique is called "Roll-Call Polling", and it is the simplest form of sharing a communication channel. Another form of centrally controlled polling protocol is "Hub Polling". This technique is closely related to "Roll Call" polling technique except the central control processor begins a polling cycle by broadcasting the address of the most distant station, thereby granting to this station exclusive access to the channel. After this station has transmitted its message, if any, it transmits an "End of Message" symbol which acts to grant access to the next most distant station. If the station knows the next distant station, it transmits the address of that station instead of "End of Message" packet. Upon receiving this symbol of its station address, the next most distant station repeats the process, passing an access symbol to the third most distant station when all its messages have been transmitted. This process continues until all stations have been given an opportunity to transmit their message whereupon the central control processor initiates a new cycle.

The main difference between two polling techniques is the time required to grant access to an individual station. In "Roll-Call" polling, the time required to transmit a message and receive a reply is typically much longer than the time required to transmit a symbol (or next station address) from one station to another. For example, under light traffic loading conditions the "Roll-Call" polling cycle time may, under the worst conditions, be twice that of "Hub-Call" polling. However, "Hub-Call" polling requires the medium to be highly reliable so that all stations can successfully receive transmissions from each other [HAYE 81].

Another form of polling technique that ameliorates the deleterious effect of overhead on the performance of "Roll-Call" and "Hub" polling schemes is "Adaptive" polling. The essence of this technique is to poll stations in groups rather than one at a time. The central control processor broadcasts simultaneously to all stations that have the same corresponding group address. If a member of a group of stations being probed has a message to transmit, it responds instantly. Upon receiving a positive response to a probe, the central processor splits the group into two sub-groups and probes each sub-group in turn. This process continues until active stations are isolated whereupon messages are transmitted. Once the active stations are identified the normal "Roll-Call" polling cycle is commenced by the central control processor. Furthermore, the processor periodically probes non-active stations to see whether any of them has a message to send or if any stations want to log on for information exchange.

In the "Adaptive" polling method the channel capacity is allocated on a demand basis and the media-access protocol adapts to the traffic load fluctuations with a predictable delay. The basic difference between "Adaptive" polling and other polling schemes is the improvement in the channel utilization due to less overhead in the station probing process. The "Adaptive" polling technique allocates a large portion of the shared channel to active stations, assigning lesser resources to non-active stations. The channel throughput is enhanced considerably in comparison to other polling schemes [HAYE 81].

After delineating polling-based media-access protocols, we will now analyze these protocols with respect to the criteria established for MANs. It should be noted that a centralized control is always required for polling protocols. Therefore, we will assume that this control is performed at a location (from now on it will be referred to as "Headend") where intelligent devices can be placed.

The signal propagation delay in a MAN environment is inherently large and its deleterious effect on network performance should be understood. This delay may range up to a couple of hundred microseconds for those networks spanning a large geographical area. Since polling protocols require some sort of centralized control, signals have to travel from the "Headend" to all network stations and back. This situation implies that the total end-to-end propagation delay is twice that of one-way signal propagation delay. This is evident with coaxial-based MANs because the signals emanating from the network stations flow in the reverse direction to reach the "Headend" and then are retransmitted back to the network stations in the forward direction. In conventional polling techniques, the "Headend" controller has to poll each network station by sending a poll command and then expect a data message or "no-data" message be transmitted back. The "Headend" controller stays idle while the signals flow through the network in both directions, the poll messages are being processed, and responses to polls are being generated. During this idle time, which equals the end-to-end propagation delay plus station and "Headend" processing time, the medium may not be accessed by any other stations. This means that the fraction of the channel capacity that is not used for data communications increases as the signal propagation delay gets larger. As is shown in Figure 4.1, as the ratio of propagation delay to packet transmission time, denoted as "a", increases, the average access delay for a desired throughput level increases drastically [HAYE 81, TOBA 76]. Since the end-to-end signal propagation delay in a MAN is large, the channel throughput would be influenced greatly by the value of "a".

In some applications of polling protocols the "Headend" controller generates a stream of polling messages, rather than probe-and-wait-for-response type of processing, and receives messages in the same sequence of polling messages. This technique is also called "Pipelined" polling and it alleviates the end-to-end-propagation delay effect to some extent. Furthermore, as a variation, the stations' distance to the "Headend" is measured and then the stations are sequenced according to their distance from the "Headend". Doing so minimizes the end-to-end propagation delay effect on network performance. This technique logistically is difficult to implement, and requires a fixed packet size. Moreover, "Pipelined" polling requires the "Headend" controller be very powerful in terms of computing power and high-speed data handling. Nevertheless, in networks where the end-to-end signal propagation delay is large, as is the case in MANs, polling protocols suffer from poor channel utilization and longer access delays.

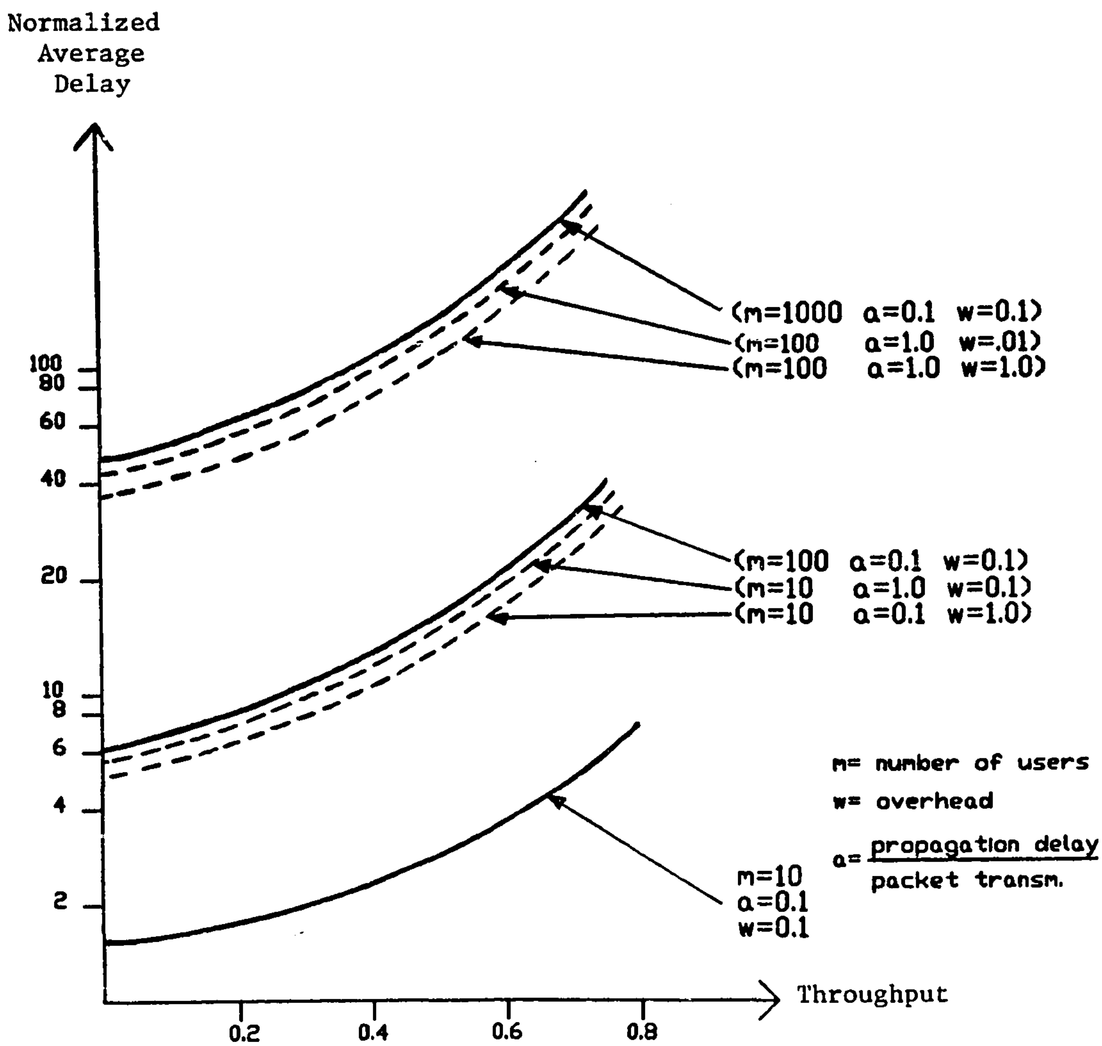



Figure 4.1. Polling Protocols: Delay vs. Load vs. Throughput

Since a MAN is considered to be supporting a large population of users we should also understand the consequences of this large load. Generally speaking, the number of stations to be served by a communication network influences the average access delay, the throughput, and the stability of the media-access protocol. In polling-based media-acess protocols the "Headend" controller polls the network stations in a predetermined sequence. Hence, as the number of network stations increases, the length of this polling cycle would also increase. In turn the average access delay increases directly proportional to the length of the polling cycle [KUO 81]. This situation is shown in Figure 4.1, if the average number of polled stations is doubled, then the average access delay would also double. However, this increase in access delay is linear and is easily calculated. From the curves in Figure 4.1, it is concluded that as the channel throughput approaches one, ideal M/D/1 system, (that is, channel traffic inputted with exponential interarrival-time probability and processed by one server with a deterministic servicing-time probability), the access delay increases rapidly [TANE 81]. Typically, in polling-based protocols, when a station has nothing to send its time slot goes by unused even though other stations could potentially utilize this unused time slot. This problem could, in particular, be acute when the number of stations served by the network is large. This problem can be overcome by using an "Adaptive" polling protocol in which empty slots may be allocated to active stations. Nevertheless, the "Headend" controller, in conformance to the "Adaptive" polling method, is required to poll all active and non-active stations periodically so that active stations can be identified.

By virtue of these observations we may conclude that a very close relationship can be drawn between the average access delay, channel throughput, and the number of stations being served on the shared channel. If longer access delays can be tolerated then more stations can be served on the same communication channel. As a result, the number of stations that can be served by a communication network using polling-based media-access protocols is limited by the access delay that can be tolerated. It has been observed that the applicability of polling protocols will be strongly dependent on the number of stations attached to the network [KURO 84]. Therefore, the polling-based protocols their descendants may not all be applicable to MANs because of the effect on the throughput and access delay induced by a large number of users.

In polling-based media-access protocols every network station is always interrogated whether it has any message to send or not. This means that if there are several thousands of network users and only a small fraction of them have messages to send, then a large fraction of the channel capacity would be wasted for polling those stations that have no message to transmit. In essence, in conventional polling protocols each station is assigned a time slot regardless of having a message to transmit. The traffic load, in conventional data communications, has the property of burstiness. If a network station has a message to send then it has to wait for its time slot to arrive. This situation may cause long delays for stations whose traffic characteristics are bursty in nature. With connection-oriented services in which a session between two communicating stations is first established before any message exchanges, "Adaptive" polling would outperform other polling-based media-access protocols because the protocol overhead is considerably reduced by assigning fixed time slots for active users. Even though the throughput of the "Adaptive" polling technique may be acceptable for certain applications, it may still be ineffective under time-varying bursty traffic conditions because of longer access delays.

Since polling is an explicit version of a reservation scheme it provides an excellent channel allocation environment for time-constrained applications, particularly for real-time voice traffic. As in circuit switching in telephony, assigned slots may comprise a permanent circuit established between two communicating stations. The periodic nature of voice traffic requires a guaranteed access to the channel at given intervals. In this case, slots may be reserved on a periodic basis, and may be released when the session is over. Because of its guaranteed access capability polling-based media-access protocols are preferred for those applications where the offered traffic load is either continuous or periodic. On the other hand, the centralized control required for polling-based protocols introduces processing delay in addition to end-to-end signal propagation delay and other system related delays. The total transmission delay may not be tolerated for real-time voice communications.

Video messages contain large amounts of data and come in bursts, thus they require a large fraction of the communication channel for short periods. In this case, it may be necessary that the total available channel capacity be allocated to the communicating station pair for the duration of the digitized and compressed video message. The polling-based media-access protocols, in general, permit only a single user station to transmit in a given time period, therefore, they may easily allocate any fraction of the channel capacity to any network station for a predetermined time period, thus allowing large amounts of data to be exchanged between stations. It is evident that the polling-based media-access protocols are ideal for the transfer of bulk data.

In light of their traffic handling characteristics, the polling-based media-access protocols are well suited for heavily loaded systems in which large amounts of data are exchanged continuously. In contrast, they perform poorly in terms of channel throughput and access delay for lightly-loaded systems in which the traffic load is fairly bursty and sporadic. It should also be noted that for lightly-loaded systems using an "Adaptive" polling technique, considerable decrease in polling cycle time can be achieved without penalty for heavily loaded systems [HAYE 78]. In terms of the applicability of polling-based protocols to MANs, we may conclude that they are not appropriate for integrated services because of their non-adaptivity to load fluctuations and their poor performance under light traffic conditions.

The strength of polling-based protocols is their obvious stability. This means that under all traffic load conditions including time-varying traffic demands, short-term overloading, etc., polling-based media-access protocols would not exhibit an unstable condition, thus no channel throughput degradation and no effect on the average access delay would be observed. In general, the access-time delay, throughput and channel stability are very closely tied to each other. For example, if lower channel throughput and longer access delays could be tolerated, then the channel stability might be improved. In another example, the channel throughput and access delay properties of a media-access protocol can be improved at the expense of channel stability.

Because of their deterministic and cyclical nature the maximum channel access-time of the polling-based media-access protocols can always be measured and a guaranteed access to the channel at given intervals can be insured. Since the polling process is centrally controlled, preventing collisions between packets, the channel backlog and other types of unstable channel conditions do not exist. The channel stability is very rigid but the weakness of the polling-based protocols lies with the large channel access delay and low throughput under light traffic load conditions. The expected delay is relatively long as compared to contention-based protocols, especially for lightly loaded systems. In contrast, when the system loading is high the channel throughput is maximized while the expected access delay is minimized. The channel stability of the polling-based media-access protocols is proven to be rugged under all traffic conditions, thus channel throughput and access delay tradeoffs should be studied rather than stability.

The channel throughput of the polling-based protocols is mainly determined by the amount of traffic load offered to the network. The higher the traffic load the better the throughput. In lightly-loaded systems, the channel throughput is at its minimum level, whereas it may reach its maximum level in heavily-loaded systems. Because of their stable nature, the polling-based protocols handle traffic fluctuations fairly well without any performance degradation. When the traffic load is bursty and sporadic the channel throughput is not affected but longer packet delays would be observed. Under continuous traffic conditions the throughput is maximized and it may reach the ideal M/D/1 level. When the traffic load has impulse characteristics then the throughput may peak with the impulse traffic load and may drop to its average level at other times. The access-time delay is adversely affected by the number of network stations attached to the network. As the number of active stations increases the access-time delay would also lengthen proportionally. In view of these discussions we may suggest that the access-time delay, channel throughput, and channel stability are not interrelated in the case of polling-based protocols, therefore, they should be considered in the scope of their relevant factors.

The overhead, in general terms, is the information (in number of bits) required to arbitrate the channel access among network stations. The channel throughput is defined as the ratio of the actual data transmitted to the actual data plus the overhead. The overhead consists of the protocol control information, synchronization information, collisions, and other protocol related operational information. In essence, the channel throughput is determined by the amount of the overhead required to operate a communication network. In polling-based media-access protocols, the overhead consists only of the information transmitted over the channel to poll network stations. If the number of stations attached to the network is large, as is the case in MANs, then an appreciable fraction of the channel capacity would be allocated for the polling cycle, thus increasing the overhead and in turn decreasing the channel throughput. This situation results in a bounded, but excessive channel access-time delay which is determined by the length of the polling cycle. The overhead in polling-based protocols is determined by the number of stations attached to the network. If the total traffic load over the channel is kept constant while the number of network stations is doubled, the overhead would also double, and in turn this will result in lower channel throughput.

Furthermore, if the overhead is incurred on per station basis then the polling cycle time is doubled with no increase in traffic [HAYE 81]. A strong dependence on overhead is also shown in Figure 4.1, increasing the number of stations attached to the network influences the channel throughput adversely. The polling-based protocols were shown to be inferior to contention-based protocols due to the large overhead caused by the need for control and slot synchronization [TOBA 76]. In order to minimize the overhead, the control information should be kept minimal by determining active stations using an "Adaptive" polling technique in conjunction with a station probing scheme. Since the number of stations supported in a MAN is relatively large, the overhead due to the polling cycle is expected to be high, and this results in lower channel throughput.

The deterministic or nondeterministic nature of a media-access protocol is determined by the probabilistic or stochastic factors that may play a major role in the operation of the protocol. Contention-based protocols, in general, are of a nondeterministic nature because of the finite probability of virtually uncontrollable collisions on the channel and probabilistic retransmission algorithms. In other words, the packet transmission time is calculated based on some probabilistic station and network parameters. In contrast, the polling protocols are deterministic in nature, therefore, the expected access delay and its variance as well as channel throughput can be estimated.

Time-constrained applications such as real-time voice communications require a deterministic media-access protocol. As a matter of fact, the stability of a media-access protocol is highly dependent on its deterministic or nondeterministic nature. If the media-access protocol is based on some probabilistic calculations of network parameters, then the channel stability would be determined by the degree of these probabilistic network parameters. In some communication networks the channel stability may be improved at the expense of channel throughput and access delay. On the other hand, since the polling-based protocols require some sort of centralized control it is likely that the protocol itself would exhibit deterministic behavior and, therefore, very solid stable network operation could be made possible. Since the channel stability in a MAN is highly critical and crucial to proper network operation in the face of a large population of users, the deterministic nature of polling-based protocols would overwhelmingly be preferred for time-constrained applications such as real-time voice communications. In data communications channel access delay may be traded for higher channel throughput and better channel stability. The use of polling-based protocols in data communications would be overkill since channel stability is not as critical as it is for time-constrained applications. In summary, the polling-based protocols are fairly deterministic in nature, and, therefore, they should be considered for those applications where channel stability is important and critical to the network operation. Specifically, for those networks supporting a large number of stations, as the case in MANs, the channel behavior should be under control, otherwise the channel may render itself unstable.

Fairness in a communication network is defined as the allocation of channel capacity on an equal access basis. The polling-based media-access protocols assign slots for each network station, thus no other station may transmit messages in these pre-assigned slots. The number of slots assigned to a network station is predetermined by the centrally located network controller. Therefore, in polling-based protocols all network stations have equal access right to the shared channel. Capture effect, collision detectability, and other issues related to fairness are of no concern in polling-based protocols. When a network station is assigned a time slot to transmit its messages, explicit signalling is present either through an actual message or a "no-data" message response. This type of slot allocation scheme guarantees channel access for only one station at a time. In time-constrained applications guaranteed channel access is critical due to the requirement of non-variant access delay. Generally speaking, polling protocols are superior to contention protocols in terms of fairness and guaranteed access.

All network topologies lend themselves to polling-based protocols. The centralized network controller typically resides at the headend location but it may also be placed at any location in the network. On the other hand, in order to minimize the signal propagation delay effect the network controller should be positioned virtually in the middle of the transmission path. Therefore, the network controller is located at the headend in coaxial-based MANs. From the standpoint of network topology polling-based protocols can easily be implemented for all MANs topologies including tree-and-branch, bus, and star.

Polling-based protocols provide robust communications between network stations since bit error rates would have very little effect on the network operation. This is evident in the case of a mutilated polling packet; the station whose packet is lost will get another chance in the next polling cycle. On the other hand, the network operation is totally dependent on the proper functionality of the centralized network controller. Since the network operation is susceptible to a single point failure of the network controller, a non-stop redundant network controller becomes a necessity for reliable network operation. In contrast to network controller operation, a single network station failure would not harm the communication network, polling-based protocols are capable of broadcasting messages to all network stations simultaneously. This characteristic would permit a reliable network management function. This means that the network controller should be able to send ubiquitous commands to all stations at once and inform them about network conditions instantly. Since a poll and its response constitute a verification and authentication of a network station, the status of each network station can effectively be monitored. This characteristic of polling-based protocols is an important feature in terms of network manageability and reliability, specifically for MANs.

### 4.3 Token-Passing Based Media-Access Protocols

In token-passing access schemes, control is passed via a token from one station to the next; the station with the token is the only one allowed to transmit. This scheme is closely related to polling techniques except that no centralized control is required. This implies that each station must have the intelligence to determine when it has control of the shared channel. However, each station can be polled either explicitly from a central station or implicitly via distributed control. Collisions are rare events except during initialization of the system, loss of a token, generation of multiple tokens, and so forth.

In token-passing protocols, stations are logically arranged in a ring, this allows separation or dedication of transmission capacity to different types of services, by simply allocating a certain number of time slots per polling cycle to each station. The process begins with the first station transmitting its message. At the end of message transmission the station transmits the token (a special packet indicating channel access status) to the next station in the sequence. The second station in the logical ring holds the access right to the channel. If it has any message to transmit, it does so, otherwise it passes the token to the next station in the logical ring. As with polling, this process is continuous. Token-passing based protocols are more complicated than polling because they require stations to have more intelligence. This complexity also stems from the need to add a new station to the logical ring, to remove a station from the ring without stalling the token, and to recover if multiple tokens are created. The token-passing media-access protocols are used because of the higher transmission efficiency possible with variable-length message sizes [KINN 82]. Almost any desired efficiency can be obtained by increasing the amount of data per message frame (within the limits set across the network).

Depending on network topology token-passing can be used in many different ways. For example, in slotted-ring networks control access is controlled by circulating a fixed number of fixed-length message frames which are initially empty. A station that wants to transmit a message waits for an empty frame to pass by (one token per slot) in which it inserts its message. The destination station (or stations) copies the message as the frame passes by. When the frame returns to the sending station, the station marks the frame empty, releasing the slot by sending a token in the frame. Stations are not allowed to reserve the same frame immediately, thus preventing a station from hogging the network. The slotted-rings allow a convenient form of message acknowledgment. This is done by a receiving station copying the information from the frame and stamping an acknowledgment bit on the frame. A drawback of slotted-ring networks is low efficiency which stems from the limited size and number of message frames that can be simultaneously accommodated on the ring. For example, the token-ring network developed by Cambridge University (called the Cambridge Ring) operates at 10 MBPS, however, the maximum effective transmission rate between stations is limited to 1 MBPS [WILK 79]. In the case of token bus networks where a logical ring rather than a physical ring is established across the network, higher transmission efficiency is possible with variable-length message sizes, thus allowing higher throughput.

The end-to-end signal propagation delay (defined as the time from the starting of a transmission to the starting of reception between the extreme stations) plays an important role in determining the throughput and access delay of token-passing based media-access protocols. Since the token is passed from one station to another positioned in the logical ring, the time spent for token passing is of prime concern, specifically for those networks spanning a large geographical area. For example, in a typical MAN environment in which a virtual bus is established by frequency translating the upstream channels into downstream channels, the token has to travel from a transmitting station to the headend and back again. The end-to-end signal propagation delay is twice that of one-way delay. As the propagation delay increases the overhead, in terms of access delay caused by passing the token, also increases. This channel overhead will automatically reduce the network capacity, in turn limiting the number of stations that can be supported by the network. As with polling, token passing has a relatively high access delay for low levels of system usage, namely bursty traffic, because the token is constantly passed around the logical ring. If the number of stations is kept constant, the throughput of a token-passing based media-access protocol would largely depend on the end-to-end signal propagation delay. The effect of the propagation delay over the channel capacity has been studied vigorously for various types of token-passing schemes including Alternating Priorities (AP), Round Robin (RR), and Random Order (RO) protocols [KLEI 80]. In Figure 4.2, the capacity "c" is plotted against the ratio of propagation delay to transmission time "a" for various values of N, number of stations. As "a" increases, for a small number of users (N=10), the token-passing based media-access protocols would have a channel capacity which is higher than all contention protocols for values of "a" not larger than 0.038. While "a" is still small the channel capacity is fairly good, but as "a" gets larger, the throughput will drop below that of the Slotted-Aloha protocol. When the number of stations is large the channel capacity "c" quickly decays as "a" increases, e.g., if N=50 for values of "a" greater than 0.35, the channel capacity even drops below that of Slotted-Aloha [KLEI 80].

In EXPRESS-NET the media-access protocol is a variation of the Round-Robin scheme, and the end-to-end signal propagation delay effect is minimized because the time to switch from one active user to the next in the ring is on

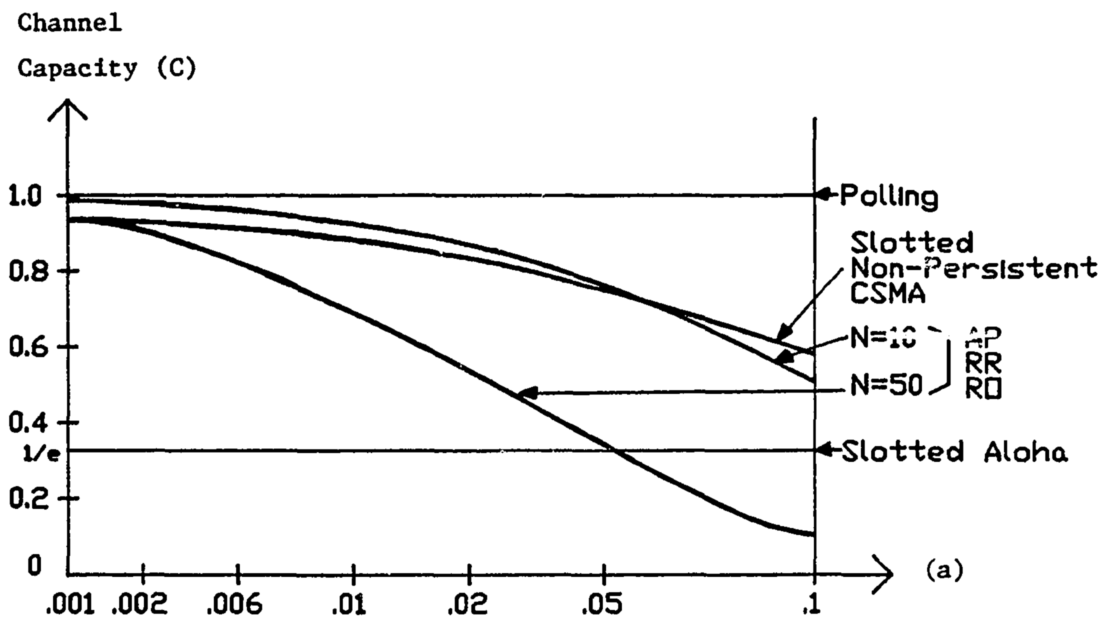



Figure 4.2. AP, RR, RO: Effect of Propagation Delay on Channel Capacity

the order of the carrier detection time [TOBA 82b]. Nonetheless, the topologies inherent to MANs will not permit stations to be physically ordered, and the end-to-end signal propagation delay will be large. Even if the stations' distance from the headend can be measured accurately and then organized such that the closest one be the first and the farthest one be the last station in the ring, the end-to-end signal propagation delay effect cannot be excluded in the calculation of the throughput of the EXPRESS-NET protocol. Moreover, the EXPRESS-NET performance measures are appropriate only with physical bus topologies, thus imposing a topological constrain. Similarly, the FASNET media-access protocol, which is based on the Slotted-Ring using a token-passing scheme, suffers from the same consequences as the EXPRESS-NET protocol [LIMB 82]. For example, as with the EXPRESS-NET, the FASNET protocol also requires a physical bus topology. If the FASNET media-access protocol is realized over the MAN topologies described in Chapter II, then the end-to-end signal propagation delay would degrade the network performance including the channel throughput and the access delay. The reason behind this deleterious effect is that the channel acquisition time in a token-passing network is mainly determined by the end-to-end signal propagation delay, and is not affected by the increase in the data rate [PARK 83]. In other words, as the data rate increases the channel efficiency will drop toward zero. It is evident from the above discussions that the throughput of the token-passing based media-access protocols is highly dependent on the end-to-end signal propagation delay. Therefore, in wide area networks including MANs where large signal propagation is inherent in the system, token-passing schemes would not be effective in terms of desired throughput levels unless the stations in the ring can physically be placed in sequence.

In determining the utilization of network transmission capacity and the traffic handling characteristics of such a system, we must not only consider the average access delay, but also how many stations can be supported by the network. Since the time spent passing the token in the ring is proportional to the number of stations, N, attached to the network (certain schemes pass the token only to active stations), the throughput of token-passing based media-access protocols is highly dependent on N. In Figure 4.3, the channel capacity "c" is plotted against the number of stations, N for various value of "a" and for AP, RR and RO protocols based on token-passing schemes [KLEI 80]. On the same figure the plotted curves show the channel capacity of polling, CSMA, and Slotted-Aloha protocols for comparison purposes. As the number of stations, N increases at fixed value of "a", the channel capacity of the token-passing protocol decays very quickly below the Slotted Non-Persistent CSMA protocol capacity. It is interesting to see that as "a" gets smaller more stations can be handled at fixed capacity. Since the overhead time per token-passing cycle is bounded by a linear function of the number of stations attached to the network, we can easily calculate the channel throughput for all token-passing protocols. This means that token-passing protocols guarantee a bounded transmission delay for each packet.

In Figure 4.3, the access delay and throughput of token-passing protocols is plotted against the number of active stations attached to the network. As is illustrated in Figure 4.4, as the number of active stations increases, the access delay also increases proportionally while the channel throughput performance improves. After a certain number of stations become active the throughput saturates at its maximum attainable level, but the access delay will continue to increase linearly to a point where the worst case loading is reached for the logical ring. Suppose that the fraction of time the transmission medium is used for

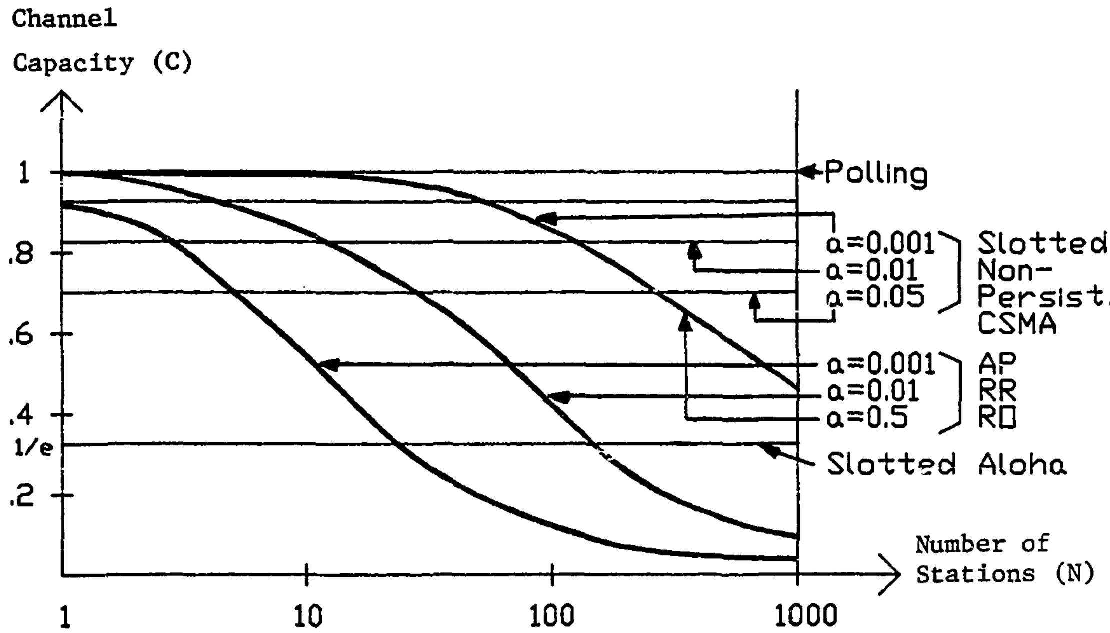



Figure 4.3. AP, RR, RO: Effect of Number of Stations on Channel Capacity

transmitting actual data is fixed. As the number of stations is increased steadily the amount of data transmitted per message and per station decreases toward zero, hence the mean waiting time or access delay experienced by any station will increase beyond any fixed threshold because more and more time will be spent in passing the token from one station to another rather than transmitting actual data.

Considering a MAN environment in which the number of attached stations is expected to be in the range of thousands, the token-passing based media-access protocols would not perform effectively due to the requirement of passing the token to all stations. This increases the overhead drastically and the time spent passing the token may exceed the time spent transmitting the actual data.

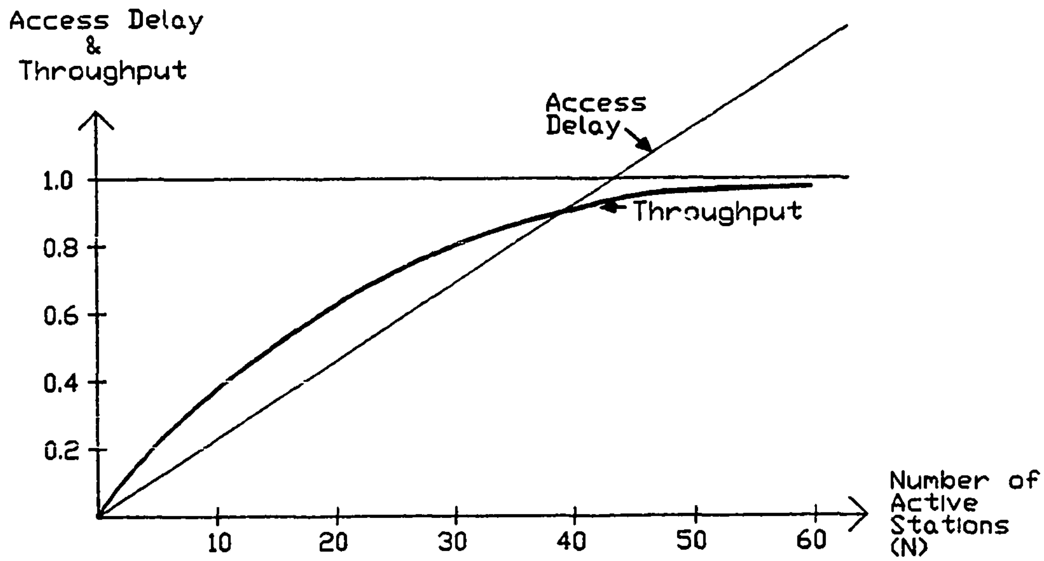



Figure 4.4. Access Delay vs. Throughput vs. Number of Active Stations

The most obvious advantage of token-passing based media-access protocols is the improved performance under heavy traffic loads with guaranteed access capability. We also note that these protocols are extremely effective when the number of attached stations is not too large; as the number of stations attached to the network increases, the large overhead leads to a performance degradation [KLEI 80]. Under heavy load, high throughput is achieved because the transmission medium will be busy transmitting actual messages rather than passing the token frequently from station to station. Under light loading conditions, a station will have to wait for the token to circulate through one half of the total number of stations, on the average, before it will be allowed to transmit [ARTH 82].

Therefore, a relatively long access delay will be observed for bursty traffic loads since the token is passed constantly around the ring. As might be expected, unlike ETHERNET (based on the CSMA/CD protocol), with token-passing protocols as the load increases the channel throughput will also increase gradually [LIMB 82]. The throughput of token-passing based media-access protocols, in general, improves as the length of a cycle increases; the cycle length is determined by the length of a packet, the number of active stations, and the number of packets that each station is allowed to transmit in a cycle.

The channel access arbitration in token-passing protocols is essentially unaffected by the medium length until the end-to-end signal propagation delay becomes greater than token duration. At that time the overhead associated with passing the token may well exceed the channel capacity utilized for transmitting the actual data. However, the throughput is greatly affected by the time taken at each station to execute the required token-passing protocol routines. We also point out that a guaranteed and bounded transmission delay can be obtained with these protocols. Therefore, token-passing schemes are very appropriate for the transmission of voice packets. Alternatively, since the mean message delay time is a linear function of the number of stations attached to the network, the mean waiting time for any one station is simply the time required for all other stations in the ring or round to transmit if each station has a message to transmit. However, the maximum message delay is bounded by the worst case loading conditions of the logical ring. If the utilization is fixed, as the number of stations increases each station will be less and less likely to have a message to transmit, but the token must still be passed through each station in the ring, hence the total message waiting time will be increased proportionally. As is shown in Figure 4.5, as the number of active stations increases the channel throughput is also improved [CARP 84]. The maximum throughput is achieved when all stations on the network become active, thus less and less time is spent passing the token. As it is illustrated in Figure 4.5, the channel throughput of the token-passing based media-access protocols is dependent on how many active stations exist on the network.

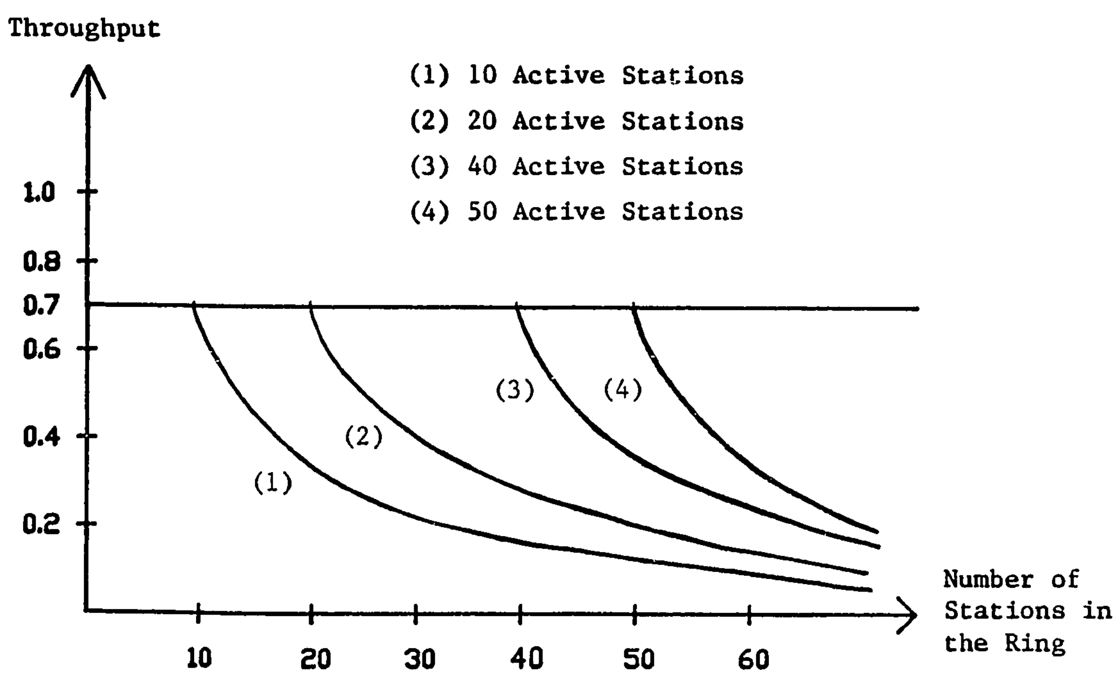



Figure 4.5. Throughput vs. Number of Stations in the Ring

Since the traffic load introduced into the network, in conventional data communications, is fairly bursty, the average access delay is relatively longer than that of contention protocols. Therefore, under light loading conditions, token-passing based media-access protocols perform very poorly, resulting in longer access delay and lower channel throughput. However, for those applications in which large data files need be transported, the token-passing schemes are ideal. In the case of real-time packetized voice communications which require a guaranteed and bounded access delay, the token-passing protocols perform very effectively. Similarly, these protocols also lend themselves very nicely to the transmission of digitized and compressed video information that is less bursty than data traffic.

Since the channel access is controlled via a token granting transmission over the medium, traffic fluctuations in the communication channel result in longer access delays, but do not affect channel throughput. However, if the logical ring consists of only active stations rather than all stations attached to the network then abrupt patterns of getting in and out of logical ring may cause more collisions. Even with this increased protocol overhead the effect on the channel utilization is minimal. Nonetheless, the traffic fluctuations over the communication channel will not impact the total system operating under token-passing based media-access protocols.

Since collisions are rare events except during initialization of the system, loss of a token, generation of multiple tokens, etc., the token-passing based media-access protocols exhibit stable operation. System stability is achieved at no expense in access delay and channel throughput. Even when collisions take place on the network under the conditions described previously, the token-passing protocols would adjust themselves so that the network would be rendered stable within a period during which the conditions causing collisions are alleviated. However, if multiple tokens are generated due to token losses caused by errors, then for the duration of the multiple token resolution process, the network is considered to be in an unstable state because of possible collisions. The length of this duration actually determines the robustness of the token-passing based media-access protocols. The shorter the duration the more deterministic the protocol. Because of their stable characteristics and guaranteed access capability token-passing based protocols are considered to be ideal for integrated bulk data and voice communications in which the traffic load is fairly continuous and periodic. Conversely, these protocols are not appropriate for those applications such as conventional data transfers, terminal attachments, electronic mail, etc., in which the traffic load is predominantly bursty.

Most broadcast media-access arbitration schemes use some fraction of channel capacity to control channel access among network stations. The token-passing techniques also incur some overhead, not due to collisions as in the contention protocols, but rather due to other factors such as tokens, end-to-end signal propagation delay, circuitry transients, processing, etc. In token-passing based media-access protocols, the overhead time per transmission is bounded by a linear function of the number of stations attached to the medium because the token (a control symbol) must be passed to each station at least once in a ring or round. The overhead incurred by collisions can be ignored because they occur rarely. A respectable fraction of the overhead is due to the requirement of passing the token to each station whether or not a station has a message to transmit. In order to mitigate the token-passing time a logical ring consisting of only active stations may be constructed. In this case, if a station becomes active, wanting a turn to transmit its message, then it should be given a turn in the ring; this process may cause some collisions introducing some additional overhead, but it is eventually better to establish an active logical ring rather than including all stations. We should also understand the implications of the overhead incurred by the station transceiver; the time taken to process a token is also critical. As such the longer the processing time the higher the overhead. In order to mitigate against the overhead caused by transceivers, high-speed circuits should be employed. Another major contributing factor to the overhead is the end-to-end signal propagation delay. As the network coverage area gets larger, the increasing end-to-end signal propagation delay (larger overhead) will adversely impact the channel throughput. In a typical MAN environment, the overhead incurred by various factors, mainly by large end-to-end signal propagation delay, may amount to a respectable fraction of the total channel capacity. In light of the above arguments, we can conclude that due to the high overhead incurred by various network parameters such as high end-to-end signal propagation delay for a larger area of coverage the token-passing based media-access protocols may not deliver the desired high channel throughput needed to support integrated services over MANs.

Since the arbitration mechanism for the token-passing based media-access protocols is based upon a deterministic process in which the channel access is controlled via a token, the probabilistic factors, unlike contention protocols, play no role in the system operation. Because of the deterministic nature of the token-passing based media-access protocols, guaranteed access to the channel can be provided for all network stations regardless of traffic loading conditions. However, some experts argue that the token-passing protocols are not deterministic because of collisions caused by possible token losses, multiple token resolution process, etc. It should also be noted that when the number of stations attached to the network is large, as is the case in MANs, the probability of incurring collisions because of token losses and multiple token generation processes will also be higher. Therefore, the token-passing protocols can only be perceived as being deterministic under normal system operation, but not under all conditions such as network initialization, getting in and out of the ring, network recovery, etc. Alternately, the transition period, the time starting with the non-deterministic network state phasing into deterministic state, can be shortened by implementing vigorous routines imbedded in the protocol itself.

Concerning the fairness issue, the token is passed to all stations attached to the network, thus each station is given equal chance to reserve a time slot in the channel, and only the station that has the token may access the medium. In the token-passing based media-access protocols load balancing with respect to fairness and guaranteed access measures can be made possible via two mechanisms: 1) controlling the number of visits per token-passing cycle per station (if one station has twice the traffic of the other stations, then visit it twice as often), and 2) controlling the maximum number of packets or frames transmitted per turn or visit [STUC 83]. In order to prevent one station hogging the shared channel by transmitting data continuously, a limit should be set for the maximum number of packets or frames that can be transmitted by the station that has acquired the token. Thus, each network station will get practically equal access and channel capacity according to its needs without hogging the network. For those MAN applications in which the fairness and guaranteed access characteristics are critical, the token-passing based media-access protocols are superior to other media-access schemes except the reservation protocols.

Even though the token-passing protocols, in essence, lend themselves to all MAN topologies, bus and ring topologies represent the most appropriate selection. However, the limiting factors are large end-to-end signal propagation delay, physical medium attachment, transceiver processing delay, etc. For the tree-and-branch MAN topology, the maximum attainable channel throughput is limited by the long signal propagation delay.

The robustness of the token-passing based media-access protocols is related to how well token losses and multiple token resolution are handled. Since the token in most implementations is not protected from errors (no error protection mechanism is normally used in the packet representing the token) there is a finite probability of losing the token. If the protocol procedures are rugged enough so that the channel can recover from token losses, multiple tokens, etc., then these protocols can be assumed to be "robust" under all network conditions. Otherwise, these protocols are not different from contention protocols in terms of stability and access resolution. One method of improving network reliability at the expense of channel throughput is to protect the token from errors by providing an error protection mechanism for the token itself. Robustness of a communication network goes hand in hand with network manageability and reliability. Since the majority of the token-passing based media-access protocols do not require centralized control, the system operation is no longer dependent on the reliability of a single station. On the other hand, token-ring networks require some sort of centralized control, therefore, they may be susceptible to a single point failure of the centralized network controller. In contrast, single or multiple network station(s) failure will not impact the total system operation, confining the problem only to the faulty stations. Moreover, token recovery and multiple token resolution procedures are part of the network management functions. The broadcasting capability and deterministic nature of token-passing protocols provide absolute command and control over all network stations, thus simplifying network management and ensuring reliable communications.

### 4.4 Contention-Based Media-Access Protocols

Contention is the generic name of a class of statistical media-access techniques which match the communication channel to the varying information traffic. Contention-based schemes, in general, are admirably suited to bursty or intermittent types of traffic load. Network stations access the channel on a message-by-message basis and, therefore, there is no concept of a "connection". Since there are no connections to be blocked, increasing traffic load results in a increased number of packet collisions and, consequently, longer access delay. The temporary overloading of a communication channel can be transferred into longer packet delays rather than blocked connections. Therefore, the contention-based media-access protocols provide the ability to trade packet loss for packet delay. Note that packet delays depend only on the traffic load accessing the channel and are independent of the number of stations accessing the network. Alternatively, the major disadvantage of these protocols is the inherent variable message delay, the existence of which cannot be tolerated in time-constrained applications such as real-time voice communications. In these protocols when two packets try to occupy the channel at the same time there will be a collision and both packets will be garbled. Hence, the mutilated packets must be retransmitted at a later time determined by a retransmission algorithm. Whenever a packet has an unsuccessful transmission, it incurs a retransmission delay equal to the amount of time measured from the packet's collision to its subsequent retransmission attempt. Each retransmission delay can be regarded as the sum of the deterministic delay due to the packet transmission and the random delay due to stochastic process of retransmission scheduling [LAM 74]. The retransmission delay is probably the most important variable in the design of networks using contention-based media-access protocols. Furthermore, the retransmission algorithm also determines the shared channel's throughput-delay-stability performance and dynamic behavior. It has been shown that the channel performance is dependent primarily upon the average value of retransmission randomization and quite insensitive to its exact distribution [LAM 74].

Regarding the performance measures, the contention-based protocols incur low access delays at low load conditions whereas degraded throughput and longer access delays are more typical at high load conditions. Under light loading conditions the contention-based media-access protocols are ideal due to low access delay with acceptable throughput level. As the load increases, contention-based protocols become increasingly less attractive because the overhead associated with channel arbitration becomes greater [TANN 81]. In retrospect, collision-free protocols incur high packet delays at low loading conditions, but as the load increases the channel efficiency improves considerably. It is obvious that the probability of some station acquiring the shared channel can be enhanced only by decreasing the amount of competition in the channel.

Several contention-based media-access protocols have been proposed and applied to radio, satellite and local area networks [ABRA 77, CROW 73, HANS 79, HOPK 80, KLEI 75a, KLEI 75b, KLEI 80, KONS 83, LAM 74, LAM 75a, LAM 75b, MAXE 82, METC 83, SIRA 84]. The origin of these techniques dates back to the Aloha protocol, that is, the stations transmit whenever they have messages to send [LAM 74]. There will be collisions, of course, and the transmitted packets may be mutilated. Due to the perfect acknowledgment property of the protocol, the sending station can always find out whether or not his packet was mutilated by just listening to the acknowledment channel. If the packet was destroyed, the station waits for a random amount of time and then retransmits the packet. This process continues until the packet is received intact by the destination station. It has been successfully shown that the maximum possible throughput of the Aloha protocol is limited to 18%, and this can be obtained only if the traffic on the channel equals the channel capacity with ideal conditions [LAM 74]. Note that the throughput of an Aloha channel is largely affected by the retransmission intervals that are chosen randomly. The key element of the Aloha protocol and its descendants is the retransmission traffic. If the rate of newly generated packets' traffic is increased, the rate of conflicts among stations increases to a point at which the channel traffic mainly consists of retransmitted packets, and the communication channel goes into saturation. Although the collision window in Aloha protocol is twice the packet length, resulting in maximum 18% channel utilization, the throughput can be improved by reducing the collision window to a packet length. The Slotted-Aloha protocol in which the transmission time is divided into equal size slots, performs twice as well as Aloha protocol because of the reduced collision window. The maximum attainable channel throughput of the Slotted-Aloha channel is only 37%, meaning that maximum 37 out of 100 packets can successfully be transmitted, the rest of the channel is wasted either due to collisions or empty slots [LAM 74]. In the Slotted-Aloha protocol a station is permitted to send packets only at the beginning of the slot time.

The basic operation of the Slotted-Aloha channel is exactly the same as the Aloha channel. It has been shown that the Aloha and Slotted-Aloha protocols perform well for lightly loaded systems [LAM 74, HAYE 81]. However, as the load increases the increasing rate of the retransmitted packets rapidly degrades the channel performance including the throughput and access delay. The throughput versus traffic characteristics of the Aloha media-access protocols are shown in Figure 4.6 [TANN 81]. Note that as the traffic load on the channel increases the channel may enter an unstable state, resulting in saturation, and the throughput approaches zero. On the other hand, in a communication network supporting a large population of users generating bursty traffic load consisting of small numbers of data packets, the Aloha protocols are ideal, especially in wide area networks such as MANs [SIRA 84]. Moreover, these protocols are appropriate for those applications in which the channel capacity must be allocated on a demand basis. In these systems a fraction of the channel capacity is reserved for contention slots required to promptly service user reservation requests. Several reservation protocols incorporate the Aloha protocols for dynamic channel allocation process [ROBE 73, KILL 80, JACO 77, HSU 78].

A considerable amount of effort has been spent in analyzing the Aloha protocols. These studies have demonstrated that the Aloha protocols are inherently unstable due to random statistical traffic fluctuations and retransmissions [LAM 74, TANN 81]. The stability of the

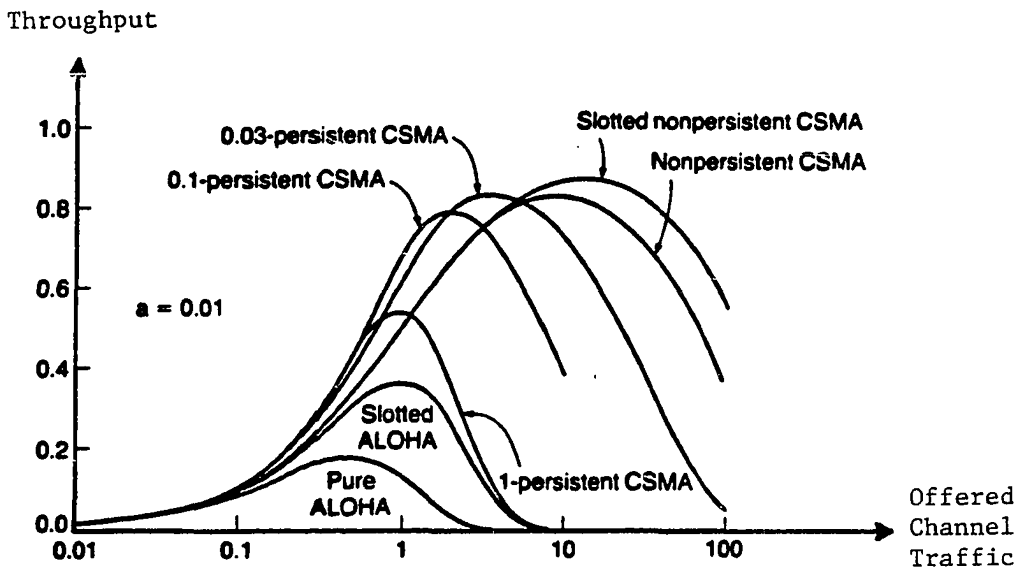



Figure 4.6. Contention Protocols: Throughput vs. Traffic

Aloha protocols can be improved by controlling the retransmission of packets and/or the traffic introduced to the shared channel. Several techniques have been proposed and implemented to enchance the stability and throughput of both the Aloha and Slotted-Aloha protocols [LAM 74]. However, these schemes proved to be inefficient in terms of access delay and channel utilization. In both Aloha protocols the throughput-delay-stability tradeoffs are closely interrelated, hence one parameter can only be improved at the expense of others. In summary, due to their low throughput, long access delay, unstable nature and inability to handle a large population of users generating varying load conditions, the Aloha and Slotted-Aloha protocols are not suited for MANs. Therefore, we will not elaborate more on these protocols, and will limit our discussion only to those media-access protocols in which carrier sensing is the basis for accessing the medium.

A logical extension of the Aloha protocols that is particularly appropriate for Local Area Networks is Carrier Sense Multiple Access, CSMA. In the CSMA protocol, a station attempts to avoid collisions by listening to the channel carriers of another station's transmission. Before transmitting a message the station listens to the channel, if the channel is free the station transmits its message, otherwise it refrains from transmission. Using carrier sensing before transmission greatly improves the channel throughput when utilizing the Aloha protocols. The increase in the throughput stems from the reduction in packet collisions, in turn mitigating the retransmissions and freeing the channel for the newly arrived packets. The variations on the basic technique involve the retransmission strategy. The retransmission strategies are defined by means of the P-persistent CSMA technique, that is, if the channel is busy then a station transmits its packet at the end of the current transmission with probability P, and with the probability of 1-P the transmission is delayed by the amount of the end-to-end signal propagation delay measured between two most distant network stations. The value of P is chosen so as to balance the probability of retransmission with the channel utilization. As is illustrated in Figure 4.6, decreasing persistency P leads to improved throughput which is obtained at the expense of increased access delay.

In the Non-Persistent CSMA protocol a station senses the channel before transmitting. If no other station is transmitting, absence of carrier, then the station acquires the channel to transmit its message. However, if the channel is already in use then the station does not continually sense the channel for the purpose of seizing it immediately upon detecting the end of the previous transmission. Instead, it waits for a random duration and then repeats the process. As is illustrated in Figure 4.6, the Non-Persistent CSMA exhibits better channel utilization, but introduces longer access delays than 1-Persistent CSMA scheme [TANN 81].

There may be more than one station transmitting on the shared channel at the same time due to signal propagation delay in which case collisions may occur due to inadequate carrier sensing. Hence, acknowledgments will be required to guarantee error-free transmissions. Furthermore, the stations will continue to transmit their messages without knowing they have collided. Since collisions will not be detected during the current packet transmissions, a certain amount of channel capacity proportional to the end-to-end propagation delay will be wasted. It is evident that the ability to sense the carrier from other stations leads to considerable improvements in throughput.

The performance of P-Persistent CSMA protocols can further be improved by adding collision detection or collision avoidance mechanisms. In the case of the Carrier Sense Multiple Access with Collision Detection (CSMA/CD) protocol a station listens to the channel while it transmits its message; if a collision occurs then the colliding stations intentionally send more data to ensure propagation of the collision throughout the system. The station remains silent for a random amount of time, backoff time, before attempting to retransmit. This mechanism decreases the amount of time during which a collision may occur, thus freeing the channel sooner than CSMA protocol.

In the case of the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) protocol, a station transmits a burst of redundant data, turning carrier on, to indicate to other stations that it wants to use the channel. If no other station collides with the data transmitted by the requesting station, then the station acquires the channel and transmits its actual messages. During this transmission no other station is permitted to transmit messages. This media-access protocol is reminiscent of the reservation protocols, simply reserving the channel by turning on the carrier to force other stations to back off. In both CSMA/CD and CSMA/CA protocols the channel throughput is enhanced by means of reducing the collision window or preventing collision at the beginning of packet transmission and while it continues.

Another variation of the CSMA protocol includes message-based priority functions, called Prioritized Carrier Sense Multiple Access (P-CSMA). The essence of this scheme is: the access right to the shared channel is exclusively granted to stations that have messages ready to send with highest priority level, reducing the competition over the channel by forcing lower-priority stations to refrain from transmitting. Similar to CSMA/CA scheme, the prioritization is done by the means of reservation bursts and carrier sensing [MAXE 82]. However, it has been clearly shown that P-CSMA scheme exhibits poor performance when the packet size is small [TOBA 82a].

In another variation of the CSMA protocol, further performance improvements have been obtained by means of a virtual clock to schedule message transmission. This scheme is referred to as Virtual-Time Carrier Sense Multiple Access (VT-CSMA) in which the virtual clock runs faster than real time, but only when the channel is idle, thus providing for each message a "virtual" arrival time during the channel idle period. It has also been shown that the Slotted VT-CSMA protocol is optimal over all possible CSMA variants under certain conditions [KONS 83].

Comparative analysis of CSMA, P-CSMA, VT-CSMA, CSMA/CD, P-CSMA/CD, and CSMA/CA protocols exemplifies the fact that all of these protocols will give essentially the same results, namely a response time that increases slightly with load for lightly loaded systems. The performance of the CSMA protocol begins to deteriorate at lower loads than other protocols in the same category, and P-CSMA/CD protocol has a performance slightly superior to the CSMA/CD protocol [OREI 84]. The performance differences are minor, therefore, we will only elaborate on the CSMA-based protocols.

In the CSMA/CD protocol the packet length must be twice the signal propagation time to the most distant network station. This requirement guarantees collision detection before transmission is completed, otherwise the channel utilization would be degraded by the amount proportional to the time wasted due to collisions. The propagation delay has an important effect on the performance of CSMA protocols and its descendants. There is a finite probability that just as a station begins sending, another station will become ready to send and sense the channel. If the first station's signal has not yet reached the second station, then the latter will sense an idle channel and will also begin sending, resulting in a collision somewhere on the transmission medium. The longer the propagation delay is, the more important this effect becomes, and the worse the performance of the contention-based protocols will be.

The signal propagation delay factor is defined as the ratio of the end-to-end signal propagation delay to the packet transmission time, and is denoted as "a". A value of zero for "a" means that the propagation delay is negligible as compared to the packet length, therefore, stations will have perfect knowledge of the channel status when they listen to it. This assumption leads to the highest possible throughput level for CSMA-based protocols. Alternatively, as the value of "a" increases, the carrier sensing process degrades until channel throughput is comparable to the basic Aloha protocols. Particularly, as is illustrated in Figure 4.7, all CSMA-based media-access protocols suffer in terms of usable channel capacity as "a" increases [ROSN 82]. For large values of "a" the throughput of the CSMA protocols may approach zero, leaving the channel totally blocked and saturated. It has been shown that the performance of the CSMA/CD protocol degrades significantly as the end-to-end signal propagation delay becomes a significant fraction of, or larger than, the transmission time of a packet. This is the case when the channel data rate becomes high, the distance covered becomes large, or the packet size becomes small [TOBA 82a]. In contrast, the channel throughput of the Aloha protocols is not dependent of "a" because the collision window is only determined by the length of a packet, but not the signal propagation delay. As a result, we conclude that the throughput of the CSMA-based protocols depends on the value of "a".

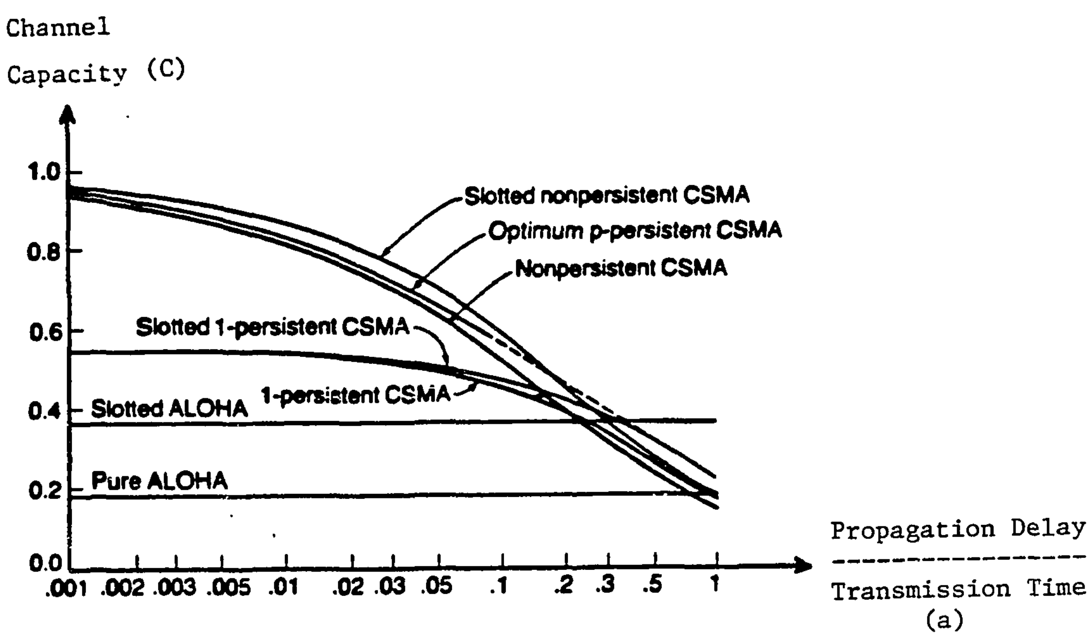



Figure 4.7. Contention-Based Protocols: Effect of Propagation Delay on Maximum Channel Capacity

As has been mentioned in the previous sections the end-to-end signal propagation delay in a MAN environment is inherently large, therefore, the throughput of the CSMA-based protocols will degrade to a point at which the performance of the Aloha protocols will equal or exceed that of CSMA-based protocols [MEIS 77]. It has been proven that CSMA and its variants perform well only if the end-to-end signal propagation delay is short compared to the transmission time of a packet and if all stations can listen to all transmissions on the channel [ROM 81]. Let "d" denote the data rate and let "l" denote the packet length. In a CSMA-based protocol, the product of "a", "d", and "l" equals a constant value ( a.d.l = constant) if the same throughput is to be obtained under all conditions. This means that if the value of "a" is fixed and the length of a packet is predetermined, the channel throughput can only be improved by reducing the data rate, decreasing the collision window. If the area coverage of a communication network using a CSMA-based protocol needs to be extended, this can be accomplished by increasing the packet length or decreasing the data rate, conforming to the formula given above. In another example, if the data rate is very high and packet length is kept very short, the area of coverage, as determined by the value of "a", should be confined to a short distance to attain respectable channel throughput levels.

In light of the above discussions, in order to use the CSMA-based protocols in a MAN environment, either the data rate should be decreased, lengthening the packet transmission time, or the minimum packet size should be lengthened. On the other hand, if the minimum packet length is increased or the data rate is reduced, the channel utilization will also be affected adversely. These ramifications would not suit those applications intended for MANs requiring high channel throughput. In a typical MAN environment it is expected that short and long packets will be generated due to the diversity of applications with varying traffic load characteristics. Therefore, the value of "a" may vary in a wide range, e.g., 0.0001, 0.5, etc. At high transmission rates CSMA-based protocols are very inefficient due to the time required to resolve contentions relative to the duration of a packet.

In order to minimize the effect of the end-to-end signal propagation delay and increase the channel throughput, a cellular P-CSMA/CD scheme has been proposed for coaxial-based MANs [MAXE 83]. In this scheme, the network is subdivided into small geographical areas and a distinct frequency spectrum is assigned to each subdivided area. This approach keeps the packet collision distances small, even though the network size is large. However, the complexity of interchannel communications and the impracticality of using frequency division multiplexing in a cellular fashion may not provide an economical solution for MANs.

In conclusion, due to their throughput dependence on the end-to-end signal propagation delay, CSMA-based protocols including all other contention-based protocols will not perform satisfactorily in MANs unless the propagation delay effect can be alleviated in some fashion.

Before investigating the large user effect on the CSMA-based and contention-based media-access protocols we must first consider all the parameters that affect the performance of these protocols. As discussed earlier, the end-to-end signal propagation delay and packet length variables play a dominant role in determining the channel throughput and indirectly the access delay. Another important performance parameter is the number of ready messages to be transmitted on the shared channel. What determines the channel throughput is not the number of stations attached to the network, but rather the traffic seen on the channel, namely the traffic generated by the active network stations. Clearly, the larger the number of stations attached to the network, the higher the probability of introducing more traffic into the channel. It can, therefore, be stated that there exists an indirect interrelationship between the throughput and the number of stations attached to the network. Considering the "Large Number Law" it is expected that there will be higher density of traffic on the channel due to the large population of users sharing the channel. In contrast, there may exist one large user transmitting a large amount of data while the rest of the stations have little to transmit. Even in this case, the average traffic seen on the channel will still equal the sum of the average demands generated by a large population of users. However, if there is only one station with large amounts of data to transmit while others introduce a bursty traffic load to the channel, the station with the high density of traffic will see less competition and will acquire the channel for the duration permitted for a single station, in turn decreasing collisions and improving the channel utilization. It is obvious that this is a rare case and over a long period of time the traffic seen on the channel will average and probably will equal the average traffic load input to the channel. If the total offered traffic load on the channel is fixed, but the number of stations is increased, more and more collisions will take place to transmit one message, hence the mean delay and message waiting time will increase proportionally. Under the conditions when a large population of users have messages to transmit, it can be stated that the occurence of collisions is a linear function of the number of stations requesting channel time.

In Figure 4.8, the channel capacity or utilization is plotted against the number of stations queued with varying packet sizes. At a fixed packet length, the throughput quickly approaches the saturation point and any increase in the traffic load has only a slight effect thereon. The channel throughput, as it is illustrated in Figure 4.8, also depends on the packet length. As the packet length increases the channel throughput improves dramatically. For very short packet lengths, the throughput may even reach levels that are not acceptable even for non-time-constrained applications such as terminal-to-host communications, electronic mail, etc. Alternatively, the mean waiting time to transmit a message grows faster than any linear function of the number of stations attached to the network [STUC 83].

In communication networks using CSMA-based protocols, the traffic can also be controlled by limiting the maximum number of packets that can be transmitted by each station; this allows stations with bursty traffic load a chance to transmit their messages, while imposing longer delays for those stations with high-density message traffic. In a typical MAN supporting a large population of users with a diversity of applications, the CSMA-based protocols should allow tolerable access delays while maintaining acceptable throughput levels.

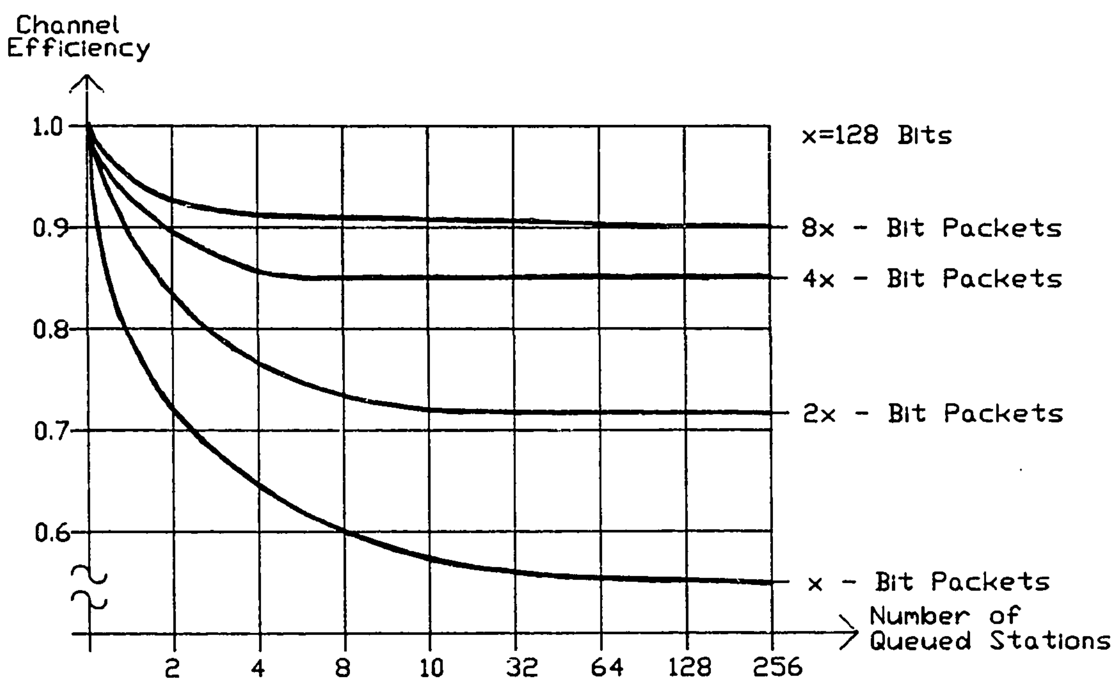



Figure 4.8. CSMA-Based Protocols: Channel Capacity vs. Queued Stations vs. Packet Length

The CSMA-based protocols allow almost immediate channel access under light traffic loads, therefore, they are ideal for data communications in which large numbers of geographically distributed stations tend to generate demands in a very low duty cycle. There is no guarantee of packet delivery time due to their nondeterministic nature, thus when the traffic loads become heavy and continuous CSMA-based protocols suffer from packet delays with a large mean and variance. In the case when stations generate instantenous high-density traffic loads, the channel capacity will be exceeded, resulting in transient conditions affecting access delay and channel throughput. In the CSMA-based protocols, this means that a large number of stations will see collisions, back off, retransmit, and again collide. The end result is: the protocols in question may assume a nondeterministic nature since the waiting time may become infinite. Under these conditions the longer and variable access delays with lower channel throughput levels may be acceptable for data communications. In contrast, for time-constrained applications such as real-time voice communications the packets cannot experience long and variable delays. If the number of real-time voice packets can be limited so that reasonable delays with a minimum amount of variance can be obtained, the transmission of voice packets in conjunction with data packets will be possible without performance degradation. It has been demonstrated that a LAN, intended for real-time voice communications, using a CSMA/CD protocol will only suffer about 2 percent or less packet loss, resulting in no perceivable replay degradation [MUSS 83]. In another study where only voice communications are considered over a channel utilizing Prioritized-CSMA protocol, it has been proven that the performance of P-CSMA protocol is comparable to that of the ideal TDMA scheme [TOBA 82a]. A variation on the CSMA/CD protocol that yields movable time-division multiplexed slots has been proposed and implemented for real-time voice communications [MAXE 82]. In this scheme the voice packets are assigned higher priority than data packets. It has been demonstrated that LANs utilizing a CSMA/CD protocol or its variants can support voice communications as well as data communications [BRUT 83, MAXE 82, NUTT 82]. However, under heavy loading conditions the CSMA-based protocols will suffer in terms of longer delays, increased delay variance, and reduced throughput.

By virtue of the previous discussion it is evident that the CSMA-based protocols can handle a broad range of applications including data, packetized voice, and digitized/compressed video. However, it should be noted that the traffic mix is an important factor. Longer access delays may be observed for data traffic while a large fraction of the channel capacity is being utilized for voice packets. Similarly, while a large amount of data, e.g., file transfers, digitized video, etc., is being transmitted the voice packets will experience longer delays of varying degrees, resulting in unacceptable network conditions for time-constrained applications. As a result, it can be stated that the CSMA-based protocols are not ideal for integrated services, but they can provide a reasonable means of channel arbitration among a large population of users.

The tradeoffs between throughput, access delay, and stability should be investigated for the CSMA-based protocols because an improvement in one parameter comes at the expense of the others. For example, if the channel throughput is to be maximized, the access delay has to be penalized, causing longer and variable delays. In contrast, if the access delay is to be minimized,the increased number of collisions lowers the channel throughput. The stability of the CSMA-based protocols is closely related to the retransmission strategies. Improvement in the channel stability can only be achieved by spreading the retransmissions over a longer time window, which in turn increases the average delay and reduces the channel throughput. In order to evaluate the impact of one measure on another, the characteristics of the services being provided over the network must be known. For example, for packetized voice communications bounded access delay and lower delay variance with good channel stability are required rather than high throughput. In data communications, longer access delay and delay variance can be tolerated, but high throughput is required.

Several back-off algorithms have been proposed to enhance channel stability while maintaining high throughput and low access delay [METC 83, TOBA 77, LAM 74]. It has been observed that the retransmission algorithms perform differently under various load conditions. Under lighly loaded system conditions the access delay may be lengthened due to the inherent properties of the retransmission algorithms while channel stability is enhanced under heavily loaded systems. In Figure 4.9, the normalized delay in terms of packet lengths is plotted against the throughput. It is obvious that the channel throughput can be maximized with an optimum value of persistence P. If the network must be operated at high throughput levels then longer access delays will be tolerated. At low throughput levels the access delay is minimized so that immediate transmission of packets is possible.

The attainable throughput levels for the CSMA-based protocols may vary from 50% up to 99% at various load conditions, and it has been shown that very high throughput levels can only be obtained with large packet sizes [METC 83]. Providing higher channel bandwidth with the hope of achieving a network throughput proportional to the channel speed gives only marginal improvement. Moreover, it has been shown that the overall performance of a channel can be improved by using a protocol that assigns different probabilities to different stations with rotating assignment on each slot to ensure fairness [TANN 81]. Alternatively, in a voice/data communication network using a CSMA/CD protocol the average access delay and its variance can be reduced by giving the periodic traffic, i.e., voice packets, higher retransmission priority [MAXE 82]. As stated previously the contention-based media-access protocols perform fairly well under light load conditions. However, under heavy traffic load conditions the access delay will be lengthened such that it is statistically possible to render the channel unstable if the backlog grows faster than the maximum channel capacity.

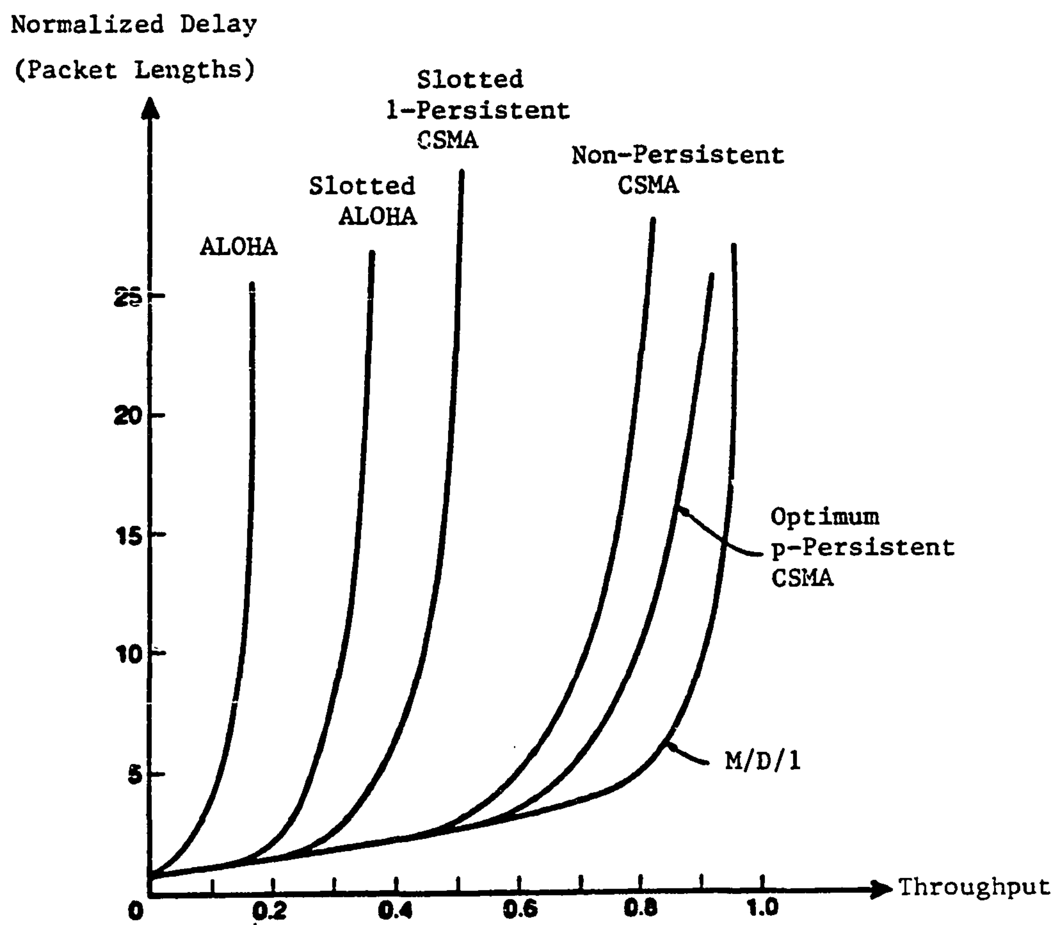



Figure 4.9. Contention Protocols: Throughput vs. Normalized Delay

Due to the possible increase in the number of collisions on the channel, the CSMA-based protocols are sensitive to load fluctuations. However, they would eventually adapt to the traffic variations within a certain period of time that is determined by the retransmission algorithm. Ideally, an adaptive back-off algorithm that matches the fluctuations of the offered channel traffic will be able to offset the deleterious effect of varying load conditions.

Because of the randomization in the retransmission algorithm the CSMA-based protocols are basically nondeterministic in nature. As a result, as channel traffic load increases the channel throughput approaches zero, and packet transmissions are suspended. The nondeterministic nature of the contention-based protocols brings out the issue of channel stability. As pointed out earlier channel stability can be enhanced by means of controlling the retry time interval and limiting the traffic input by the network stations. This subject will be discussed later in the section. For time-constrained applications such as real-time voice communications and factory automation, the contention-based media-access protocols are not well suited because of their unbounded access delay and inherent delay variance. Since the time required to resolve the conflicts is not bounded, the access time cannot be estimated correctly and, therefore, guaranteed access cannot be ensured. Not being able to predict the worst case delay prevents the contention-based protocols including CSMA and its variants from being used for those applications in which guaranteed access and bounded delay with respectable throughput levels are the requirements.

A communication channel may not provide guaranteed access, but it may well lend itself to fair channel access among all network stations. As long as all network stations are assigned the same priority and can sense the carrier equally, the channel access is considered to be fair. To ensure fairness across the network, regardless of the existence of weak and strong stations, no capture effect phenomenon is permitted; that is, a station which has stronger signal strength and a station which has lower signal strength may acquire the channel with equal probability. If collision detection can be guaranteed at all channel conditions, leading to a 100% collision detection capability, then the fairness criterion on the channel can be met.

All the CSMA-based protocols with the exception of some descendants provide fair access to the channel under all circumstances. However, some of the CSMA-based protocols utilize priority mechanism to increase the channel throughput and/or to allow real-time applications such as voice communications to coexist with data communications on the same channel [MAXE 82, TOBA 82a]. If desired, all contention-based media-access protocols can be configured to provide fair access under all network conditions.

The overhead associated with the CSMA-based protocols consists of collisions due to conflict resolution, end-to-end signal propagation delay, minimum frame length, interframe gap, and carrier sensing. As stated earlier, the collisions that may occur because of channel access resolution can be reduced by decreasing competition in the channel. The interframe gap that represents a fraction of the overhead is required because there must be an interval between packets on the network so that a receiving station can recognize the end of packet and prepare to receive a new packet. It is evident in MANs that the end-to-end signal propagation delay is the major element contributing to the overhead. In order to ensure collision detection across the communication channel, the minimum packet size is determined by the end-to-end signal propagation delay, thus a station's packet transmission continues until all other network stations can, at least, sense the carrier from the sending station. If the end-to-end signal propagation delay is large, some fraction of the channel capacity will be wasted due to the requirement of minimum frame size. Another contributing factor to the overhead has to do with the carrier sensing capability of each network station. The time taken to sense the carrier is also an important factor in determining the overall channel performance. Due to the carrier sensing process, in other words, the time lost at the beginning of each packet transmission, some fraction of the channel capacity will also be wasted. It is clear that the overhead caused by the CSMA-based protocols is less than that of the token-passing and polling protocols.

The essence of contention-based protocols (here we will only consider the CSMA-based protocols, not the Aloha protocols) is the carrier sensing capability. This means that all network stations should be able to listen to other stations' transmissions at all times. Therefore, bus network topology lends itself to the implementation of CSMA-based protocols. In coaxial-based broadband networks where the signal flow is unidirectional, the carrier sensing is done over the broadcast channel, the downstream channel. This indicates that the end-to-end signal propagation delay would be twice as large as the one-way signal propagation time. Even though the CSMA-based protocols can be applied to all network topologies including the star topology, the end-to-end signal propagation delay and collision detection capability should be the main considerations in choosing the proper topology.

Because of the decentralized control characteristics the operation of CSMA-based media-access protocols is not dependent on any central controller. Furthermore, single or multiple station failures will not, under all circumstances, impact the network operation. Communication networks utilizing contention-based media-access protocols including CSMA and Aloha are considered to be robust and reliable under all circumstances because stations have perfect information about the channel status at all times. The reliability and robustness of the contention-based protocols can further be improved by providing each station with autonomous control and utilizing a broadcast channel to pass valuable channel information to all network stations simultaneously. The communication networks using the contention-based media-access protocols are readily managed because of their broadcasting capability. A network management function can easily be instituted to further improve network stability and reliability. In other words, a network manager may send network related information without interrupting the normal system operation. On the other hand, the stations should be equipped with sufficient intelligence so that they can make certain decisions independent of each other. In light of the above discussions, we recommend that a network management function should be instituted for MANs, and the contention-based media-access protocols are recommended for this use.

### 4.5 Reservation-Based Media-Access Protocols

The essence of reservation-based media-access protocols is the ability to allocate a fraction of the communication channel capacity to a single station, thus preventing interference from the other stations. The channel capacity allocation function can be implemented either in a centralized fashion where a central controller assigns time slots, or in a distributed fashion where stations follow a set of instructions to allocate time slots in a predetermined manner without requiring any centralized control. The allocation of the channel capacity can take two forms: 1) static or fixed assignment, and 2) dynamic or demand assignment. With the fixed assignment technique, the channel is allocated to users independent of their activities by simply partitioning the time-bandwidth space into slots that are assigned to the network stations in a static predetermined fashion. This technique also takes two common forms: 1) "orthogonal" such as frequency-division multiple access (FDMA) and synchronous time-division multiple access (STDMA), and 2) "quasi-orthogonal" such as code-division multiple access (CDMA). In a communication network where all stations are assumed to be identical, FDMA consists of assigning to each station a fraction of the total bandwidth along with the buffering capability required to handle the statistical fluctuation of traffic load due to the random message arrivals. STDMA consists of assigning a fixed number of channel time slots to each station whether or not the station has messages to send. This scheme results in assigning a fraction of the total channel capacity and also requires message buffering capability. The Fixed Priority-Oriented Demand Assignment (FPODA) protocol and Binder Method are typical examples of the STDMA technique [TANN 81, FALK 83]. A number of disadvantages of the FDMA techniques as compared to STDMA are: wasted bandwidth due to sub-channel frequency separation, lack of flexibility in achieving dynamic allocation of bandwidth, and lack of message broadcasting capability. The only major disadvantage of the STDMA technique is the requirement to provide synchronization and time slot separation. In CSMA-based protocols, more than one station may share a common channel in a nondestructive fashion [TOBA 76]. For example, stations or a group of stations can share a common slot(s) by taking turns in a predetermined fashion.

With the dynamic or demand assignment technique, a station makes a request for allocation of channel capacity whenever it has a message to transmit. The channel time slot allocation mechanism is activated only when such a request is made. If a station is given a time slot only when it has a packet of information to transmit, the mode of channel operation is asynchronous, and this technique is called Asynchronous Time-Division Multiple Access, ATDMA. The protocols and slot allocation rules characterize the architecture of an ATDMA protocol.

Due to rapidly varying demands to share a channel among a large population of users, ATDMA protocols are generally utilized in Radio and Satellite Networks. Variations of ATDMA protocols such as Urn, Reservation Aloha, Yemini Method, Adaptive Tree Walking, Robert's Method, Contention Priority-Oriented Demand Assignment (CPODA), Crowther Method, Broadcast Recognition with Alternating Priorities (BRAM), Multi-Level Multiple Access (MLMA), Global Scheduling Multiple Access (GSMA), Generalized TDMA, have been proposed and implemented for those applications in which fixed channel time allocation was not acceptable. The reader is referred to the following references for detailed information concerning the protocols mentioned above: [ARTH 82, BOSE 80, CAPE 79, CHLA 80, FALK 83, HANS 79, KILL 80, KLEI 80, KUO 81, LAM 74, LAM 79, MEIS 77, STUC 83, TANN 81].

A station may request a time slot either by using the actual message transmission channel in a shared fashion or by using a separate request channel. Several contention-based protocols, particularly Aloha, Slotted-Aloha, and CSMA protocols, have been suggested as a method for multiplexing the requests on the channel [CAPE 79, KILL 80, KLEI 80, LAM 79, MEIS 77]. It has also been proven that major improvement is gained when the request channel is operated in Slotted, Non-Persistent CSMA as opposed to the Aloha protocols [TOBA 76].

In order to prevent collisions between the request packets and the actual message packets, the channel is either time-divided or frequency-divided between two types of data [TOBA 74]. In both cases, the available bandwidth is divided into two logical channels: one used to transmit request information, and the second used for actual messages themselves [TOBA 76]. Dynamic allocation of the channel time slots can be accomplished in two ways: 1) centralized, and 2) distributed. In a centrally controlled demand assignment system, the central controller receives the stations' requests and allocates channel capacity, in terms of time slots, according to the need of each requesting station. This type of channel allocation results in higher throughput in the request channel along with improved channel stability. If the channel time slot allocation is done in a distributed fashion, then the rules established for slot reservation should strictly be followed by all stations. However, this will result in lower throughput over the request channel because of the collisions caused by stations requesting channel capacity on a demand basis. For example, in a communication network in which an ATDMA protocol is used along with a CSMA-based request channel, the carrier sensing slot is made equal to the end-to-end signal propagation delay, thus preventing extraneous collisions and, at the same time, compensating for the signal propagation delay [HANS 79].

So far, we have discussed several reservation techniques that are commonly used and applied in Radio and Satellite Networks. Reservation protocols have not yet been applied tc Local Area Networks because of their inefficiency under lightly loaded systems and their poor traffic handling characteristics for a large population of users. A considerable amount of effort has been spent on the analysis of several reservation protocols and their descendants. These techniques are well understood and are discussed in detail in the literature [ABRA 77, BALA 79, FALK 83, HSU 78, HUYN 77, JACO 77, KILL 80, KLEI 80, LAM 78, NUSP 77, ROBE 73, SAIT 79, TOBA 76]. We concentrate on studying their implications and applicability to Metropolitan Area Networks supporting integrated services such as data, packetized voice and digitized/compressed video.

Reservation protocols are mainly intended for Radio and Satellite Networks where the signal propagation delay is inherently long, thus the effect of propagation delay has been carefully studied in these protocols. Clearly, reservation protocols can easily and successfully be applied to wide area networks such as MANs because the signal propagation delay is compensated for and has almost no impact on the performance of the network with the exception of long transmission delays. It has been proven that when the value of "a" is large the reservation-based media-access protocols are the best schemes in terms of channel throughput [KLEI 80]. In contrast to CSMA-based protocols, as the end-to-end signal propagation delay gets longer the channel throughput does not degrade, rather it stays unchanged. As a matter of fact, in geosynchronous satellite networks where the end-to-end propagation delay is in the range of 250 milliseconds, STDMA protocols are successfully used with the throughput reaching the ideal level of M/D/1. The effect of the signal propagation delay is reflected as the time difference between the reverse transmission and the forward reception. The time base difference between these channels equals the end-to-end signal propagation delay [FALK 83]. It is evident that reservation protocols do not suffer from large signal propagation delays and, therefore, they are well suited for MANs.

It has long been recognized that fixed allocation of a scarce communication resource is extremely wasteful when the channel is being shared by a large population of users with bursty traffic load. In general, if stations are willing to tolerate greater message delay, then more stations can share the same channel. As it is illustrated in Figure 4.10, the average access delay is almost a linear function of the number of stations accessing the channel. The minimum achievable average delay increases directly proportional to the number of stations for fixed data rate and message length. If the number of stations is small, then the fixed assignment of a common channel, namely STDMA, is desirable.

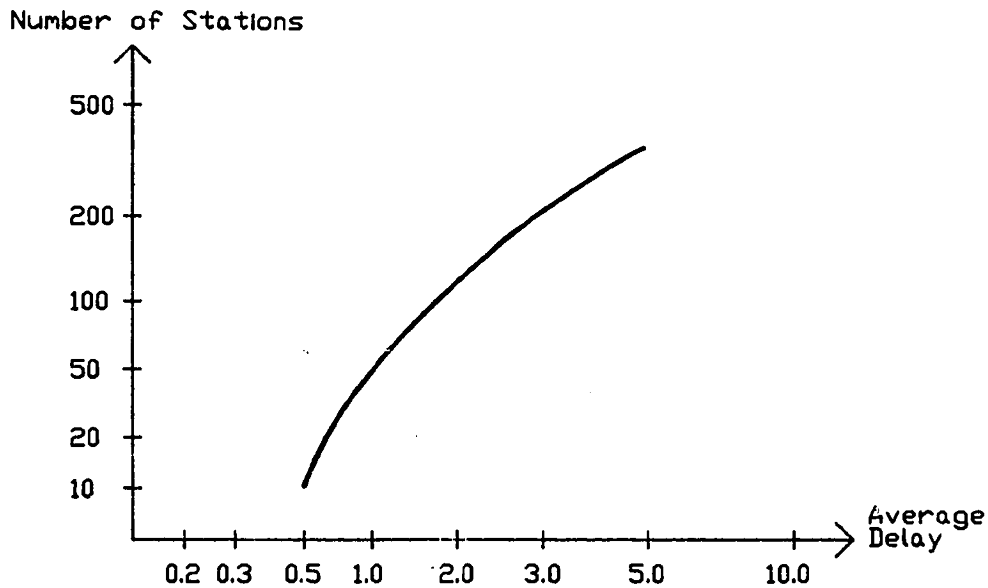



Figure 4.10. Reservation-Based Protocols: Number of Stations vs. Delay

For a large population of low-duty cycle stations, mixed station populations, or situations in which traffic requirements are not well known, ATDMA schemes are prefered. If the channel time slots are allocated in a fixed fashion, then it is clear that the number of stations that can be connected to the network will be limited. On the other hand, if the time slots are assigned dynamically on a demand basis, then a larger number of stations can be serviced. Higher traffic density on the request channel results in a lower probability of success, thus when there is a large number of stations requesting time slot(s) the request channel will see a surge of requests causing performance degradation in the request channel; in turn, fewer stations will be able to reserve time slots.

In a typical ATDMA scheme the actual message slots are assigned according to the channel capacity need of requesting stations. If a large population of users requests channel access simultaneously it will take longer period of time to resolve the reservation requests. In this case, the transmission resource will be allocated gradually until the channel capacity is totally utilized. This situation may cause longer message delays when a large population of users are requesting time slot(s). As illustrated in Figure 4.11 with STDMA protocols, as the number of stations increases the average access delay will increase proportional to the number of stations accessing the channel. Alternatively, for ATDMA protocols, see Figure 4.11, if the number of stations accessing the channel is greatly increased the average access delay will increase only slightly. This means that, in terms of access delay and channel throughput, ATDMA protocols basically can support a larger number of stations than STDMA protocols. As a general result, it can be stated that the reservation-based media-access protocols can support a large population of users if the offered traffic load is bursty in nature and not all network stations are requesting channel time slots simultaneously. Thus, the reservation-based media-access protocols are very appropriate for applications such as LAN interconnections, real-time voice communications, and bulk data transfer, which are typically supported by a MAN.

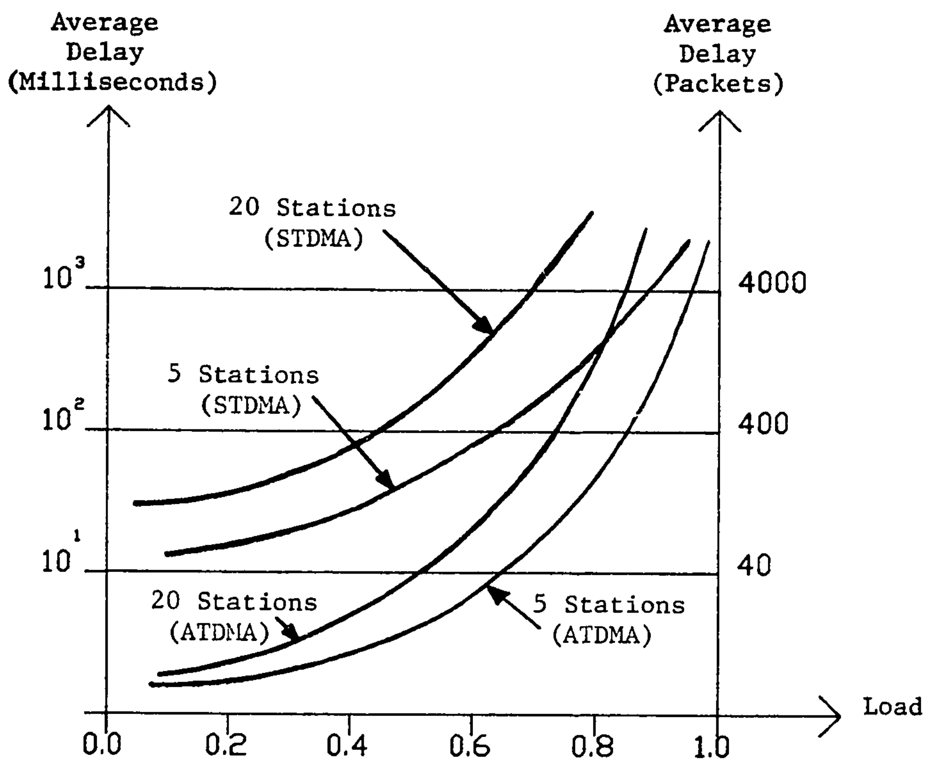



Figure 4.11. STDMA and ATDMA Protocols: Average Delay vs. Number of Stations

When a large population of users with bursty traffic load characteristic is to be supported for typical data communication applications, the reservation-based media-access protocols perform poorly because of the time it takes to reserve and release a time slot. It has been established that the reservation protocols, in general, are inferior to contention protocols under light traffic loads [TOBA 76]. In the case of terminal traffic wherein messages arrive randomly and sporadically, the message access delay of the reservation-based protocols will not be acceptable. Therefore, it is concluded that these protocols are not appropriate for typical data communications with the exception of those applications such as file transfer, bulk data transfer, and interconnection of data multiplexers, where the messages require a large number of time slots and the system is heavily loaded. In this case, a large portion of the channel capacity can easily be reserved for the transmission of actual messages. The reservation-based media-access protocols, in general, allow maximum channel utilization on a demand assignment basis, providing an ideal channel access mechanism for transfering large amounts of information. Therefore, these protocols are very appropriate for those applications requiring high channel throughput and guaranteed access over MANs.

Real-time voice communications can readily be provided by means of the reservation protocols since the channel access delay variance is not an issue due to guaranteed channel access. In essence, these protocols are particularly tailored for voice communications in which the traffic load is continuous and periodic. Typically, when a voice circuit is to be established between two stations, a time slot reservation is requested. Once a time slot is reserved, a permanent circuit is established, and it is released when the conversation is over, allowing other stations to utilize the time slot. For multiplexed voice circuits, STDMA or FTDMA protocols are commonly used. It is unequivocally evident that for those MAN applications wherein the traffic load is continuous and non-bursty, the reservation-based protocols will be superior to other protocols including polling, token-passing and contention-based schemes.

For digitized and compressed video applications in which the traffic load is slowly-varying, bursty, and at times heavy, it is reckoned that the reservation-based media-access protocols will perform exceptionally well. For example, in video conferencing applications a connection must be established for the duration of the session. This implies that time slot(s) will be reserved till the session ends at which time the slots can be released. As in bulk data transfer the digitized and compressed video images contain large amounts of information, and the traffic characteristic is fairly bursty in nature. Since the transmission of digitized and compressed video information requires a large portion of the communication channel capacity for a limited period of time, an adequate portion of the total channel capacity can be reserved dynamically for the duration of the transmission, and released when the channel is no longer needed. It appears that reservation-based media-access protocols are very suitable for the digitized and compressed video communications assumed to be supported in MANs.

One of the reservation protocols' parameters that impacts the channel traffic-handling characteristics is the traffic concentration per frame slot. The benefits resulting from concentrating more traffic per station can be extrapolated as a function of average traffic per station. It has been proven that for sufficient traffic concentration the reservation-based media-access protocols can support an arbitrarily large number of stations and still have an acceptable average access delay [KUO 81]. Another important protocol parameter that affects the channel throughput is the slot length. If the channel time is highly partitioned, creating slots of small transmission capacity, then a greater fraction of the channel capacity will be dedicated for slot synchronization, thus increasing the overhead. The larger the number of slots, the lower the channel utilization. If the messages consist of many multiples of slots then decreasing the slot size may eventually decrease the overall channel utilization, because the overhead carried with each slot may amount to a larger fraction of the channel capacity than that of larger slot sizes. In contrast, if the slot size is very large and the traffic generated per slot is much less than slot transmission capacity, then a larger fraction of the channel capacity will be wasted as compared to smaller slot size configuration, resulting in lower channel utilization. In conclusion, the slot size plays an important role in determining the channel throughput.

In ATDMA reservation protocols, some fraction of the channel capacity is reserved to allow reservation requests to be made over the communication channel. The amount of channel capacity reserved for time-slot requests largely determines the channel throughput. As the overhead associated with time-slot requests and slot synchronization increases the channel throughput will suffer accordingly. However, the channel efficiency can be improved at the expense of longer access delays by simply decreasing the request channel capacity or by increasing the actual data frame size.

Another parameter affecting the channel throughput of the reservation-based protocols is the average message length of the offered network traffic. Since the number of reservation bits required to specify the message length being reserved is small compared to the total burst overhead for the reservation packet, the channel efficiency increases with increasing message length.

Reservation-based media-access protocols, in general, are not very sensitive to changing traffic load conditions. For a fixed channel rate, the increased delay due to the support of a larger number of stations applies only to the reservation portion of the communication channel. With high traffic concentrations and non-varying traffic load conditions the fixed channel capacity assignment, as in the STDMA technique, will give the best overall performance. In the case of bursty traffic or variable loading conditions, the demand assignment protocols, like the ATDMA schemes, will provide significant improvements in both channel efficiency and access delay.

In a MAN environment, the reservation-based media-access protocols, in general, will provide an acceptable level of channel utilization and low access delays for heavily loaded systems, but low utilization and longer delays for lightly loaded systems as compared to polling and token-passing protocols. It is unlikely that the channel throughput and delay performance of the these protocols will be affected by changing load conditions.

In FDMA and STDMA techniques a station simply transmits messages in the slots assigned to it. The shortcoming of these techniques for lightly loaded systems is the fact that very often stations with terminal type bursty traffic have nothing to send, thus channel capacity is wasted due to unused reservation slots. By the same token, some empty slots may be passing by stations that require more channel capacity and have messages to send. The access delay in the fixed assignment reservation protocols is actually predictable and in the worst cases it has a fixed value. Since all messages have a deterministic waiting and transmission time, and 100 percent of messages are delivered with a fixed time delay, a severe price is paid in terms of channel utilization for this 100 percent reliability and fixed delay. However, these attributes of the fixed reservation protocols are highly desirable for time-constrained applications such as packetized voice communications, etc.

The access delay can easily be calculated by knowing the assigned channel capacity and the amount of offered traffic load. The channel throughput of the explicit reservation protocols is not influenced by the protocol parameters such as access delay and stability, rather by the traffic density at each network station. This means that in order to obtain higher channel throughput levels, higher traffic concentration, namely heavy traffic loading, will be required at each station to which some fraction of the total channel capacity is assigned.

In view of the limitations presented here, it is concluded that for lightly loaded systems the reservation-based media-access protocols, in which fixed assignment technique is utilized, are inferior to load adaptive techniques such as token-passing and contention-based protocols.

In the case of ATDMA protocols based on a demand assignment technique, there is an increase in the utilization of the communication channel over the STDMA protocols at the cost of an increase in the complexity of the stations. For lightly loaded systems ATDMA reservation protocols are superior to STDMA protocols by a factor equal to the number of stations sharing the channel [HAYE 81]. As the traffic load increases the difference between STDMA and ATDMA protocols decreases because more and more stations may have messages to send.

The access delay in demand assignment reservation protocols is a function of the number of stations attached to the network and loading conditions on the communication channel. Since the reservation requests are sent over an access control channel, a separate channel or fraction of the actual data channel, as the number of stations requesting channel capacity increases, more and more collisions will be observed on the control channel, thus lengthening the access delay. This delay can be reduced by simply increasing request channel capacity at the expense of the channel utilization, resulting in lower channel throughput. In this case, only control channel throughput will be affected because the actual message slots do not impact the total channel throughput due to the fact that there is no message interference over the shared channel. Depending on the loading conditions on the communication channel throughput and delay can be traded off at the expense of the other.

Several techniques have been proposed to resolve and reduce the competition over the request channel while preserving acceptable throughput levels [CAPE 79, KLEI 80, LAM 74, TANN 81,TOBA 76]. The average access delay is plotted against channel throughput for various reservation protocols (see Figure 4.12). It is evident that higher throughput levels with lower access delays can be achieved with the demand assignment reservation protocols as opposed to the fixed assignment reservation protocols. It is noted that the demand assignment reservation protocols may reach ideal perfect scheduling, M/D/1, throughput level faster than the fixed assignment reservation protocols even with lower access delays (see Figure 4.12). With Urn and BRAM schemes in which the protocol behavior is load adaptive, very high channel throughput levels can be achieved with very low access delay [KURO 84]. If the channel capacity is totally used then a phenomenon of station blocking may take place and long access delays may be seen by those stations trying to request channel capacity.

Since the actual data slots are reserved according to a deterministic scheme, stability is an issue only for the request channel. If contention-based techniques are used over the request channel, then the same stability considerations presented for these protocols will apply in the same manner. Therefore, we will not elaborate more on the stability issue for the reservation-based protocols. However, these protocols are proven to be very rugged and stable.

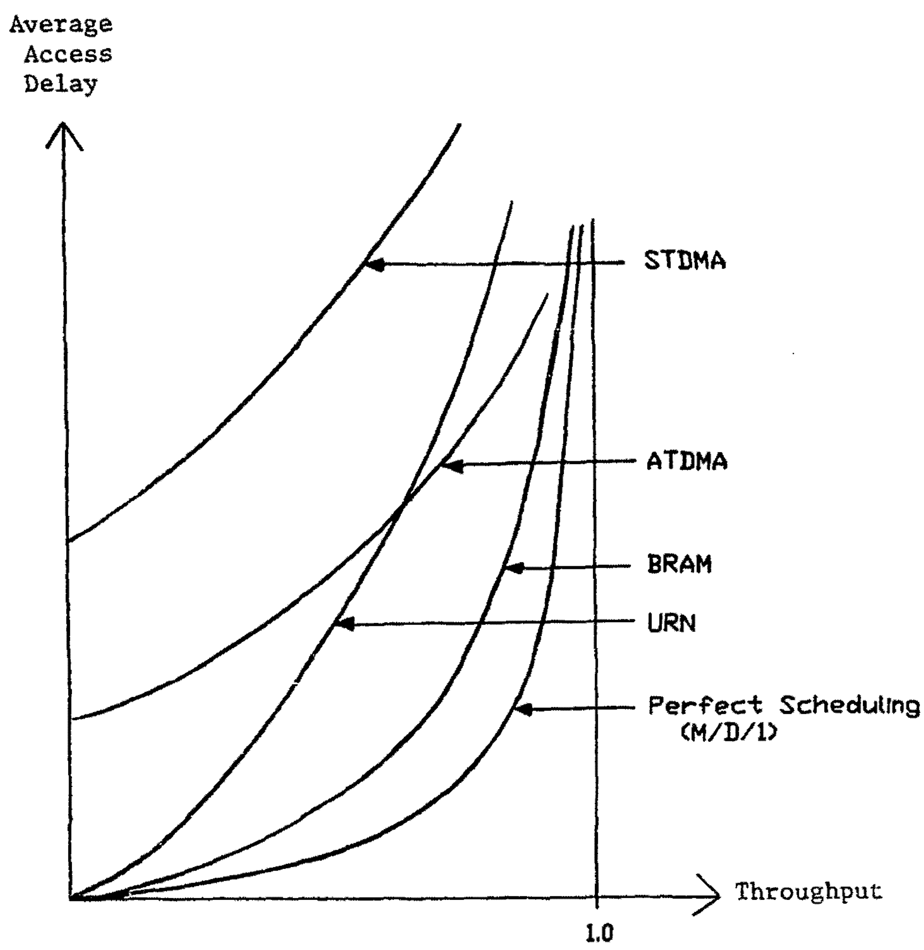



Figure 4.12. Reservation-Based Protocols: Throughput vs. Average Access Delay

When considering a MAN environment it seems that a reservation protocol using distributed control and demand assignment technique will be ideal for lightly loaded systems with bursty traffic, whereas a reservation protocol using centralized control and fixed assignment technique will be superior to other protocols for heavily loaded systems.

Since the essence of the reservation-based protocols is based upon a deterministic approach, the channel access can equally be guaranteed under all network conditions. Concerning the fairness issue for fixed assignment schemes, all network stations may send their messages only in the slots assigned to them, and access to the shared channel is equally guaranteed under all circumstances. In demand assignment techniques all network stations have equal chance, with the same probability, to request channel capacity. Once a station is successful in its reservation request it owns the time slot until it releases it intentionally, thus the channel access is guaranteed under all network conditions. The fairness of the channel access should only be considered on the request channel. As discussed previously, contention-based and load adaptive protocols are commonly used to resolve the reservation requests. The arguments presented for these protocols are also valid for the demand assignment reservation protocols. In summary, the reservation-based protocols are ideal for those applications wherein fairness and guaranteed access is important.

The fixed and demand assignment reservation protocols can readily be applied to all MAN topologies. In particular, star and bus topologies are ideal for centrally controlled reservation protocols. Since frame and slot synchronization is required for the implementation of the protocols, the network topologies that permit broadcast transmission will be very appropriate. It can clearly be stated that all MAN topologies lend themselves to the implementation of reservation-based media-access protocols.

Since centralized control, and frame and slot synchronization are required for the reservation-based protocols, single-point failure becomes a concern in terms of network reliability and robustness. To alleviate the possibility of a single-point failure, the implementation of a fault-tolerant central network controller is highly recommended. Since message broadcasting is provided with the existence of a centrally located network controller, the network management and diagnostic functions can readily be performed in networks with stations scattered over a large geographical area. The reservation-based protocols, in particular, allow easy network manageability and provide reliable network operation as long as frame and slot synchronization is maintained. Moreover, in contrast to token-passing based protocols, if the message slots take error hits the system operation will still continue tp operate smoothly. Continuous frame and slot synchronization provides very robust operation because stations can monitor the communication channel continuously, and if the slots are lost no transmission will be permitted over the channel. Even though collisions are only permitted over the request channel, stable network operation can still be maintained by the centrally located network controller whose functionality may include the determination of the retransmission algorithms and retransmission parameters based on network traffic conditions. As a conclusion, the reservation-based media-access protocols lend themselves to ease of network manageability and provide robust and reliable network operation under all circumstances.

### 4.6 Summary of Comparative Protocol Analysis

The discussions thus far have concentrated on the analysis of various media-access protocols as applied to MANs. We have studied almost every basic technique that can be used for the transmission of data, packetized voice, digitized and compressed video over MANs. As stated in the previous sections, none of these basic media-access techniques can effectively be applied to MANs and meet the requirements set forth for the integrated services encompassing data, voice and video applications. Although some adaptive techniques may overcome the inherent weaknesses of each media-access protocol presented in the previous sections, dramatic performance improvement cannot be achieved. In this section, we will summarize our results and conclusions, and then delineate the main attributes of a media-access protocol that is required to support a complete set of services over MANs.

First of all, we point out that the salient characteristics of a media-access protocol such as throughput, access delay, and stability, are mainly determined by the channel traffic conditions, message length, end-to-end signal propagation delay, number of stations attached to the network, and other network parameters. The objective of the new media-access protocol is to minimize the overhead incurred in the arbitration of channel access while maintaining low access delays, and to utilize the channel bandwidth more efficiently even when packet transmission times are short and the end-to-end signal propagation delay is long.

The key measures of performance for a media-access protocol are the channel throughput and access delay as a function of offered channel traffic. In general, the performance characteristics depend on a multitude of variables including the number of stations attached to the network, message length, offered traffic load, end-to-end signal propagation delay, and the overhead associated with the channel access resolution algorithm. In the previous sections we vigorously studied the performance attributes of several media-access protocols under various loading conditions. The following paragraphs summarize the salient performance characteristics of major media-access protocols applicable to MANs for lightly and heavily loaded systems.

The polling-based media-access protocols perform very poorly under light traffic load conditions, resulting in low channel throughput and long access delays, whereas the channel utilization may reach almost its maximum level under heavy loading conditions. For continuous and periodic traffic the access delay is minimized and bounded as a function of the number of stations attached to the network.

Under lightly loaded systems the token-passing based media-access protocols will exhibit low channel utilization while longer access delays are imposed on the active network stations. Alternatively, for heavily loaded systems the channel throughput is maximized and the access delay is bounded, resulting in guaranteed channel access and controlled delay variance. Furthermore, these protocols are assumed to be deterministic, thus allowing stable operation in the shared channel.

The contention-based media-access protocols basically provide fast channel access, in other words low access delay, under light traffic loading conditions while maintaining high channel throughput. For heavily loaded systems the channel throughput degrades as a function of offered traffic, and in turn the mean access delay and its variance becomes intolerably large. Furthermore, channel stability is also hampered by the traffic backlog caused by excessive collisions, thus possibly rendering the shared channel unstable under certain conditions. Under heavy traffic conditions the channel access cannot be guaranteed since the access delay is not bounded.

The reservation-based media-access protocols provide the best channel utilization because the slot or channel interference among stations is totally eliminated during message transmission. For lightly loaded systems the channel access delay, even though the channel throughput is maintained at its possible maximum levels, is large due to the time taken to establish a reservation slot. On the other hand, for heavily loaded systems generating continuous and/or periodic traffic and for time-constrained applications these protocols provide the best channel utilization while maintaining bounded and controlled channel access delay. Furthermore, the channel throughput levels close to ideal M/D/1 can be obtained for heavily loaded systems, and the channel access is guaranteed under all circumstances.

As discussed earlier, the performance of a media-access protocol is strongly dependent on the traffic model and channel traffic loading. Generally speaking, some channel traffic characteristics do favor one class of protocols over others. In Table 4.2, appropriate media-access protocols are listed for various traffic models including traffic burstiness, message length, and number of users.

It is evident that for each service class to be supported on a MAN, one class of media-access protocols will provide the best network performance over others. Alternatively, single-mode channel access algorithms, allowing only one class of media-access schemes such as contention-based protocols to exist on the shared channel, may not deliver the network performance required for a broad range of integrated services assumed to be offered by a MAN. This means that for certain applications or services one

Table 4.2. Traffic Model vs. Media-Access Protocol

| Traffic Model | Media-Access Protocol |
|---|---|
| Non-Bursty Users | Fixed-Assigned Channels (FDMA and TDMA)<br>Token Passing (Token-Bus, Round-Robin) |
| Bursty Users and Short Messages | Pure Aloha,<br>Slotted-Aloha<br>CSMA |
| Bursty Users, Long Messages, and Large Number of Users | Reservation Protocols With Contention-Reservation Channel (R-TDMA, R-Round-Robin, BRAM, Robert's Scheme) |
| Bursty Users, Long Messages, and Small Number of Users | Reservation Protocols with Fixed TDMA<br>CSMA/CD, CSMA/CA, CSMA<br>Round-Robin with Priority Assignment |
| Various Combinations of Traffic Models | Adaptive Protocols (URN, Adaptive Polling, R-Aloha, R-CSMA) |

class of media-access protocols is preferred over others while another class of protocols is ideal for some services included in the Service Matrix. In order to circumvent this problem, a new media-access scheme, which adapts to changing traffic loading condition, will be required. A multi-mode media-access protocol which allows more than one class of media-access schemes to exist on the shared channel simultaneously or in a time-division fashion will be adequate to support a broad range of integrated services.

Little effort has been devoted to the development and analysis of multi-mode media-access protocols supporting integrated services encompassing data, packetized voice and digitized/compressed video communications. Alternatively, almost all single-mode media-access protocols such as contention, token-passing, and polling, have been studied vigorously and implemented for various applications. The need to provide multiple classes of services over an ubiquitous medium has prompted some research activity in the industry. The UNILINK protocol which encompasses the performance qualities of both CSMA/CD and token-passing access schemes has been proposed and implemented [DAHO 83]. However, the UNILINK protocol, which posses the attributes of a true dynamic TDMA scheme, requires large overhead and high computing power to provide a load-adaptive channel access, thus improving access delay at the expense of channel throughput.

In view of the previous presentations and discussion it is concluded that a load-adaptive, multi-mode media-access protocol is necessary to meet the service requirements of a MAN spanning over a large geographical area. As a conclusion, a media-access protocol intended for a MAN to support integrated services including data, packetized voice, and digitized/compressed video should have the following basic attributes:

- Multi-mode channel access,
- Load adaptability,
- Dynamic nature,
- Configurability based on service requirements,
- Channel speed independence,
- Medium independence.

A new media-access protocol which possesses these attributes will be proposed and described in the next chapter. This new protocol will allow integrated services to be offered over an ubiquitous medium defined in a MAN constructed over a large geographical area.

# CHAPTER V

# NEW DEMAND-ADAPTIVE MEDIA-ACCESS PROTOCOL CHARACTERISTICS

## 5.1 General Overview

The essence of the new proposed media-access protocol is the ability to operate in various channel access modes depending on the user traffic offered, which can range from data and packetized voice to digitized/compressed video, and the property to adapt to traffic load fluctuations in real-time. This protocol employs two basic channel arbitration and allocation techniques simultaneously: 1) a centrally controlled media-access scheme, and 2) a distributed media-access scheme. In the centrally controlled channel access schemes, the channel time is statically assigned to stations with heavy and continuous traffic load. In the distributed channel access technique, a dynamic channel reservation mechanism is employed to assign channel time on a demand basis.

The proposed media-access protocol basically is a combination of several well-known protocols such as reservation, and Slotted CSMA/CD, which are modified to coexist simultaneously in the ubiquitous channel. Furthermore, it possesses the following attributes: 1) the deterministic nature of polling protocols, 2) low delay and high throughput characteristic of contention protocols under light traffic conditions, 3) guaranteed access and low delay variance characteristic of token-passing protocols, and 4) demand adaptivity characteristic of ATDMA protocols. This protocol combines the best attributes of several of the best-known media-access protocols and constitutes a unique multi-mode channel access operation ranging from synchronous and asynchronous TDMA to Slotted CSMA/CD protocol. For example, a dynamic reservation mechanism such as ATDMA is employed for time-constrained applications whereas a static reservation mechanism like STDMA is used for point-to-point continuous information exchanges. Channel reservations and single datagram transmissions are handled through a Slotted CSMA/CD protocol coupled with new retransmission backoff algorithms.

It is assumed that multiple distinct services with broad varying channel demand characteristics will be supported by a MAN. In order to provide data, packetized voice, and digitized/compressed video services over an ubiquitous channel for a large number of users, a new media-access protocol in conjunction with the new communication network topology will be required. In the proposed network topology it is assumed that the medium is comprised of multiple physical channels, referred to as "Segments", and that the transmit and receive channels use different frequencies in the spectrum. For example, in coaxial-based MANs, as shown in Figure 5.1, a transmit channel in the reverse direction is translated to a receive channel in the forward direction, thus creating a virtual bus topology folded in the middle. This type of topology is also used in fiber optics-based MANs as well as radio and satellite networks. The proposed network topology can also be applied to other communication networks including LANs, radio networks, and satellite networks, because it lends itself to all types of media such as fiber optics, coaxial cable, radio, microwave, etc. With its flexibility and applicability to all types of networks and its medium-independence, the proposed network topology provides the means to support a large number of users with applications ranging from data and voice to video communications. The proposed network topology will be described in detail and its relation to the OSI Reference Model will be discussed further in this chapter.

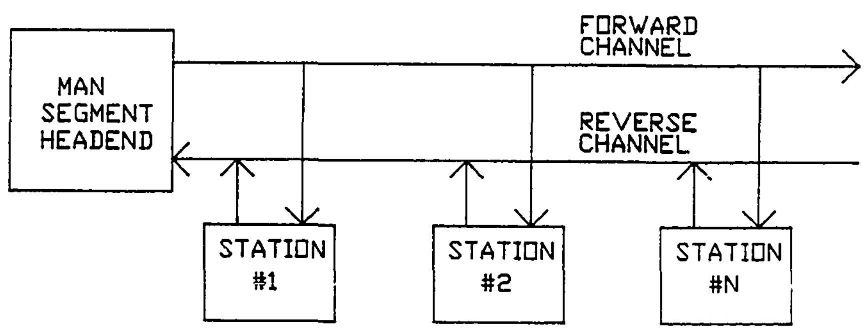



Figure 5.1. Segment Structure in Coaxial-Based MANs

For the purpose of explaining the new network topology and the demand-adaptive media-access protocol, a coaxial-based MAN will be considered. A MAN, as described in this dissertation, consists of one or more connected segments forming the new communication network topology. A MAN Segment, as illustrated in Figure 5.1, consists of a Segment Headend and a plurality of stations which communicate over transmit and receive channels. A station can monitor channel activity in the receive channel, listening to broadcast transmission, and transmit its message in the transmit channel, sharing the channel in multi-access mode.

The proposed media-access protocol operates over a MAN segment to which thousands of stations are attached. The interconnection of MAN segments is accomplished through data-link level bridges, thus allowing high-speed data exchange between segments without incurring additional overhead. The information transfer between bridges is through a backbone network, possibly using token-passing based media-access protocol for high throughput under continuous traffic loads. Connecting MAN Segments through bridges and subsequently interconnecting bridges through a backbone network forms a hierarchical network topology.

The new media-access protocol fundamentally possesses the property of a connection-oriented demand-assignment system built upon and sharing a single communication channel with virtual circuit and static and dynamic reservation facilities, and the property of a connectionless system built upon datagrams. The proposed media-access protocol does not require centralized control for dynamic channel-time reservation and datagram services. Each network station has intelligence to execute the channel access procedures whereas centralized control is needed for static channel-time reservation. The channel can be operated using only the dynamic channel-time reservation mode, if static reservation is not required, since centralized control is not required.

For the purpose of integrating data, voice and video services in a shared channel, the channel is slotted and the network stations are globally synchronized to a time-base reference regardless of their physical locations. The time-slotting and sychronization is made possible by compensating for the round-trip propagation delay of each station. The channel time is partitioned into fixed intervals, called "Frames", and each frame is further subdivided into fixed intervals, called "Subframes". A frame consists of a frame header and a data field. The frame data field may consist of a subframe, called "Super Subframe", or sixteen fixed size subframes. A frame may be partially or totally reserved for a single user. The frames and subframes are numbered so that slot reservations are possible and the network stations can be kept synchronized at all times.

In addition to time-slotting, a priority mechanism is employed to reduce the competition in the channel and assign different access rights to various service classes, i.e., assigning the highest priority to time-constrained applications such as real-time voice communications and the lowest priority to low-duty cycle applications such as electronic mail, and utility reading.

The reservation of frames and subframes can be done statically and/or dynamically, based on the class of services. In this protocol all network stations monitor the receive channel at all times and act upon the information contained in the frame header. If a frame is reserved by a station, then other stations refrain from using that frame. Otherwise, stations compete for the empty frames and subframes depending on the urgency and priority of users' messages. The channel access algorithm will be delineated later in this chapter.

## 5.2 Communication Network Topology and Protocol Architecture

A MAN, as considered in this chapter, consists of multiple network segments to support a large population of users and to integrate data, packetized voice and digitized/compressed video services over a single shared channel. It has been previously demonstrated that the concept of providing a single physical connection for each user to support a combination of data, voice and video services, effectively an information pipeline, is not feasible with the present network topologies and media-access protocols used by LANs, satellite and radio networks, etc. Another major problem with the present system architecture is the interconnectivity of various networks employing different media-access protocols. It has been argued that the delay involved in going through a gateway may not be acceptable for certain time-constrained applications such as real-time voice communications or video conferencing [BRUT 83]. It has been established that providing a media-access protocol for each class of service is technically feasible, but the economic feasibility could be far reaching due to the high cost and difficulty of implementing such a system to support integrated services [BARA 82, BHUS 84, KUO 81].

The essence of an integrated communication network topology is to:

1) Carry multiple distinct services over a single shared channel,
2) Provide a global and transparent connectivity among a large number of users,
3) Offer flexible system architecture for expansion,
4) Provide reliable network operation,
5) Permit low-cost implementation.

Broadband media, coaxial cable and fiber optics, all provide the means to construct such a network topology. The proposed communication network topology addresses the above issues effectively and offers a generic solution for various broadband networks including MANs and LANs as well as satellite and radio networks.

Generally speaking, the network topology determines the applicability of a media-access protocol and vice versa. For example, a token-passing protocol can be used in conjunction with bus, ring and star topologies, whereas a CSMA/CD protocol requires each network station to listen to its own transmission for collision detection, permitting network operation only with a bus topology. The properties of a media-access protocol will be influenced by the limitations imposed by network topologies. Therefore, a proper network topology should be architected so that the new media-access protocol can support integrated services for a large population of users.

To allow connection-oriented and connectionless services to be offered in a shared channel and to ease bridging between MAN segments, a new media-access protocol architecture is required. The proposed protocol architecture addresses the issues related to supporting integrated services and uses a layered architecture similar to the ISO Reference Model. For the purpose of facilitating network management and diagnostic functions a system-level network management layer coupled with layer management is proposed.

### 5.2.1 Hierarchical Network Topology

A single MAN segment will serve a certain number of network stations with integrated services, depending on traffic load and distribution, but multiple MAN segments probably will be required to support a large population of users. It is conceivable that each MAN segment may provide one specific service, and the stations may use the appropriate MAN segment as they wish. For example, a MAN Segment may be allocated for voice communications, another one for conventional data communications and another one for compressed video services. When a station wishes to establish a voice circuit it may use the MAN segment assigned for voice communications. If the station, at a later time, wants to use the channel for data communications it may use the MAN Segment assigned for data services.

It is highly desirable that a MAN Segment provide all integrated services in a single channel. In this case, the total user population is divided into groups, and such a group is assigned to a unique MAN Segment since a single MAN Segment may not be able to support all MAN users at once.

The need for use of multiple MAN Segments to support a large population of users with integrated services necessitates a mechanism for interconnecting MAN Segments in a transparent fashion. In order to provide interconnectivity between MAN Segments in real-time, Segment Bridges are employed since all MAN Segments use the same media-access protocol. Message routing between MAN Segments is implemented at the Data Link layer, and the process is transparent to users. Thus, the sending station does not have to know in which MAN Segment the receiving station is located. Each MAN Segment has a unique address which is used by a station as an address extention to its unique individual address. When a user selects another MAN Segment it also acquires the selected MAN Segment address which becomes a part of the station address.

In a MAN environment a physical channel constitutes a MAN Segment. When a station chooses a physical channel the corresponding channel number becomes the MAN Segment address. In coaxial-based MANs, a MAN Segment is physically represented by a pair of transmit and receive channels (see Figure 5.1). A MAN Segment consists of a headend and a plurality of network stations transmitting and receiving information on the designated channels. As shown in

Figure 5.2, a MAN Segment Headend is comprised of a Demodulator (radio-frequency receiver), a Modulator (radio-frequency transmitter), and a Segment Data Processor. The Segment Data Processor is connected to the Segment Bridge Interface to interchange information between MAN Segments, and optionally to the Network Segment Monitor.

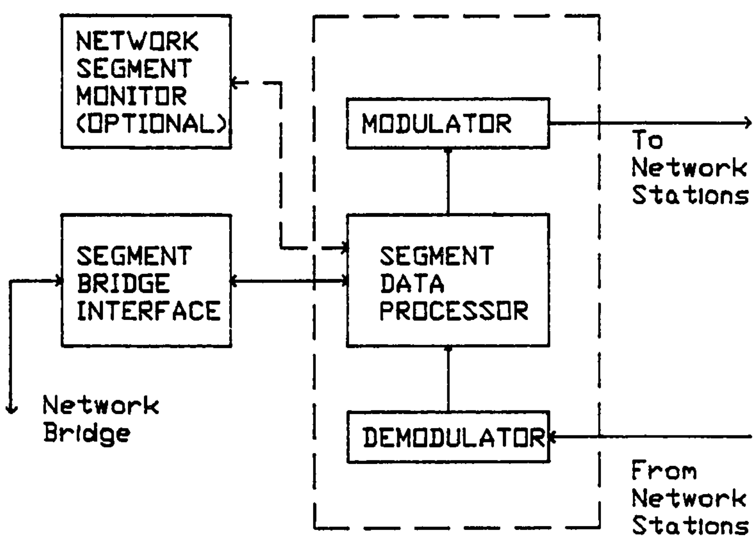



Figure 5.2. MAN Segment Headend

The function of the Demodulator is to receive signals from network stations in the radio-frequency spectrum and convert them into digital form suitable for TTL, etc. The function of the Modulator is to transmit signals to network stations in the radio-frequency spectrum by modulating the digital input signal. The RF modulation and demodulation processes allow signals to travel the long distances inherent in MANs. The functions of the Segment Data Processor are:

1) Generation of frame headers,
2) Frame synchronization,
3) Message buffering,
4) Message deletion and insertion,
5) Message routing,
6) Message flow control to and from the Segment Bridge Interface and Network Segment Monitor.

In summary, the basic function of the MAN Segment Headend is to regulate the information flow between network stations and MAN Segment Bridge.

The interconnection between MAN Segments is provided by the Network Bridge whose basic function is to route messages between MAN Segments. In order to switch messages in real-time, a number of MAN Segments are clustered and connected to the Network Bridge. The Network Bridge Architecture is shown in Figure 5.3. In a fairly large MAN a single bridge will not have the processing power and speed to route information between MAN Segments in real-time. Therefore, as depicted in Figure 5.4, the Network Bridges serving clustered MAN Segments are interconnected via a backbone network. As illustrated in Figure 5.4, a Network Bridge can handle a certain number of MAN Segments which is determined by the signalling speed of the backbone network. The Network Bridges are interfaced to the backbone network via the Interbridge Network Access Controller. In order to allow all

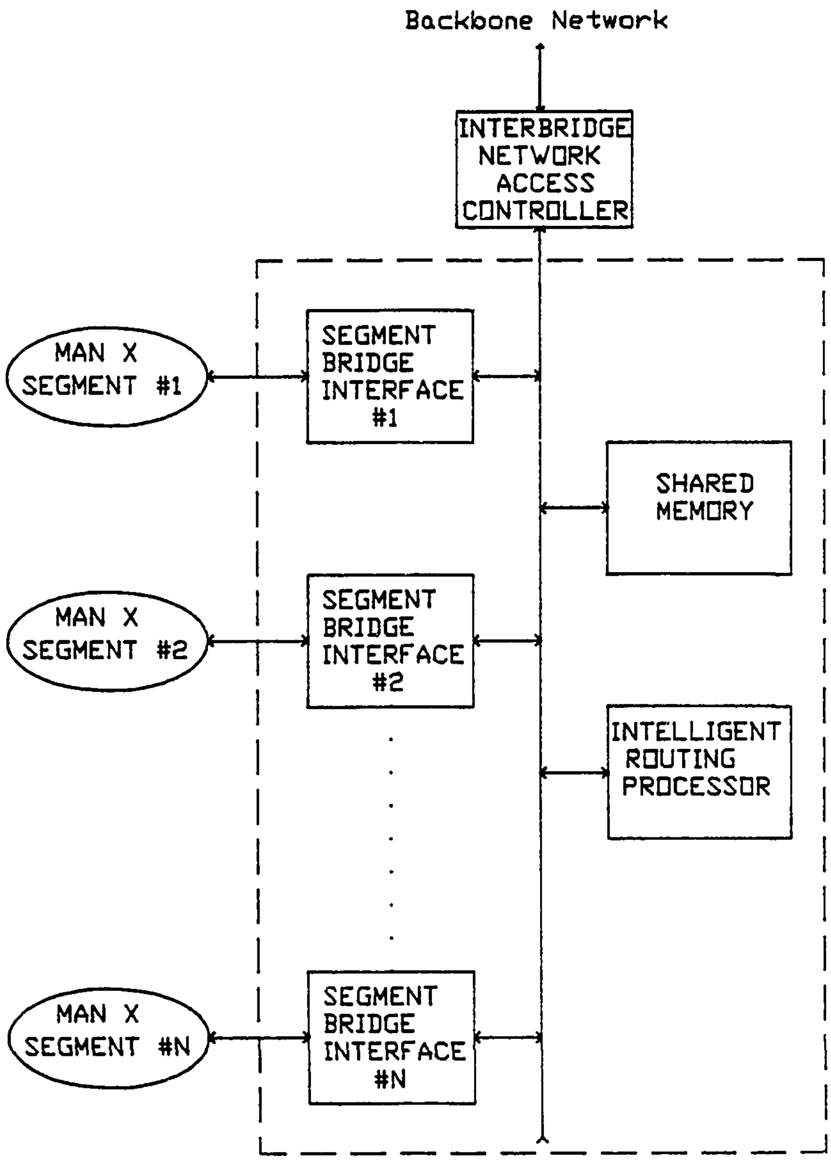



Figure 5.3. Network Bridge Architecture

MAN stations to access dissimilar networks, gateways are attached to the backbone network (see Figure 5.4). If the traffic flow from a MAN Segment to a gateway which is required to access dissimilar networks and is connected to the backbone network is heavy, then a gateway must be provided in that segment to prevent traffic congestion on the backbone network. The signalling speed of the backbone network should be sufficient to allow message switching between Network Bridges at maximum speeds without requiring excessive buffering. A hierarchical network topology, in which messages are routed in real-time, is formed by clustering the MAN Segments and interconnecting them via a Network Bridge, and interconnecting the Network Bridges via a backbone network.

If a message generated in a MAN Segment is intended for a station located in another MAN Segment served by the same Network Bridge, it is intercepted by the corresponding Segment Data Processor and sent to the Network Bridge via the Segment Bridge Interface. The Network Bridge routes this message according to its segment address and sends it to the destination MAN Segment. The destination Segment Data Processor inserts the message into the channel and sends it to the destination address in the first empty frame or subframe, depending on the message size.

If the destination address of the transmitted message is not located in the same Network Bridge group, it is routed to the Interbridge Network Access Controller for transmission to the proper Network Bridge located in the backbone network. This routing between MAN Segments and Network Bridges is done in real-time with the use of segment

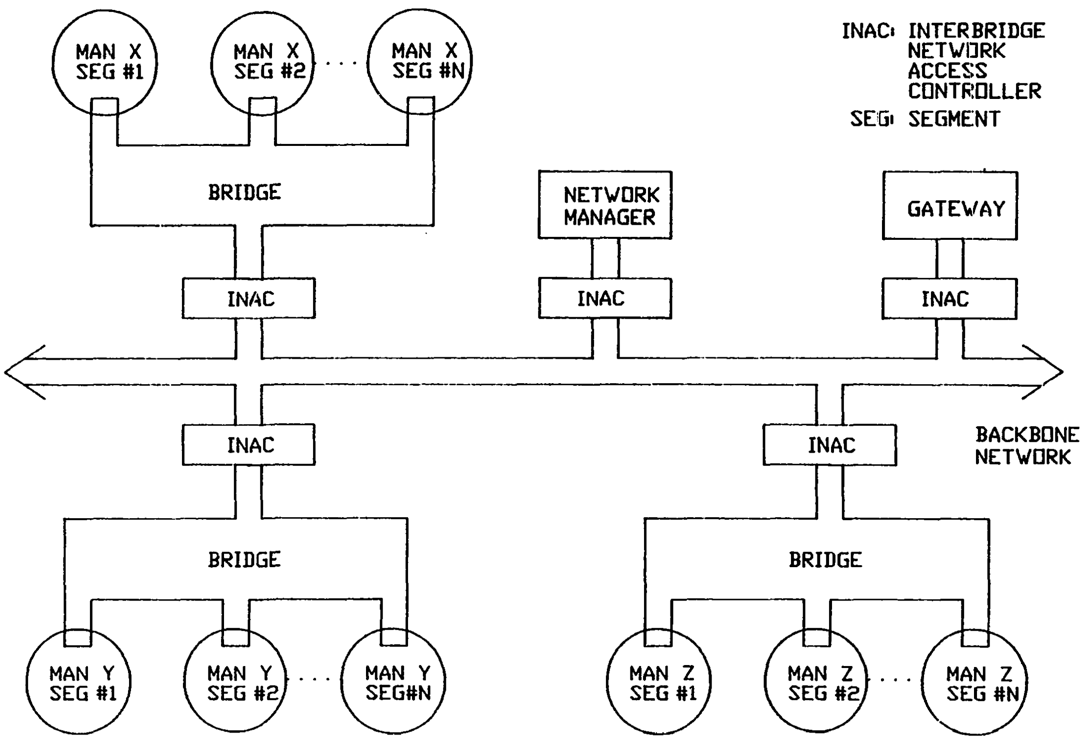



Figure 5.4. Modular Structure of Hierarchical Network Topology

addressing. The overhead associated with routing is minimized since the same media-access protocol is employed across all MAN Segments.

The hierarchical network topology, as described here, allows a large number of users scattered over a large geographical area to be interconnected without incurring additional access delay. This real-time interconnection process is made possible by dividing network stations into groups and by assigning them MAN Segments, each operating on a pair of transmit and receive channels in which the same media-access protocol is employed. The hierarchical network topology, as depicted in Figure 5.5, is created by use of intelligent Network Bridges and a backbone network. With this type of topology, network stations can be assigned to any segment, independent of their physical locations. For example, the station $S_a$ located in Segment #1 of the MAN X group can be physically next to the Station $S_m$ located in Segment #2 of the MAN Y (see Figure 5.4). The hierarchical network topology also allows the network to be tailored for specific applications. For example, refer to Figure 5.5, Segment #1 of the MAN X can be used exclusively for voice communications, whereas Segment #2 of the MAN X can be dedicated for conventional data communications. Assignment of a network station to a MAN Segment is accomplished by setting the station address field and by tuning to the corresponding transmit and receive channels.

The network management and diagnostic functions can either be performed by a centrally located Network Manager

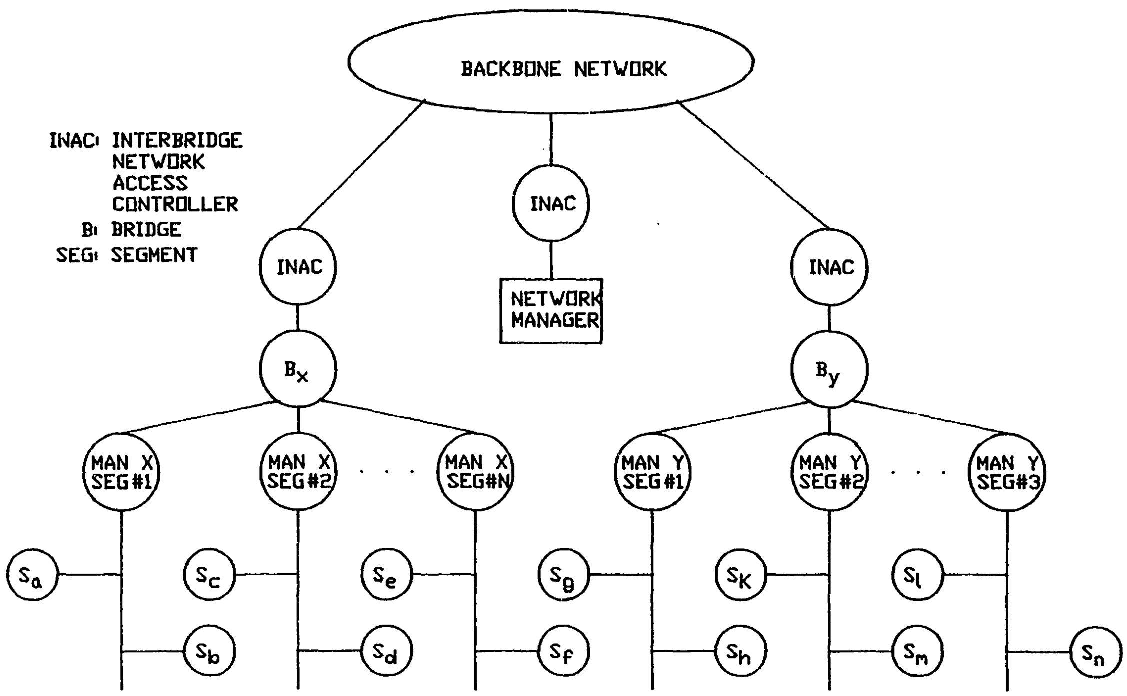



Figure 5.5. Hierarchical Network Topology

connected to the backbone network, (see Figure 5.4), thus eliminating the need for providing each MAN Segment with its Network Manager, or by the Network Segment Monitor located at the MAN Segment Headend (see Figure 5.2). As it is displayed in Figure 5.4, the Network Manager and Gateway(s) are attached to the backbone network and shared by MAN Segments X, Y, and Z. A MAN Segment can operate without a Network Manager or Monitor. However, network management and diagnostic functions are crucial in large broadband networks such as MANs. Specifically, a failure in any MAN Segment can be bypassed with use of the centrally located Network Manager by switching all affected network stations in the faulty MAN Segment to an alternate MAN Segment.

In summary, the hierarchical network topology as proposed in this chapter, in conjunction with MAN Segments, Network Bridges, and a backbone network, provides the means for reliable and economical communication network support for a large number of users scattered across a large geographical area for a multitude of integrated services ranging from data and packetized voice to digitized/compressed video. The proposed media-access protocol is based upon the hierarchical network topology. The flexibility of location-independent hierarchical network architecture will allow the proposed media-access protocol to be applied for other types of networks such as fiber-optics based MANs, satellite and radio networks, LANs, etc.

### 5.2.2 Comparison of OSI Reference Model vs. Proposed Protocol Architecture

A continuum of functions is involved in a communication network. Therefore, a logical order needs to be established for better understanding of peer-to-peer communications between stations. The technique of layering has been employed in the Open System Interconnection (OSI) Reference Model to group functions that are most logically related to each other in supporting the communications between two stations.

The OSI Reference Model is commonly used and referenced by the IEEE 802 Standards Organization for LAN and MAN standards [TANN 81, P802 82, P803 85, P804 82]. However, the majority of existing LANs, global networks, and satellite and radio networks do not adhere to this model. Rather each network uses a unique layered architecture resembling the OSI Reference Model. The purpose of the seven-layer OSI Reference Model is to provide a first step toward the international standarization of various protocols encompassing media-access and high-level protocols. This layered architecture has been adopted by all standard organizations such as CCITT, IEEE 802, ECMA, and PROWAY, and is serving its purpose. The basic idea behind layered architecture is to reduce network design complexity. In the OSI Reference Model, the (n) layer receives services from the (n-1) layer and provides specific services to the (n+1) layer, thus shielding those layers from the implementation details of the services offered. In this chapter, we will define the functions of each layer, but will not get into the details, rather we recommend to read the following references to the reader: [TANN 81, KUO 81, ROSN 82].

The proposed protocol architecture adheres closely to the OSI Reference Model in terms of layering and the functions performed by each layer. As it is depicted in Figure 5.6, each layer of the OSI Reference Model has an equivalent layer in the Proposed Protocol Architecture including the Network Manager layer encompassing all seven layers. However, some layers are not required for certain services and they are treated as null layers. In the OSI Reference Model, the Protocol Data Unit (PDU) is defined as the total information that is transferred between peer entities as a unit, whereas the Service Data Unit (SDU) is defined as the total unit of information transferred across the service access point.

The Physical Layer of the Proposed Protocol Architecture is concerned with transmitting and receiving raw bits over a pair of communication channels forming a MAN Segment. (The basic function of this layer is to ensure that when a station sends a "1" bit, it is received by the other station as a "1" bit, not as a "0" bit.) Other functions of the Physical Layer are:

1) Determination of the number of volts to be used to represent a "1" and a "0",
2) Definition of the modulation technique such as Frequency Shift Keying (FSK), etc.,

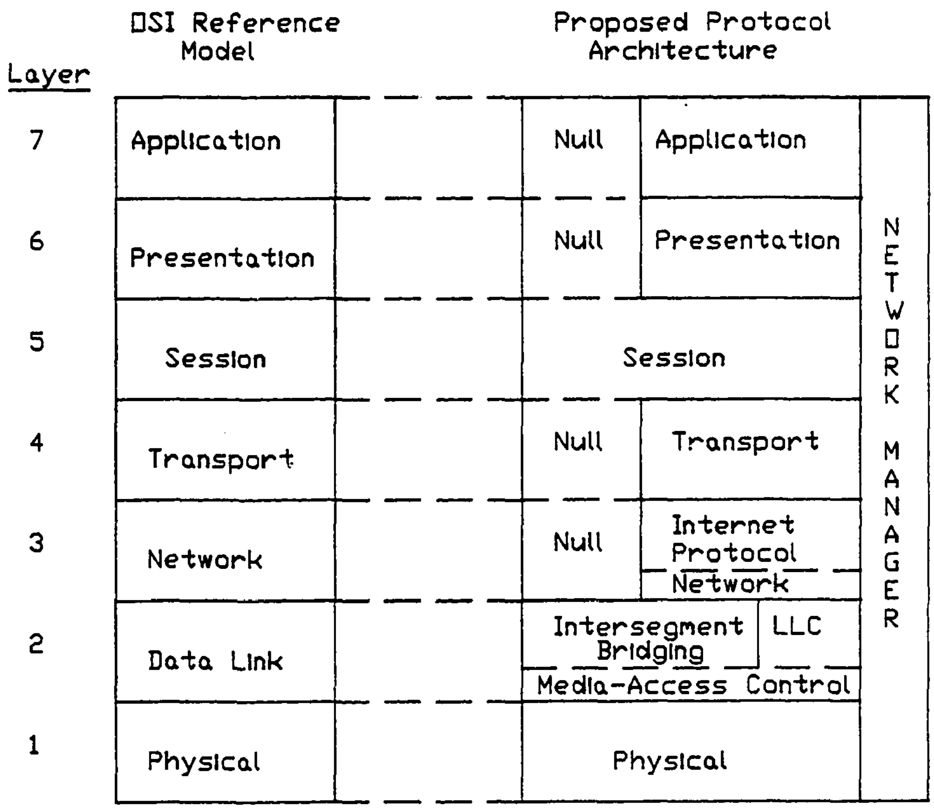



Figure 5.6. OSI Reference Model vs. Proposed Protocol Architecture

3) Definition of the signal strength at the receiver and transmitter,

4) Definition of the network interface connector configuration,

5) Specification of half and full duplex operation,

6) Control of the frequency agility for multiple channels.

In summary, the Physical Layer provides the functional and procedural characteristics to activate, maintain, and deactivate the physical links that pass the bit stream from one place to another in a manner transparent to higher layers.

The Data Link Layer of the Proposed Protocol Architecture is concerned with the media-access mechanism and intersegment bridging (see Figure 5.6). The basic function of this layer is to take a raw transmission facility and transform it into a line that appears free of transmission errors to the network layer it serves. This is accomplished by breaking up the Service Data Unit (SDU) passed by the network layer into the Protocol Data Units (PDUs), transmitting PDUs sequentially, and processing the acknowledgment SDU sent back by the receiving station. The functions of the Data Link Layer are to:

1) Create and recognize frame boundaries,
2) Handle the problem of duplicate packets,
3) Sequence frames,
4) Execute the media-access procedures,
5) Control flow,
6) Detect errors,
7) Bridge between segments.

The proposed protocol will provide the following services at the Data Link Layer: 1) datagrams, 2) virtual circuits via datagrams, 3) static slot reservation, and 4) dynamic slot reservation. As depicted in Figure 5.7, these services can be directly accessed by the session and other upper layers without going through network and transport layers, thus allowing a wide range of services to be supported by the integrated communication network without introducing extra overhead. For example, for voice communications, the network and transport layers can be bypassed and a connection can be made by using the session layer requesting a static or dynamic slot reservation directly from the data link layer. The compression and decompression of digitized voice, if employed, is performed at the presentation layer. It is obvious that not all layers are required in voice communications. Therefore, the Proposed Protocol Architecture provides the flexibility to bypass certain layers, if they are not required, while maintaining a layered structure.

Using a hierarchical network topology coupled with network segmentation allows similar networks to be interconnected via data-link layer bridges with little overhead. The salient advantages of providing bridging capability at the data link layer are: 1) low overhead, 2) real-time packet routing, and 3) flexibility for expansion. As illustrated in Figure 5.6, the Intersegment Bridging is a part of the Data Link Layer and sits on top of the Media-Access Control and serves the network layer.

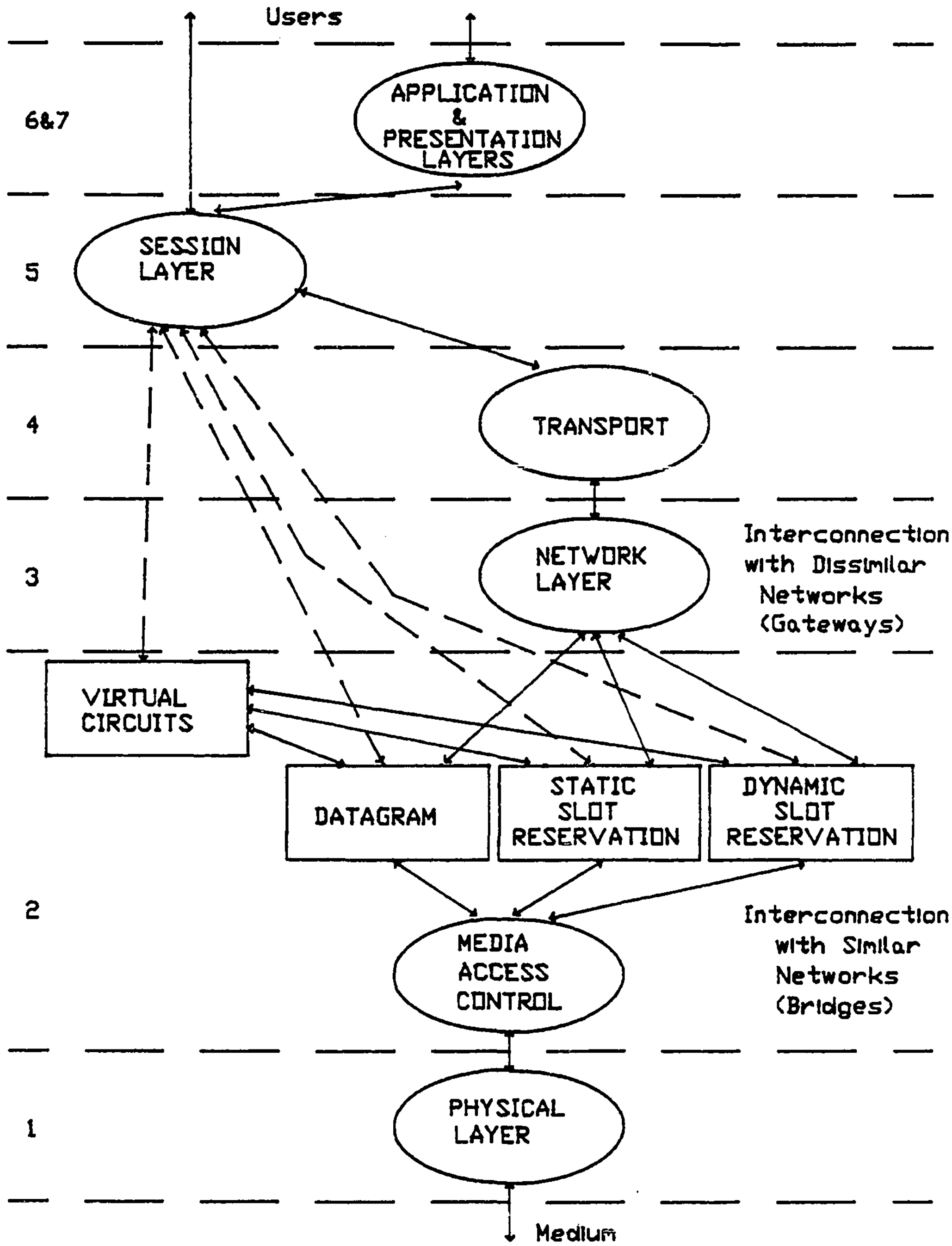



Figure 5.7. Expanded Definition of the Proposed Protocol Architecture

In summary, the Data Link Layer in the Proposed Protocol Architecture provides the functionality of transferring information over the physical link with the necessary synchronization, error control, and flow control functions as well as intersegment bridging.

The Network Layer in the Proposed Protocol Architecture is mainly concerned with receiving the SDUs from the transport layer and converting them into the PDUs and then directing them toward their destination. This process is also known as packet assembly and disassembly. Furthermore, this layer is responsible for message routing and determines whether the message belongs to the network itself or has to be routed to another network. Therefore, the internetworking between dissimilar networks, as shown in Figure 5.7, is handled in this layer. Other functions include message buffering, congestion control, and mapping routing tables. In the Proposed Protocol Architecture, the Network Layer can also be null, simply allowing the layers above to bypass it for certain applications, as shown Figure 5.6. Since ISO has already adopted a standard for the network layer, we propose that this standard be used for the Network Layer with the provision of null network layer capability. In summary, the Network Layer in the Proposed Protocol Architecture provides the switching and routing functions needed to establish, maintain, and terminate switched connections and transfer data between the communicating end systems. The network layer can be bypassed for certain applications where its services are not required.

The basic functions of the Transport Layer in the Proposed Protocol Architecture are to accept the SDU from the session layer, disassemble it into smaller units (PDUs), if need be, pass these to the network layer, and ensure that the PDUs all arrive correctly at the destination station. Furthermore, this should be done in a way that isolates the session layer from changes in the computer hardware. The transport layer is the first layer in OSI Reference Model to provide true end-to-end or source-to-destination communications. The other functions of the transport layer are to:

1) Make connection multiplexing transparent to the session layer,
2) Provide a varying degree of services such as error-free virtual point-to-point, non-guaranteed transmissions.
3) Establish and terminate connections,
4) Provide naming facility,
5) Control data flow.

The corresponding transport layer in the Proposed Protocol Architecture can either be null or provide the basic functions mentioned above (see Figure 5.6). A null transport layer capability is provided to allow this layer to be bypassed for certain applications, i.e., voice communications, point-to-point permanent connections, etc. There exists an ISO Standard for the transport layer, thus we recommend that this standard with Class 4 capability (an error-free virtual circuit) be used in the Proposed Protocol Architecture with the provision of null layer functionality. In summary, the transport layer provides end-to-end control and information interchange with varying degrees of reliability and quality of services that are needed by the layers above as well as null layer functionality.

The Session Layer interfaces to the user on the network. The basic function of this layer is to negotiate for establishing a connection with a process or another machine or a station. The functions of the session layer are to:

1) Establish a connection,
2) Terminate a connection,
3) Convert a session address to its transport address,
4) Request for a transport layer connection set-up,
5) Manage a session,
6) Sequence and order messages.

The corresponding session layer in the Proposed Protocol architecture treats all these functions. As illustrated in Figure 5.6, the session layer is always required to support the integrated services. For example, in voice communications a user can directly interface to the session layer by dialing its destination address to establish a connection, thus bypassing application and presentation layers (see Figure 5.7). When a request is received from a user to establish a connection with another user, it is first processed at the session layer and passed to the data link layer directly without going through the network and transport layers. A standard for the session layer has been drafted by the ISO Standards body, therefore, we recommend that the session layer in the Proposed Protocol Architecture adhere to this standard when it is approved. In summary, the session layer supports the dialog between users by providing the functionality of establishing, maintaining and terminating sessions.

The Presentation Layer is concerned with data transformation, message compression, encryption, etc. This layer attempts to alleviate the problem of exchanging incompatible data between machines, users, etc. The corresponding presentation layer in the Proposed Protocol Architecture could be either null or perform the functions mentioned above, refer to Figure 5.6. For example, in digitized/compressed video and compressed voice communications the presentation layer performs the compression and decompression functions. In other applications, the presentation layer may not be needed if data transformation is not required. The ISO Standards body is currently working on a draft proposal for the presentation layer. Therefore, we recommend that the presentation layer of the Proposed Protocol Architecture conform to the standard, when it is finalized, with the provision of null layer functionality. Briefly, the presentation layer provides the services that allow users, implicitly application processes, to interpret the meaning of the information being transferred, and perfrom syntax selection, conversion and encryption between themselves.

The Application Layer treats the issues directly related to the individual users. It provides network transparency, hiding the physical distribution of resources from the user. In the Proposed Protocol Architecture, the application layer could be either null or provide the functions mentioned above, as illustrated in Figure 5.6. The ISO Standards body is also working on a draft proposal for the application layer. We recommend that the application layer of the Proposed Protocol Architecture conform to the standard, when finalized, with the provision of null layer capability. In summary, the application layer could be null for certain applications or directly serve users by providing access to the communication network and provide the distributed information services to support the users and manage the communication link.

In addition to the seven protocol layers described here, an additional layer, called the Network Manager Layer, is proposed to handle the problems of network management and diagnostics. The Network Manager Layer can interface with each layer directly on one-to-one basis and coordinates the communications between layers for network management and diagnostic functions (see Figure 5.6). For example, in case of a faulty MAN Segment, the Network Manager, for the purpose of providing reliable network operation, exchanges data between layers to inform all stations located in the faulty MAN Segment to switch over to an alternate MAN Segment. The Network Manager Layer provides a system level network control as well as layer-level management functions. Because of the importance of network management functions in MANs, even in LANs, it is recommended that the Network Manager Layer be included in the Proposed Protocol Architecture to ensure reliable network operation.

### 5.3 Frame Structure

For the purpose of explaining the frame structure we will first describe the MAN Segment Headend. Each MAN Segment uses a pair of communication channels, one in the forward direction and one in the reverse direction. A virtual bus topology is created by use of the MAN Segment Headend. The Headend contains a Demodulator, a Segment Data Processor, and a Modulator (see Figure 5.2). This type of headend is referred to as a "Remodulator Headend". The signals in the reverse direction are received by the Demodulator and converted into baseband signals, namely TTL levels. The demodulated signals are passed to the Segment Data Processor whose functions include generation of frame headers, synchronization, message insertion/deletion, message routing, flow control, etc. The information processed by the Segment Data processor is passed to the Modulator for transmission in the RF spectrum to all network stations in broadcast fashion. The processing delay at the MAN Segment Headend will be considered as a part of the system delay and be treated later in the chapter.

The channel time is partitioned into fixed size intervals, called "Frames". As shown in Figure 5.8, a frame consists of a "Frame Header" and a "Data Field". The Frame Header contains a Preamble Field and Guard Band, a Start Delimiter, a Control Field, a Frame Number Field and a Frame Check Sequence Field. The Data Field contains the information to be transferred. To allow shorter packet transmissions on the channel the data field can be subdivided into sixteen fixed size intervals, called "Subframes". As it is shown in Figure 5.8, there are two types of frames: 1) Non-Slotted, and 2) Slotted. The data field of a non-slotted frame is referred to as a "Super Subframe" which represents the largest packet size in the channel. The data field of a slotted frame contains sixteen fixed size subframes each of which represents the smallest packet size. The slotted and non-slotted frames can be reserved statically and dynamically. A non-slotted frame permits more information to be transmitted without requiring message fragmentation. In high-speed point-to-point communications where the information transfer is continuous, non-slotted frames may be reserved at network initialization. For example, a multiplexed voice channel using a T1 carrier can be accommodated by simply reserving a frame statically during network configuration. When interconnecting LANs there may be a requirement of heavy and continuous information transfers, therefore, a non-slotted frame should be used. Alternatively, the slotted-frames are used for reservation requests, datagrams, etc.

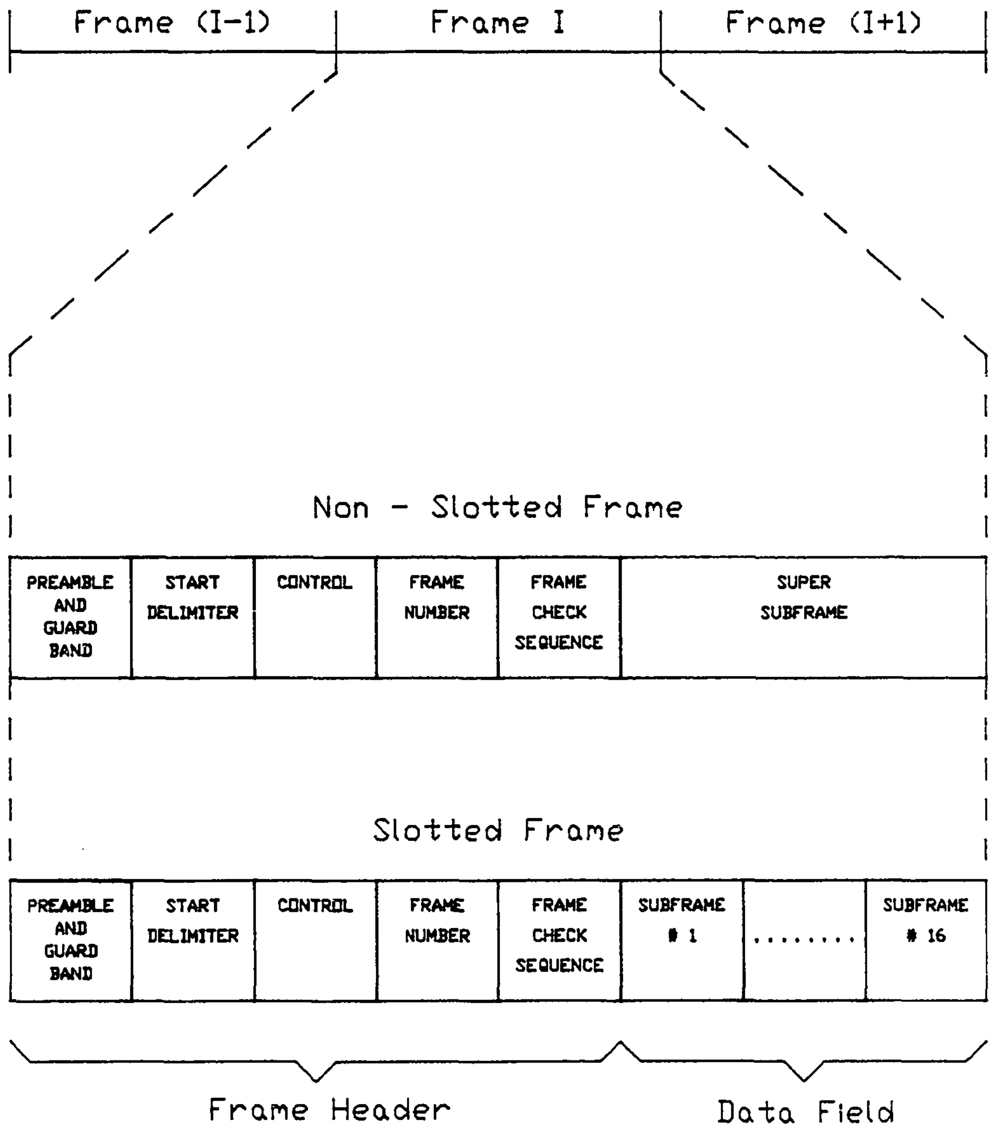



Figure 5.8. Frame Structure

To maintain slot synchronization and global time-base reference in the channel at all times the frames are numbered sequentially ranging from 0 up to 255. This provides a mechanism to partition the channel time and to allow the channel time to be reserved in slot units. The frame headers are generated in the forward channel by the Segment Data Processor. In the reverse channel, the header is used to provide stations with global time-base reference, to provide channel time-slotting for multi-access operation, and to define the characteristics of the corresponding frame. The network stations monitor the frame headers and they act upon the information contained in the header. If a frame header is not present in the forward channel, stations will stay idle until such time as frame headers are generated and channel synchronization is recovered. Therefore, a redundant headend may be required to provide highly reliable network operation.

5.3.1 **Preamble and Guard Band Fields.** The preamble in the frame structure is required to allow data receivers to phase-lock to incoming signals, consequently permitting a clock recovery circuit to generate proper timing for data recovery. In packet-switched broadband networks and baseband LANs, the front-end data receivers require a certain amount of time to lock onto the incoming signals before retrieving the data [P803 85, P804 82]. This time is dependent on the response time of the receiver circuitry. However, an eight-bit preamble, as shown in Figure 5.9, should be sufficient for retrieving the frame headers since they are generated at periodic intervals. Similar to Ethernet and the IEEE 802.3 CSMA/CD Standard, the bit pattern of the preamble is selected to be alternating ones and zeros [P803 85]. The frame header also provides the timing base to further partition the data field of the slotted frame into sixteen fixed size subframes.

The period from the end of a transmitted frame to the beginning of a new frame preamble is referred to as the "Guard Band". It is intended to compensate for time discrepancies among stations caused by varying signal propagation delay, tolerances of electronic circuitry, etc. The length of the guard band is determined by the data rate, area of network coverage, and the type of electronic components being used.

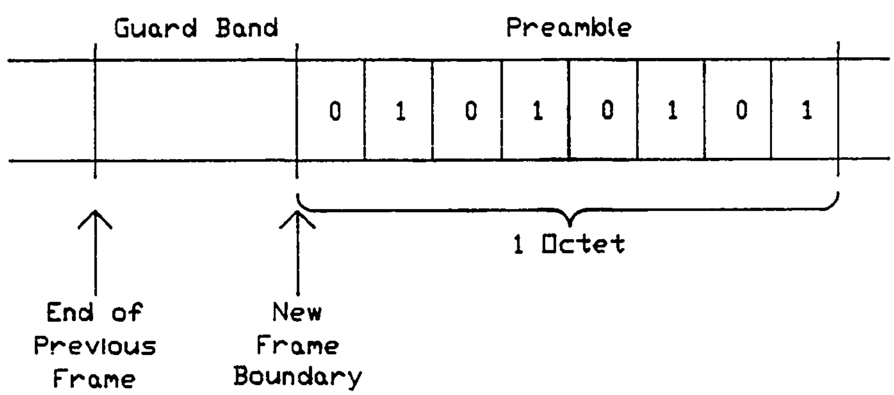



Figure 5.9. Preamble and Guard Band Fields of a Frame

**5.3.2 Start Delimiter Field.** The function of the Start Delimiter is to mark the beginning of a new frame header, allowing stations to synchronize to the frame timing. The length of this field, as shown in Figure 5.10, is one octet and its content is selected to be a unique synchronization character similar to the "01111110" flag in HDLC frame. The uniqueness of this character should be tied to the preamble to reduce the probability of false detection of a frame header. Reliable detection of a frame header will enhance data and clock recovery, thus reducing the bit error rate. If the frame header is not recognized then that frame slot would be lost, therefore, it is crucial that a unique synchronization character be selected in conjunction with the preamble.

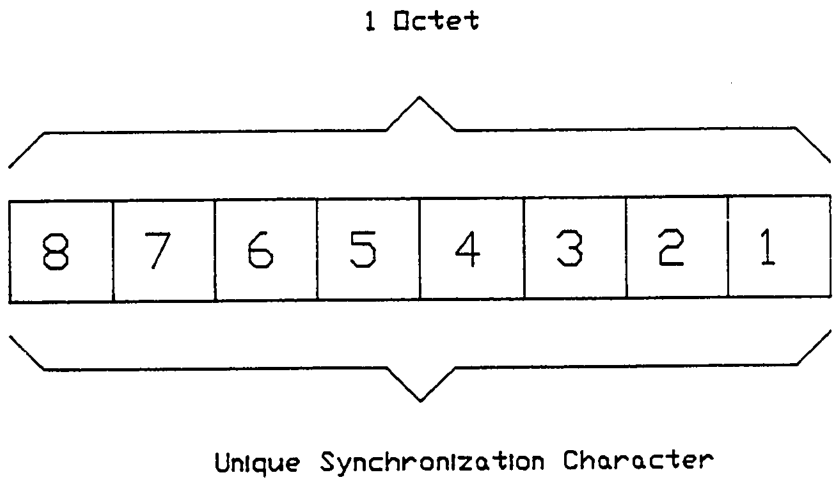



Figure 5.10. Start Delimiter Field of a Frame

**5.3.3 Control Field.** The control field of a frame defines the interpretation of the data field, priority levels, network management messages, and frame and network status (see Figure 5.11). The length of this field is one octet and its contents pertain only to its associated frame. In other words, each frame can individually be defined and controlled through the information contained in its control field. The control fields are defined as follows:

1) The Frame Service Priority Level field determines the priority level of the corresponding frame. Eight priority levels are defined for each frame, thus allowing each frame to be defined for various prioritized services.

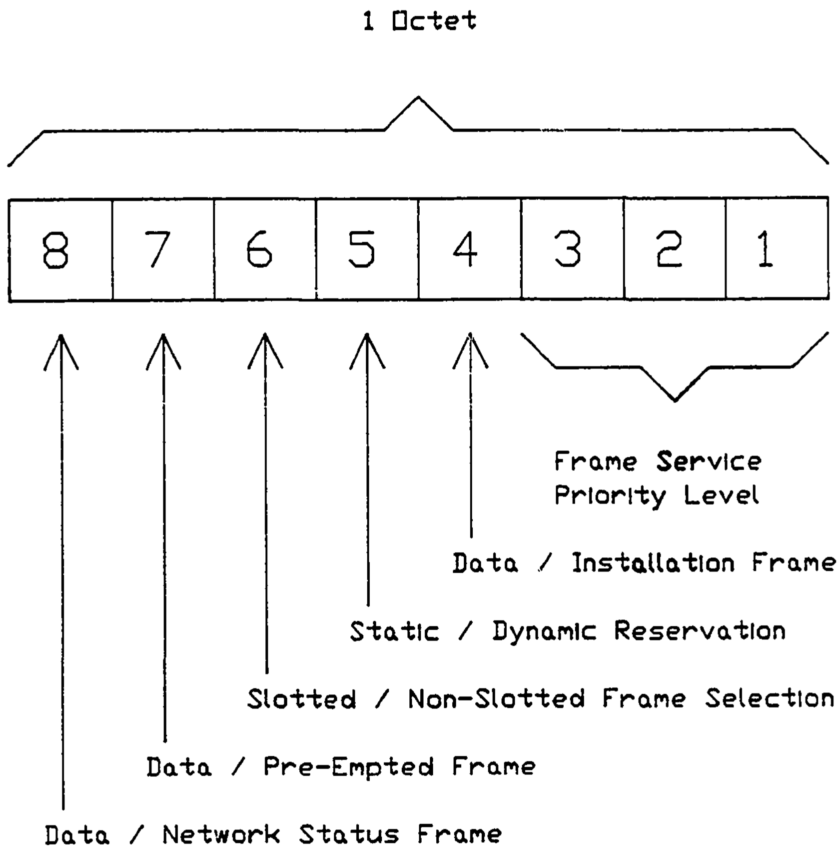



Figure 5.11. Control Field of a Frame

The priority assignment and differentiation will be discussed later in this chapter.

2) The Data/Installation Frame field indicates whether the frame is available for transmission of data packets or is used as an installation frame.

3) The Static/Dynamic Reservation field determines whether the frame is statically or dynamically reserved.

4) The Slotted/Non-Slotted Frame Selection field determines whether or not the data field of the corresponding frame is further partitioned into to

subframes. If the frame is not partitioned then it is a super subframe with a packet length of sixteen subframes.

5) The Data/Pre-Empted Frame field determines whether the frame contains packets generated by the stations located in the corresponding MAN Segment or is pre-empted to transmit data in that Segment from other MAN Segments. In this pre-emption process, all stations which are supposed to be transmitting will refrain from transmission for one frame cycle, and after one cycle the normal operation will resume.

6) The Data/Network Status Frame field determines whether the frame is available for data packet transmission or reserved for network management and diagnostic messages. The frame control fields and all possible conditions are summarized in Table 5.1.

5.3.4 **Frame Number Field.** The Frame Number Field of a frame contains the frame number corresponding to the frame being transmitted. The length of this field is one octet and the maximum number of frames that can be numbered is 256 (see Figure 5.12). When the maximum frame number is reached the frame numbers are recycled sequentially. Another function of this field is to provide frame and subsequently subframe synchronization. If a frame is lost due to errors caused by channel noise, the frame timing and subframe synchronization can be re-established at the next frame cycle. Frame numbering provides the means to operate the channel in slotted contention and TDMA protocols simultaneously.

Table 5.1. State Diagram of a Frame Control Field

| Control Field Value | | | | | | | | Description |
|---|---|---|---|---|---|---|---|---|
| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
| - | - | - | - | - | X | X | X | Service Priority Levels |
| - | - | - | - | 0 | - | - | - | Data Frame |
| - | - | - | - | 1 | - | - | - | Installation Frame |
| - | - | - | 0 | - | - | - | - | Static Frame Reservation |
| - | - | - | 1 | - | - | - | - | Dynamic Frame Reservation |
| - | - | 0 | - | - | - | - | - | Non-Slotted Frame |
| - | - | 1 | - | - | - | - | - | Slotted Frame |
| - | 0 | - | - | - | - | - | - | Data Frame Available |
| - | 1 | - | - | - | - | - | - | Pre-Empted Frame |
| 0 | - | - | - | - | - | - | - | Data Frame Available |
| 1 | - | - | - | - | - | - | - | Network Status Frame |

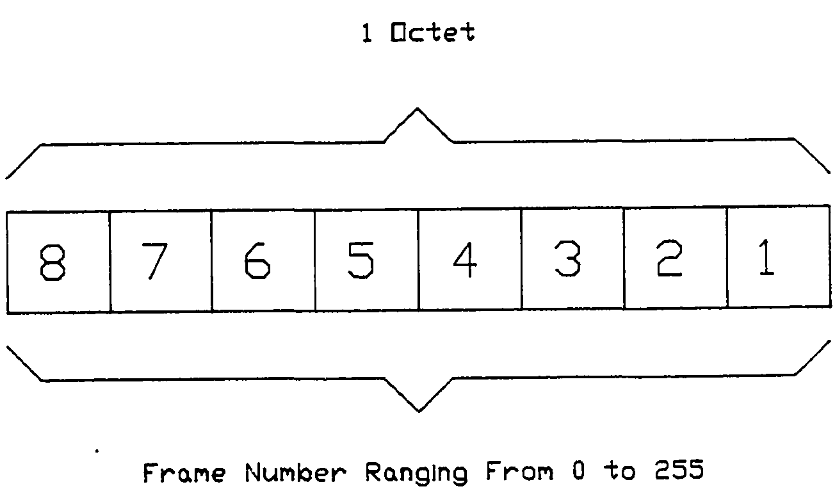
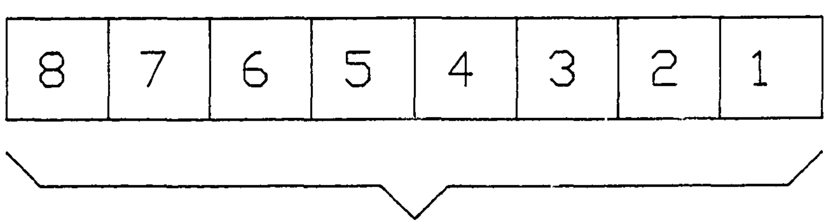



Figure 5.12. Frame Number Field of a Frame

**5.3.5 Frame Check Sequence Field.** The Frame Check Sequence Field provides an error protection mechanism for the frame header. The length of this field is one octet, an 8-bit Cyclic Redundancy Check code ($x^8+x^7+x^4+x^3+x+1$) is used to detect errors in the frame header (see Figure 5.13) [SIRA 84]. The Hamming distance of this CRC is 4, therefore, three bit errors in a frame header can be detected. The corruption of a frame header is detected by the computation of the frame check sequence. This field is generated by the Segment Data Processor and inspected by all network stations. The necessity of this field stems from the need to provide reliable channel slotting and synchronization so that no data packets are mistakenly transmitted in the reserved frames. If there is an error in the frame header, detected through the frame check sequence field, then stations will not transmit in the data field of that affected frame, but will wait for the next frame cycle. If frame header is received intact, network stations will act according to the information contained in the frame header.




8-Bit Cyclic Redundancy Code

CRC: $X^8 + X^7 + X^4 + X^3 + X + 1$

Figure 5.13. Frame Check Sequence Field of a Frame

## 5.4 Subframe Structure

Two types of data packet entities are defined in the data field of a frame: 1) super subframe, the largest packet entity, and 2) subframe, the smallest packet entity, by further subdividing the data field of the slotted frame into fixed size intervals. The packet length of the super subframe is fixed and its structure is the same as a subframe with the exception of a longer data field. Alternatively, a subframe is logically recognized by its location in the data field of a slotted frame. The length of a subframe is fixed and can be as long as that of sixteen subframes. Longer subframes can be constructed by using contiguous subframes, but in integer multiples only. The flexibility of having a fixed and variable size packet length provides a mechanism for multiplexing a broad range of services over a single shared channel. The frame headers are transmitted by the Segment Data Processor whereas the subframes and super subframes are generated by network stations.

The implicit address of a subframe is determined by the frame number plus its location in the data field of a slotted frame, i.e., frame number 126 and subframe slot number 5. The structure of a subframe is illustrated in Figure 5.14. A subframe contains a Start Delimiter, a Control Field, Destination and Source Address Fields, a Data Field, a Subframe Check Sequence Field, and a Guard Band Field. It should be noted that subframes and super subframes preserve the same structure in forward and reverse channels. The subframe structure allows the reservation and contention protocols to be operated in a shared channel.

| START DELIMITER | CONTROL | DESTINATION ADDRESS | SOURCE ADDRESS | DATA | SUBFRAME CHECK SEQUENCE | GUARD BAND |
|---|---|---|---|---|---|---|
| 1 | 1 | 3 | 3 | N | 2 | < 1 |

Numbers Indicate Field Length In Octets

Figure 5.14. Subframe and Super Subframe Structure

**5.4.1 Start Delimiter Field.** The Start Delimiter is used to mark the beginning of a subframe, consequently allowing stations to maintain synchronization to the global time-base reference at each subframe slot. The length of this field, as shown in Figure 5.15, is one octet and its content is defined to be a unique character different than the frame start delimiter, thus frames and subframes can easily be discerned. This field is mainly used to mark the packet start and separate the consecutive packets from each other since the clock timing and global time-base referencing is provided by the frame header. False packet start detection will not occur because the location and exact timing of a subframe is supplied by the corresponding frame header. In the proposed protocol, the slot activity status is determined by listening to the channel at the beginning of each slot, namely testing for carrier in the collision window corresponding to the start delimiter.

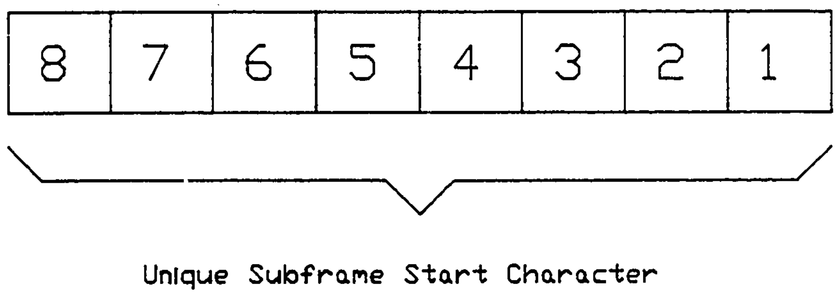



Figure 5.15. Start Delimiter of a Subframe

**5.4.2 Control Field.** The Control Field of a subframe defines the length of a subframe, the interpretation of information content, the packet type, and the acknowledgment process. The length of this field is one octet and its contents pertain only to its associated subframe (see Figure 5.16). In other words, each subframe can uniquely be defined and controlled by using its control fields. The Number of Subframes field specifies the number of consecutive subframes. This field allows variable length packets, integer multiples of subframes, to be transmitted in contiguous subframe slots. Thus, a station may dynamically reserve up to sixteen subframe slots in a frame period without additional overhead. However, the number of slots reserved should be consecutive and packets should not spill over slot boundaries. Network stations are permitted to transmit packets as small as a subframe or as large as a super subframe by setting the Number of Subframe field accordingly.

The Packet Type field, as shown in Figure 5.16, uses three bits of the contrcl field and determines the type of data packet that is being transmitted. Eight packet types are defined by use of this field. The packet types are summarized in Table 5.2. The last bit of the control field is used as an Acknowledgement Request field. Its function, when set to "1", is to ask the receiving station to acknowledge the packet being transmitted. If this field is set to "0" packets are not acknowledged. The control field structure will permit the implementation of various types and classes of services defined by IEEE 802.2 Logical Link Control standard document [P802 82].

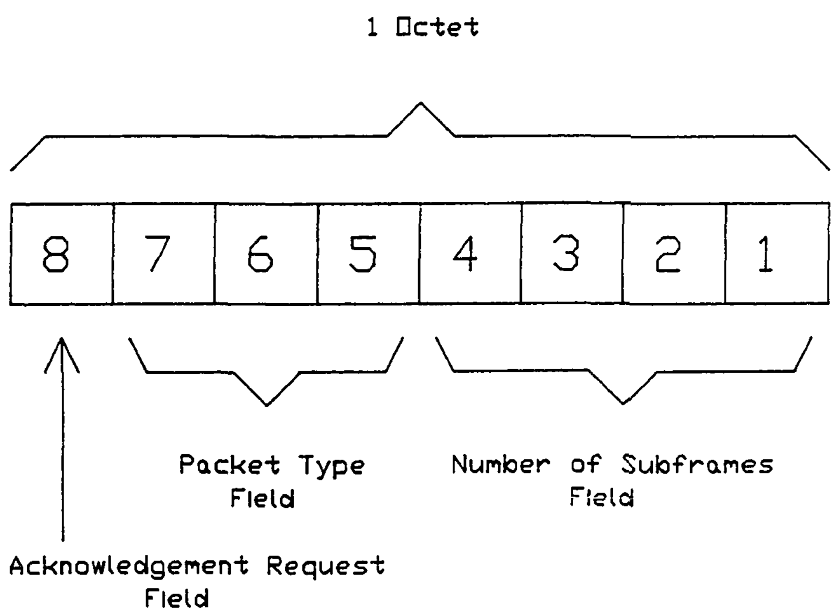



Figure 5.16. Control Field of a Subframe

Table 5.2. Subframe Packet Types

| Value | Packet Type |
|---|---|
| 0 0 0 | Installation |
| 0 0 1 | Network/Station Status |
| 0 1 0 | Acknowledgment |
| 0 1 1 | Reservation |
| 1 0 0 | Datagram |
| 1 0 1 | Statically-Reserved |
| 1 1 0 | Dynamically-Reserved |
| 1 1 1 | Reserved |

**5.4.3 Source and Destination Address Fields.** The Source and Destination Address Fields are intended to uniquely identify the address of a station transmitting a subframe (source address), and the address of a station to which a subframe is being sent (destination address). The format of source and destination address fields is the same and each address consists of three octets (see Figure 5.17). The address field contains a 15-bit unique Individual Station address, a 1-bit Individual/Group address, a 7-bit Segment address, and a 1-bit Global/Local address. The addressing and channel segmentation will be discussed later in this chapter.

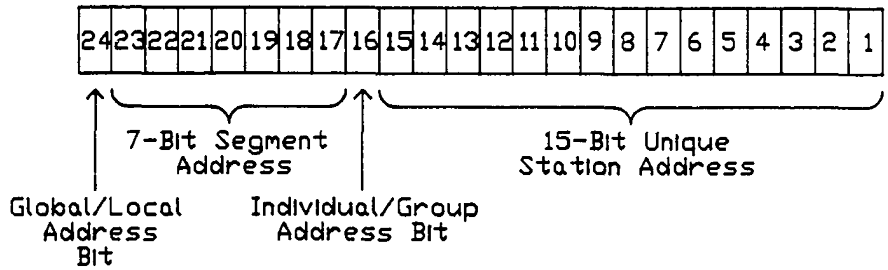



Figure 5.17. Source and Destination Address Fields of a Subframe

5.4.4 **Data Field.** The Data Field of a subframe carries the information being exchanged between stations. As shown in Figure 5.18, this field may contain a number of data symbols, which may assume any possible value. Bit transparency is provided and the length of the data field is determined by the subframe control field, and is interpreted by the destination station through the higher level of protocol layers such as the network and transport layers. In a single subframe the length of the data field is fixed. Alternatively, it may vary for variable length packets depending on the number of consecutive slots being occupied. However, it may not exceed the data field length of a super subframe which occupies sixteen subframe slots.

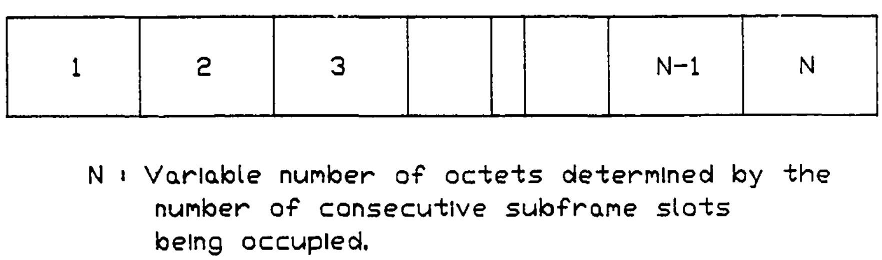
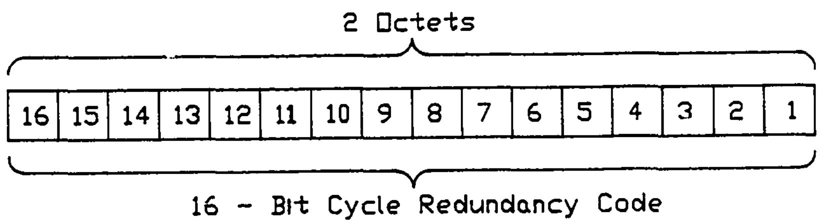



Figure 5.18. Data Field of a Subframe

**5.4.5 Subframe Check Sequence Field.** The corruption of data in a subframe is detected by the computation of the Subframe Check Sequence. As shown in Figure 5.19, the length of this field is two octets, and a 16-bit Cyclic Redundancy Check code, i.e., CRC-16 or CRC-CCITT, is proposed. The subframe check sequence is generated by the transmitting station and inspected by the transmitting and receiving stations to detect errors caused by collisions or channel noise in the slot.




CRC: CRC-16 or CRC-CCITT

CCITT: $X^{16}+X^{12}+X^{5}+1$

CRC-16: $X^{16}+X^{15}+X^{2}+1$

Figure 5.19. Subframe Check Sequence Field of a Subframe

**5.4.6 Guard Band.** The Guard Band Field of a subframe is defined as the period between the Frame Check Sequence Field of a subframe being transmitted and the Start Delimiter field of the succeeding subframe. This field is used to compensate for propagation delay variances, tolerances of electronic components, etc. As shown in Figure 5.20, the length of the Guard Band Field is less than one octet, and no padding is required.

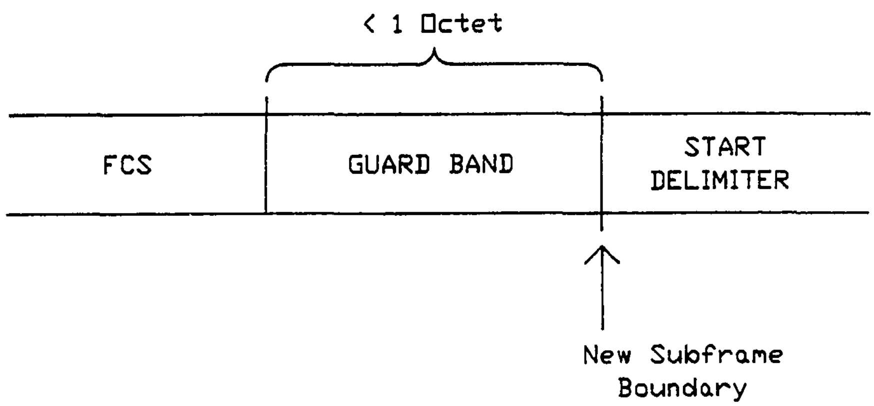



Figure 5.20. Guard Band Field of a Subframe

### 5.5 Packet Formats

The information entity transmitted in a subframe or in a super subframe is also referred to as a "Packet". As discussed earlier eight packet types are defined by use of three bits of the subframe control field. The following packet types are supported by the proposed media-access protocol:

- Installation packet
- Network/Station Status Packet
- Acknowledgment Packet
- Reservation Packet
- Datagram Packet
- Statically-Reserved Packet
- Dynamically-Reserved Packet
- Reserved

With the defined packet types the slotted contention-based and reservation protocols can easily be implemented.

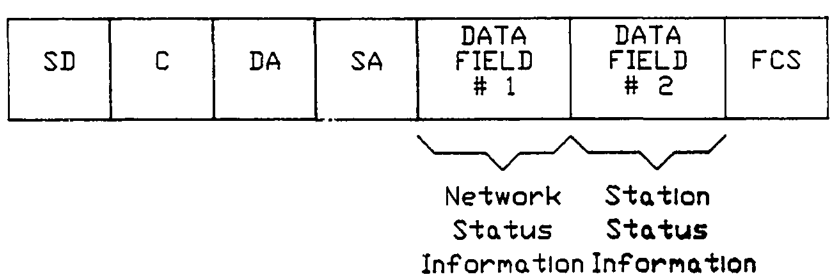
Each packet type and its functionality will be discussed in the following paragraphs.

**5.5.1 Installation Packet.** The function of an Installation Packet is to allow new network stations to adjust their latency time before they are permitted to share the channel. The latency adjustment is defined as the delay to be added at each network station to make the round-trip delay equal to the frame slot period. When a new station is attached to the network, it first transmits a self-addressed installation packet in the frame reserved for station installation, then it listens to its packet and adjusts its latency time according to an algorithm to be described later in this chapter. As illustrated in Figure 5.21, the installation packet is a self-addressed subframe and its data field is null. In a MAN, the signal propagation delay is inherently large and may take values from a couple of microseconds to a couple of hundred microseconds. Therefore, the installation packet is defined to be as short as possible to prevent spilling over to an adjacent slot.

| | | | | |
|---|---|---|---|---|
| SD | C | SA | SA | FCS |

Figure 5.21. Installation Packet Format

**5.5.2 Network/Station Status Packet.** The function of a Network/Station Status Packet is to allow the Network Manager or Monitor and network stations to exchange information for services such as network management, network initialization and restart, network crash recovery, network and station diagnostics, station authentication and verifications, etc. As illustrated in Figure 5.22, this packet occupies a frame slot, as a super subframe, and its data field consists of two portions: one for network status information and the other for station status information. When the Network Manager or Monitor wishes to talk to a station it sends a Network/Station Status packet with the desired station address in the destination address field and the unique Network Manager's or Monitor's address in the source address field plus any relevant network and station status information in the data field of this packet. Any network station may, in the same manner, send messages to the Network Manager.




DA: Network Manager or Station Address
SA: Station or Network Manager Address

Figure 5.22. Network/Station Status Packet Format

**5.5.3 Acknowledgment Packet.** The function of an Acknowledgment Packet is to ensure that information interchange between stations is accomplished free of errors by either sending the received packet or simply an acknowledgment message back to the sending station. As shown in Figure 5.23, the received packet is sent back to the sending station by swapping the source and destination addresses. Alternatively, a short acknowledgment packet may be sent back to the sending station by setting the Acknowledgment Request bit in the control field to "0" and by swapping the source and destination addresses. With the availability of the Acknowledgment Packet a virtually error-free transmission can be guaranteed and all IEEE 802.2 LLC service types and classes can be supported.

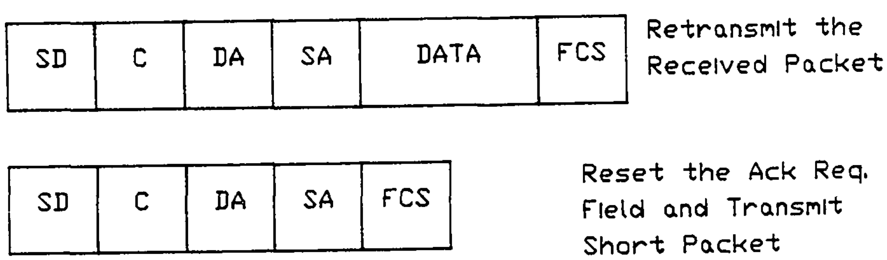



Figure 5.23. Acknowledgment Packet Formats

**5.5.4 Reservation Packet.** Multi-Priority Reservation Packets allow stations with different channel capacity demands to effectively access the channel depending on their service priority. As illustrated in Figure 5.24, there are eight types of Reservation Packets each with different packet length depending on the service priority level. The length of the Reservation Packet, as shown in Figure 5.24, is determined by the service priority level; the shortest packet is for the lowest priority level and the longest packet for the highest priority level. The purpose of defining different length reservation packets is to provide an arbitration mechanism to differentiate the service request priorities so that higher priority stations can be given the first channel access rights over lower priority stations. The shortest Reservation Packet, the lowest priority, contains no data field whereas the next priority level, Level 1, contains eight octets of null symbols in the data field and the highest priority packet carries 56 octets of null symbols in the data field. All types of Reservation Packets are self-addressed. The priority resolution among the Reservation Packets will be discussed later in this chapter.

**5.5.5 Datagram Packet.** The Datagram Packet may range from a single subframe up to a super subframe consisting of sixteen subframes. The Datagram Packet is used to allow stations to exchange information among each other. Datagram Packets are also referred to as "Addressed Information Entities". Virtual circuits can also be supported by use of datagrams. As illustrated in Figure 5.25, a Datagram Packet may contain a fixed number of octets in a subframe slot. However, a variable length Datagram Packet may also occupy a multiple of consecutive subframe slots. Each packet contains a destination and a source address.

Priority Level 0

| | | | | |
|---|---|---|---|---|
| SD | C | SA | SA | FCS |

Priority Level 1

| | | | | | |
|---|---|---|---|---|---|
| SD | C | SA | SA | 8 - OCTET NULL DATA | FCS |

Priority Level 7

| | | | | | |
|---|---|---|---|---|---|
| SD | C | SA | SA | 56 - OCTET NULL DATA | FCS |

Figure 5.24. Reservation Packet Formats

Single Subframe:

| | | | | | |
|---|---|---|---|---|---|
| SD | C | DA | SA | DATA FIELD | FCS |

Fixed Number of Octets

Multiple Subframes:

| | | | | | |
|---|---|---|---|---|---|
| SD | C | DA | SA | DATA FIELD | FCS |

Variable Number of Octets

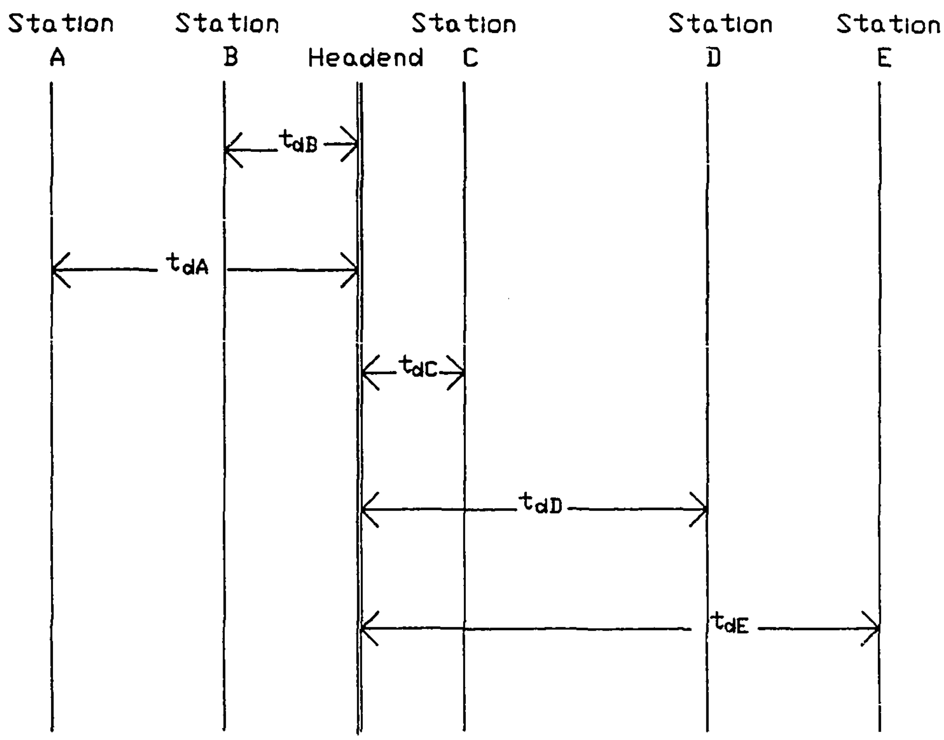
Figure 5.25. Datagram, Statically-Reserved and Dynamically-Reserved Packet Formats

**5.5.6 Statically-Reserved Packet.** The Statically-Reserved Packet has the same attributes as a Datagram Packet. This type of packet is used when the frame or subframe slot is statically reserved. The frame field in the associated frame header signifies the type of packet(s) being transmitted in the succeeding frame data field. What distinguishes the Datagram Packet from the Statically-Reserved Packet is the information carried in the subframe control field. The Statically-Reserved Packets may take several forms ranging from a single subframe up to a super subframe occupying sixteen subframe slots. As illustrated in Figure 5.25, a Statically-Reserved Packet may contain a fixed number of octets in a subframe slot. Similar to a Datagram Packet a variable length Statically-Reserved Packet may occupy a multiple of consecutive subframe slots. The destination and source addresses are optional once the connection is establihed between two or more stations.

**5.5.7 Dynamically-Reserved Packet.** The Dynamically-Reserved Packet has the same attributes as a Datagram Packet and its size may range from a single subframe up to a super subframe consisting of sixteen subframes. A super subframe is the largest packet in a frame slot. The Dynamically-Reserved Packet is used when the channel time is dynamically reserved. As shown in Figure 5.25, a Dynamically-Reserved Packet may contain a fixed number of octets in a subframe slot. However, a variable length Dynamically-Reserved Packet is also permitted to occupy a number of consecutive subframe slots. Each packet contains a destination and a source address.

### 5.6 Propagation Delay Compensation

In a typical MAN stations are scattered across a large geographical area, up to 30 miles in radius, and signals are received by network stations at different times due to varying signal propagation delay. The propagation delay difference between the station closest to the headend and the station farthest from the headend could be as high as a couple of hundred microseconds. As illustrated in Figure 5.26, network stations are virtually located at different distances from the headend. The signals generated by a station are first sent to the headend and then retransmitted back to all stations on the forward channel, hence the traveled distance is twice the cable length from the headend. In Figure 5.26 Station E is the farthest with one-way signal propagation delay of $t_{dE}$, and Station C is the closest one with one-way signal propagation delay of $t_{dC}$. The maximum signal propagation delay is denoted as $t_{dmax}$, equal to $t_{dE}$. Signals transmitted by Station E will be heard by the same station after twice the station's one-way propagation delay, $2*t_{dE}$, plus the time spent processing the information at the headend. On the other hand, these signals will be received by Station C much earlier than Station E, simply after $t_{dE} + t_{dC}$ time. In similar fashion the signals transmitted by Station C will be received by the same station before Stations A, B, D, and E receive the signal. It is obvious that at any instant in time the channel status is seen different by stations in different states, busy by some and idle by others.

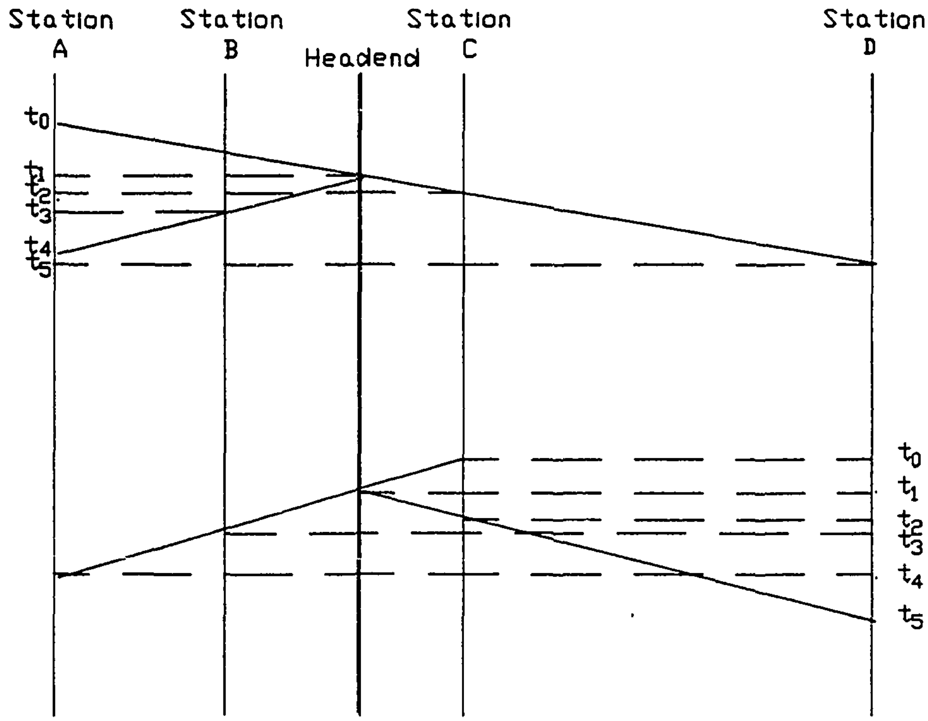



C: Closest Station to the Headend
E: Farthest Station from the Headend
$t_{dx}$: One-way Propagation Delay For Station X
$t_{dmax}=t_E$ : Maximum One-way Propagation Delay

Figure 5.26. Propagation Delay Definition

In coaxial-based networks the signal propagation delay is measured to be in the range of 6.5 to 7.5 microseconds per mile depending on the cable grade [COOP 83]. Such a wide variation in propagation delay creates the problem of synchronization, global time-base reference, slotting, and improper collision detection. In order to provide synchronization and proper time-slotting, a global time-base reference to which all stations can synchronize their clocks and timings must be established across the network. Furthermore, slotted media-access protocols require that time-slots be seen by all stations virtually at the same time regardless of their distance to the headend. This means that current channel status should appear the same to all stations to ensure fairness in the channel and provide proper carrier sensing and time-slotting. If the channel status is current with all stations and a global time-base reference is established, then slotted and contention-based protocols can effectively employ carrier sensing and collision detection techniques to shared the channel.

As illustrated in Figure 5.27, when Station A transmits signals at time $t_0$, traveling in the reverse direction, no signals will be heard in the forward channel until such time the signals arrive at the headend at time $t_1$ and then are retransmitted in the forward channel after they are processed. Station C will be the first to receive the signals at time $t_2$, and consequently Station B at $t_3$, Station A at time $t_4$ and finally Station D at time $t_5$. This example clearly shows that signals will be received by the stations in a wide time-window from $t_2$ to $t_5$. Another example is also shown in Figure 5.27 when Station C transmits signals.




$t_x$: Packet Transmission and Reception Time at Different Stations

Figure 5.27. Propagation Delay Effect on Synchronization

In order to provide a global time-base reference and allow proper time-slotting all network stations should be logically relocated so that they will all be virtually at the same distance from the headend. Since stations cannot physically be moved the alternative is to logically relocate stations next to the farthest station in the network. In other words, the signal propagation delay is compensated for each station so that signals are received by all network stations at the same time determined by the maximum value of the round-trip signal propagation delay. In order to compensate for this delay the following steps are taken: 1) the maximum round-trip signal propagation delay is downloaded to all network stations (this value depends on the network size), 2) the signal propagation delay is measured by each network station, 3) the compensation factor is calculated by subtracting the station's round-trip signal propagation delay from the maximum round-trip signal propagation delay, and 4) the received packets are delayed by the amount calculated previously. Note that the compensation factor for the farthest station is virtually zero.

In the proposed media-access protocol the channel time is divided into fixed size frames, therefore, the time delay between the forward and reverse channels seen by all network stations is required to be exacly one frame slot. Consequently, the frame slot should be equal or greater than the maximum round-trip signal propagation delay plus the headend processing time. The frame cycle time should be less than the maximum tolerable delay for voice communications, ranging from 1 to 4 msec. Denote the frame slot as $t_f$, the headend processing time as $t_h$, and the propagation delay compensation time as $t_c$.

$$t_f >= 2 * t_{dmax} + t_h$$

As shown in Figure 5.27, Station E is the farthest one from the headend and Station C is the closest one. Since the headend processing delay is the same for all network stations there is no need to include it in the calculation of the propagation delay compensation factor. It is also assumed that the signal propagation delay can be measured by all network stations with a tolerance level which can be compensated for by the Guard Band of a frame. By referencing Figure 5.27, the propagation compensation factor is calculated for Stations A, B, C, D, and E in the following way:

Station A: $t_c(A) = t_f - 2 * t_{dA}$

Station B: $t_c(B) = t_f - 2 * t_{dB}$

Station C: $t_c(C) = t_f - 2 * t_{dC}$

Station D: $t_c(D) = t_f - 2 * t_{dD}$

Station E: $t_c(E) = t_f - 2 * t_{dE}$

For example, assuming a signal propagation delay of 7.5 microseconds per mile and a frame slot time of 250 microseconds, let Stations A, B, C, D, and E be at a distance of 10, 4, 3, 12, and 15 miles from the headend respectively. It should also be noted that the frame slot time will at least be equal or greater than the round-trip signal propagation delay. The latency adjustment for Stations A, B, C, D, and E is calculated as follows:

Station A: $t_c(A)$ = 100 microseconds

Station B: $t_c(B)$ = 190 microseconds

Station C: $t_c(C)$ = 205 microseconds

Station D: $t_c(D)$ = 70 microseconds

Station E: $t_c(E)$ = 25 microseconds

In the proposed media-access protocol the propagation compensation process is performed in the following manner:

1) When a station is newly attached to the network it first sends an Installation Packet in the frames reserved for this purpose.

2) The station resets a counter which is used to measure the elapsed time between transmission and reception of an installation packet,

3) The station listens for its packet.

4) When the station receives its self-addressed installation packet it stops the counter and stores its content.

5) Once the round-trip signal propagation delay is measured the station sends a station status message to the Network Manager or Monitor to request the frame size or the maximum round-trip signal propagation delay and other related network status information.

6) The station then calculates its propagation delay compensation factor (latency adjustment).

As it is illustrated in Figure 5.28, the propagation delay compensation factor (latency adjustment value) is loaded into the Programmable Delay Circuitry residing on the receive path of a station. Note that there is no delay element on the transmitter path.

When all network stations execute the propagation delay compensation procedure, the shared channel becomes totally synchronized across the network and the channel time can readily be divided into slots. Moving all network stations logically to the same location allows the channel time to be slotted and the channel status to be current and the same for all stations.

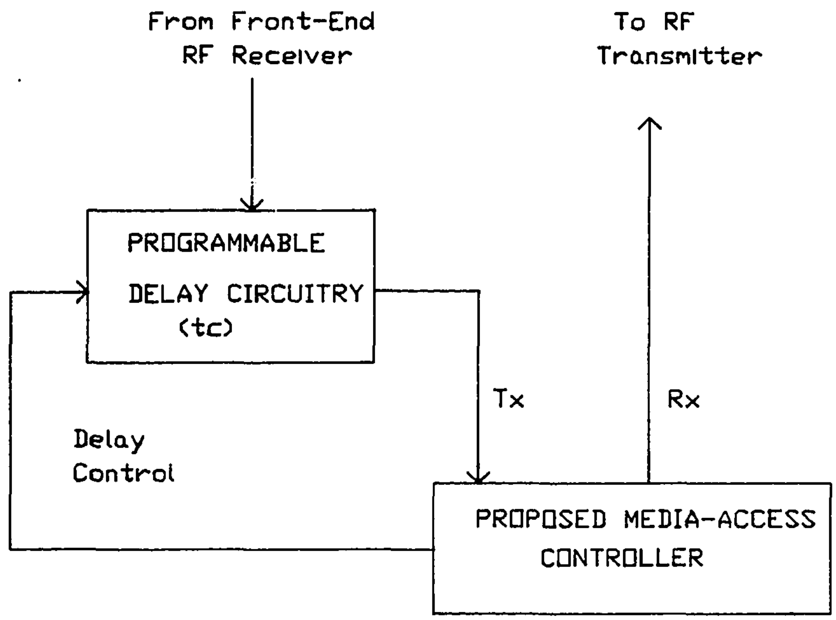



Figure 5.28. Implementation of Latency Adjustment

## 5.7 Frame Synchronization and Subframe Slotting

In the proposed demand-adaptive media-access protocol the channel time is partitioned into fixed size frames and the frames are further subdivided into fixed size subframes. For the purpose of synchronizing frames and subsequently subframes a global time-base reference must be established across the network to guarantee that transmissions from different stations do not overlap. The natural place to generate this global time-base reference is the centrally located Segment Headend. As discussed earlier, the Segment Headend generates the frame headers periodically by numbering them sequentially MOD $N_X$, where $N_X$ is the number of frames in a cycle, with a maximum value of 256. The frame headers are generated by the Segment Headend only in the forward direction whereas subframes are generated by the network stations in the reverse direction and retransmitted back in the forward direction after they are processed at the headend. A centrally located global time-base reference generator provides an excellent frame and subframe sychronization mechanism. Consequently, further subdivision of frames also permits fairly accurate subframe time slotting. If the frame header recognized by a station indicates that its data field is to be used in a slotted form then the data field of the received frame is logically subdivided into sixteen subframe slots by each one of the network stations. This global time-base referencing process is quite accurate because the timing and synchronization information is available in each frame header. Briefly, the subframe slotting is logically performed by all network stations simultaneously. By using the signal propagation delay compensation process all stations will see the same frame and subframe slots virtually at the same time and will maintain slot synchronization as long as the frame headers are received intact by all network stations. If a frame header is not received intact (checking the frame header check sequence) the stations will refrain from using the corrupted frame slot.

All network stations will execute the signal propagation delay compensation procedure so that they can monitor the channel virtually at the same time to ensure proper carrier sensing and collision detection in the frame or subframe slot(s). As illustrated in Figure 5.29, the elapsed time between forward and reverse channels is exactly one frame slot period. This one frame slot delay between forward and reverse channels allows stations with varying propagation delays to be synchronized to the frame boundary timing. All network stations adjust their latency time such that a packet transmitted by a station in the reverse channel will be received by the same station in the forward channel exactly after one frame slot period elapses. As shown in Figure 5.29, when the header of the frame "n" is received in the forward channel the stations will act upon the information contained in the frame header and transmit their packet(s) in the frame or subframe slot(s) marked in the frame "n" while the data field of the frame "n" contains the packet(s) belonging to the previous frame "n-1". Briefly, the received frame header pertains to the frame "n" whereas the data field of the frame "n" contains the packet(s) transmitted in the previous frame slot belonging to the frame "n-1". In similar fashion, also illustrated in Figure 5.29, the network stations may transmit their packet(s) in the frame slot marked by the frame header belonging to the frame "n+1" and these packets will eventually be received in the frame slot "n+2".

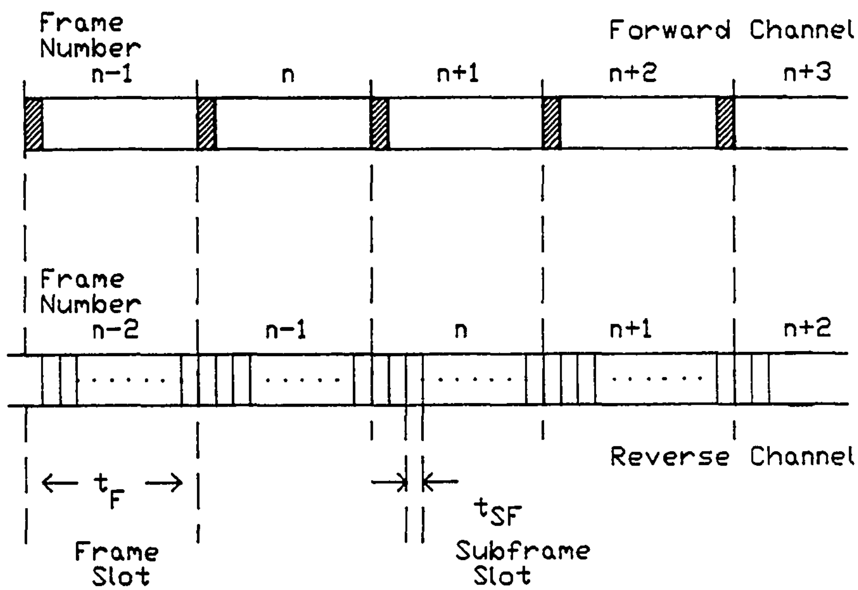



Figure 5.29. Frame Synchronization and Subframe Slotting

In summary, the forward channel provides a global time-base reference, frame slotting and synchronization for the multi-access channel by carrying sequenced and periodically transmitted frame headers which are generated by the Segment Headend. Consequently, the timing and synchronization for subdividing the frame slot into sixteen subframe slots is generated from the information contained in the frame headers. Because of exactly one frame slot difference between forward and reverse channels, the broadcast and multi-access channels are synchronized as long as frame headers are generated and latency adjustment is performed by all network stations.

**5.8 Addressing and Channel Segmentation**

For the purpose of allowing a large number of users to be supported in a MAN, a two-level hierarchical addressing scheme is employed. Since a single physical channel may not have the capacity to support the diversity and volume of applications in a MAN serving a large number of stations, several physical channels will be required. A hierarchical addressing scheme has been selected to allow multiple physical channels to be supported in a MAN. This addressing scheme takes into consideration the physical channel addressing still providing a large addressing range for individual users. For practical purposes and to provide reliable network operation with redundancy capability the network stations are divided into groups and each group is assigned a physical channel. An area code concept similar to telephone switching network is also applied by assigning a unique address for each physical channel. Contrary to the telephone system, a network station can access any physical channel by tuning to it. The hierarchical addressing scheme is ideal for the proposed media-access protocol since it allows operation on multiple physical channels and provides a means for each station to operate on any one of the channels by simply changing its address and tuning to the corresponding channel.

The destination and source address fields use the same format and consist of three octets. As shown in Figure 5.30, a station address consists of a 15-bit Individual address, a 1-bit Individual/Group address, a 7-bit Segment address, and a 1-bit Global/Local address. The 15-bit Individual Station address is unique in each MAN Segment and allows 32K users to share the channel. It should be pointed out that a couple of the highest addresses, up to 32 addresses depending on the implementation, are reserved for the broadcast addressing, the Network Manager, the Network Monitor, etc. Under normal network operation the number of stations assigned to a physical channel will not exceed the range of the individual address. However, if the channel experiences heavy traffic load frequently, it is recommended that a new channel segment be created and some of the stations be assigned to the alternate physical channel. The ability to expand the network as the demand increases is provided by the multiple physical channel assignment coupled with hierarchical addressing. Each physical channel is assigned a unique address out of 127 possible addresses. The segment address is a part of the station address, therefore, the total address range is actually 22 bits, allowing over four million users to be attached to a MAN. This channel segmentation provides the means to divide all network stations into groups and assign them a MAN Segment associated with the 7-bit segment address. Furthermore, the hierarchical addressing scheme provides global address transparency among all network stations. A network station does not have to know the physical channel location of the destination address and any station can be moved to any other MAN Segment without requiring notification of the new segment address to all other stations.

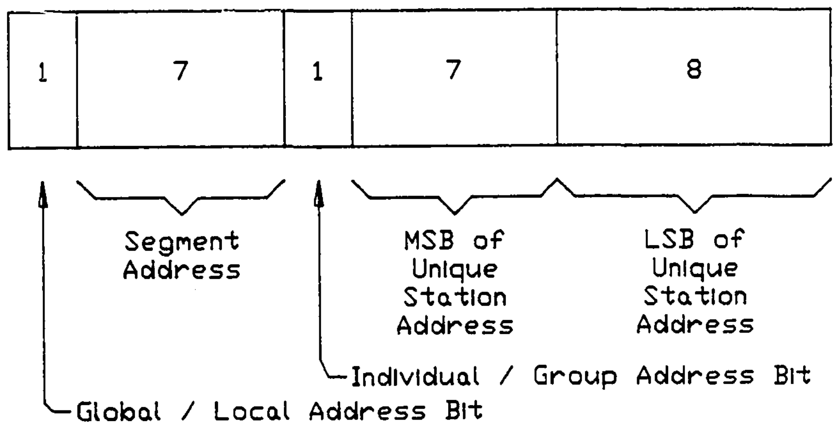



Figure 5.30. Source and Destination Address Formats

Another reason to use a hierarchical addressing scheme is to allow bridging between MAN Segments to be performed in real-time without overhead. The intersegment bridging is accomplished at the data link layer by using the segment address for routing the packets between MAN Segments. When a station sends a message to another station located in a different MAN Segment, the Segment Bridge routes the message to the destined MAN Segment by checking the segment address, but no extra delay will be incurred. A Global/Local address bit determines whether the 22-bit address including Segment, Individual/Group, and Individual addresses is to be recognized locally in the corresponding MAN Segment or is to be broadcast to all network stations located in all MAN Segments. The Individual/Group address bit determines whether the message is for a individual user or for a group of users. In summary, the hierarchical addressing scheme that is being proposed for the demand-adaptive media-access protocol provides sufficient address range for individual stations, broadcast and multicast capability, local and global addressing capability, and allows easy packet routing between MAN Segments through use of Segment Bridges.

### 5.9 Priority Assignment and Differentiation

As discussed in Chapter III, a prioritized service capability is one of the functional requirements of a media-access protocol intended for MANs. In the proposed media-access protocol the prioritized services are supported at various levels. Since the channel time is partitioned into fixed size frames and each frame can be configured independently, different priority levels are possible for each individual frame. The priority level of each frame is defined in the control field of the frame header. The Frame Service Priority Level in the control field consists of three bits and eight priority levels are possible. Priority services demand varying degrees of access and channel capacity, and the priority levels are defined to meet the specific service requirements of MANs. In Table 5.3, the service priority levels are summarized for various services.

The Network Monitor may assign a different priority level to each frame, thus providing a means of partitioning the channel time to various services. When a frame is assigned the highest priority level ("111") no other type of traffic except voice is permitted in the corresponding frame. Only stations with voice traffic may contend for that

Table 5.3. Summary of Service Priority Levels

| Frame Service Priority Level Field | Priority Level | Allowed Services |
|---|---|---|
| 0 0 0 | 0 | No Priority, All Services |
| 0 0 1 | 1 | Reserved |
| 0 1 0 | 2 | Reserved |
| 0 1 1 | 3 | Datagrams |
| 1 0 0 | 4 | Dynamic Frame and Subframe Reservations |
| 1 0 1 | 5 | Static Frame and Subframe Reservations |
| 1 1 0 | 6 | Continuous and Bulk Traffic |
| 1 1 1 | 7 | Period Traffic, Time-Constrained Applications |

frame. However, if the channel is not being used for voice traffic then there is no point in assigning the highest priority to frames, subsequently the channel utilization will adversely be affected and the frames will go by empty while stations with lower priority traffic will experience excessive access delays. Alternatively, when the channel is to be used heavily by stations with bursty traffic then the majority of frames are assigned the appropriate priority level such as the Priority Level 3 of Datagrams. The process of assigning a priority level to frames will impact the channel throughput because the usage of frames is limited to stations with a specific priority level. Therefore, when the network is initialized traffic load distribution analysis is recommended to maximize the channel throughput and minimize the access delay. One of the functions of the Network Monitor is to monitor the channel activity and determine load distribution, and assign a proper priority level to frames accordingly so that the channel utilization can be maximized. Assigning a priority level to frames can be a continuing process even when the channel is active. If a frame is received by a network station free of errors then its priority level is checked. If the station has received a request from its user for a service with a certain priority level, it compares the service's priority level with the frame's priority level, if the service's priority level is equal or greater than that of frame then the station is free to contend for the frame or subframe slot(s), otherwise it refrains from transmission. If a frame is assigned the lowest priority level all stations may transmit their packets in the frame in question regardless of their service priority levels and traffic patterns. The frame priority level is the first field checked by a station to determine whether or not it is allowed to contend for that frame or subframe slot(s). Denote the user service priority level as $P_u$, the frame service priority level as $P_f$, and let $C_{ef}$ be the condition when the frame header is received free of errors. The priority differentiation procedure is summarized as follows:

<u>Priority Differentiation Procedure</u>

Step 1: GET the Frame Header of Frame $N_f$

Step 2: IF Frame_Header ($C_{ef}$) is false then

$N_f := N_f + 1$

GO TO Step 1,

ELSE

Step 3: IF $P_u >= P_f$ then

The station is free to contend for the frame or subframe slot(s),

ELSE

The station refrains from transmission,

$N_f := N_f + 1$

GO TO Step 1.

The priority scheme described here provides a hierarchical independence of performance in a frame or subframe slot(s) by preventing lower level priority messages from affecting the network performance in a frame slot assigned to a higher priority level. In addition, fairness within each priority class is also maintained by allowing several messages of the same priority class to contend equally for the designated frame or subframe slot(s).

Robust network operation and performance is ensured by protecting the frame header from errors by use of the frame check sequence. The subframe slots within a frame carry the same priority level assigned to the related frame. The overhead required to handle the prioritized services is minimal, hence, the channel throughput is not adversely impacted. The proposed priority assignment and differentiation scheme is ideal for the diversity and volume of applications supported in a MAN.

**5.10 Static Channel-Time Reservation**

The proposed demand-adaptive media-access protocol employs a centrally-controlled mechanism to statically assign channel time to stations with heavy and continuous traffic load. The Static/Dynamic Reservation field in the frame control byte determines whether the frame in question is statically or dynamically reserved. If this bit is set to "0" the frame or subframe slot(s) can be reserved statically. The channel, if desired, can be operated exclusively in the static-reservation mode by setting the Static/Dynamic Reservation field to "0" in all frames. If the traffic load in the channel exhibits heavy and continuous patterns, the channel should be used in the static-reservation mode. For example, in point-to-point T1 carrier transmissions the communication link is totally occupied by two stations. However, a high-speed channel can also be shared by several stations generating T1 carrier type of traffic, resulting in a multiplexed T1 channel providing an economical communication link. As discussed earlier, instead of using several dedicated channels to provide point-to-point communication links between stations, it is more efficient to use a high-speed communication channel that can be shared among stations on a demand basis.

In light of these observations, for point-to-point and point-to-multipoint communications such as T1 carriers, PBX interconnects, and LAN interconnects, in which the traffic load exhibits heavy and continuous patterns, the static channel-time reservation is strongly desired. However, it should be pointed out that the proposed media-access protocol can easily be tailored and optimized for specific applications as well as for integrated services encompassing data, packetized voice, and digitized/compressed video services.

In the static channel-time reservation mode of operation, once a frame or subframe slot(s) is statically reserved, no other stations are allowed to transmit in that reserved slot. This is similar to the circuit-switching technique used in telephony in that a circuit, here a frame or subframe slot(s), is permanently established between at least two network stations, and a message broadcast capability is added with the help of a broadcast channel. In the static channel-time reservation mode, the reserved slot may be shared by a number of communicating devices, but the station which has the reserved slot has control over the slot as to who may transmit in that slot.

Reserving channel time statically requires a centralized control mechanism. This capability is provided by the Network Segment Monitor located at each MAN Segment or by the Network Manager attached to the backbone network. The static reservation of frames or subframes can be established during the network initialization process or while the network is in normal operation. It is assumed that the Network Monitor has enough intelligence and sufficient memory to manage and control static slot reservations. All static slot reservation requests are processed by the Network Monitor. The channel reservation map containing reserved frame and subframe slots is stored at the Network Monitor and is copied to the Segment Headend. When a reservation request is made it is first verified and authenticated by the Network Monitor and then it is registered in the corresponding Segment Headend. When a station is granted a reserved slot the Network Monitor keeps a record of the reserved frame or subframe slot(s) in case the network must be initialized or restarted. This process reduces overhead at the Network Monitor and allows it to perform other tasks such as network management, diagnostics, or traffic pattern analysis. The Segment Headend repeatedly cycles the frame headers carrying the information which indicates whether or not the frame or subframe slot(s) is reserved. If a slot is no longer needed a station informs the Network Monitor and subsequently the reserved slot(s) is released and the frame and subframe reservation map is updated and changes are registered at the Segment Headend.

**5.10.1 Frame Reservation.** If the channel traffic patterns and load distribution among stations are known ahead of time, frame and subframe slot(s) can be reserved statically during the network initialization. For example, for T1 carrier type of traffic for multiplexed voice communications, PBX interconnects, etc., in which the traffic patterns are continuous and periodic, frame and subframe slots are reserved for those stations at network initialization time. The Network Operator first determines the service requirements and the traffic load distribution by using network monitoring and statistical data collection tools. When channel requirements have been determined the static frame and subframe slot reservations are made by assigning frames for static reservation and setting the frame service priority level to "111". Once the frame control fields are assigned values, as shown in Figure 5.31, the Network Monitor downloads the frame reservation map to the Segment Headend and sends messages containing the reserved frame and/or subframe slot numbers to those stations which were identified for static slot reservation. The frame control field settings and the process of reserving frame and subframe slots statically during network initialization is summarized as follows:

- Determine the traffic patterns and load distribution in the channel,
- Identify the stations with continuous and heavy traffic,
- Set the Data/Network Status Frame field for data transmission, "0",
- Set the Data/Pre-empted Frame field for data transmission, "0",
- Set the Slotted/Non-Slotted Frame field for non-slotted frame format,

- Set the Static/Dynamic Reservation field for static reservation "0",
- Set the Data/Installation Frame field for data transmission, "0",
- Set the Frame Service Priority Level field to "111",
- Insert the proper frame number into the Frame Number Field,
- Store the frame reservation map in the memory of the Network Monitor,
- Reset the frame check sequence field and download the frame header to the Segment Headend for cyclic transmission in the forward channel,
- Send messages containing the frame and/or subframe slot numbers to those stations which have been assigned permanent slots.

| | | | | | |
|---|---|---|---|---|---|
| PREAMBLE | SD | 00000111 | FRAME NUMBER | FCS | SUPER SUBFRAME |

Figure 5.31. Frame Header Format of Statically-Reserved Frame Slot

Once the network is initialized, the reservation confirmation messages are immediately sent to the stations. After this process the stations may use the slots which have been assigned to them statically. As with polling and token-passing based media-access protocols, the reserved slots can be used only by the designated stations. Slots may go by empty but no other stations will be allowed to transmit. A station does not have to transmit packets in reserved slot(s) in each frame cycle. If a message generated by the user is larger than the super subframe, then the message has to be segmented into super subframes. If the channel capacity required is greater than provided by one super subframe then this protocol can be extended to allow several frame slots to be statically reserved in a frame cycle. However, the channel capacity reservation should not exceed a certain channel capacity.

If the statically-reserved frame slot is not occupied at all, it should be assigned for the dynamic reservation mode. The Network Monitor continuously monitors the channel activity and keeps statistical data for the purpose of optimizing the channel throughput.

The frames can also be reserved statically during the normal network operation. This process still requires the centralized control capability of the Network Monitor. This type of static frame reservation is highly desirable if the network traffic patterns and load distribution are known at the time of network initialization and the traffic flow between two stations is heavy and continuous. The frame control field is set to "00000000" for static reservation at the time of network initialization. The frame service priority level is set to "000" to indicate that the frame can be statically reserved during normal network operation. Once the frame header is initialized properly, network stations can now freely contend for the statically-reserved frame slots.

When a station receives a request from its user to reserve a frame slot statically, it monitors the channel for statically-reserved frame slots. If a frame fits the criterion, frame control field value of "00000000", the station contends for the frame slot by using Slotted CSMA/CD protocol. The current frame status is available to all stations simultaneously since the signal propagation delay difference is compensated for and a global time-base reference is established, thus allowing proper carrier sensing and collision detection in a given frame slot.

Assuming that statically-reserved frames are available in the channel, the station first executes the Random Number Generation Procedure used for the retransmission backoff algorithm which is described later in this chapter. The random number, denoted as $N_r$, obtained via the Random Number Generation procedure points to the first frame to be checked for the availability of statically-reserved slots. The station monitors the channel and looks for the frame number match, in the range of 0 through 255. When the station finds the desired frame, denoted as $N_f$, it checks it for the following criteria:

1) Error-free frame header, denoted as Condition $C_{ef}$,

2) Statically-reserved frame, Condition $C_{sr}$,

3) Non-slotted frame, Condition $C_{ns}$,

4) Frame service priority level value of "000", Condition $C_{pl}$.

If the criteria are met then the station listens to the channel to determine whether or not carrier is present. If the data field of frame $N_f$, a super subframe slot, is being used by other stations, then the station skips this frame and examines frame $N_f+1$ for the same criteria stated above, and it continues this process until it finds an available frame. If no carrier is sensed in the data field of frame $N_f$, then the station records the frame number, denoted $N_{af}$, and waits for the next frame cycle. In the next frame cycle the station looks for frame $N_{af}$ and verifies that the criteria for the frame header are met. If the criteria are met, the station sends a reservation request in the frame $N_{af}$ slot to the Network Monitor. The station listens to its packet in the forward channel to ensure that the request packet was received by the Network Monitor. If the reservation request packet is not received intact by the sending station, this means that the request packet either collided with another station's packet, requesting the same slot, or was mutilated by channel noise. As discussed earlier, broadband MANs are considered to be highly reliable and provide bit error rates better than $10^{-9}$. Therefore, it can be assumed that under normal operating conditions a request packet will only be corrupted due to the collisions. If the reservation request packet is not received intact by the sending station, a collision resolution algorithm, described later in this chapter, is applied by the requesting stations so that reservation requests can be scheduled for retransmission at different times in the channel.

If the reservation request packet is received intact by the sending station, this indicates that the Network Monitor received the reservation request, and the station will record the frame number, $N_{af}$. Undetected errors are negligible because the 16-bit CRC provides a BER of $10^{-9}$. While the reservation request is being processed by the Network Monitor the requesting station will transmit a self-addressed packet in the requested frame slot in succeeding frame cycles to prevent other stations from contending for the same frame slot. In the meantime, the Network Monitor processes the reservation request by altering the frame control field value to "00000111" to mark it as a statically-reserved frame, and at the same time updates its frame reservation map. Once the frame header is modified it is sent to the Segment Headend for transmission in the forward channel. When the station sees that the frame header has been modified, it is free to use the frame slot, confirming the station's static reservation request. Once the reservation request is confirmed the requesting station stops transmitting self-addressed packets and prepares to transmit actual messages. A station may not request more than one frame slot a time, and the number of frame slots that can be reserved by a station is limited to ensure that a single station does not seize the total channel capacity. When the frame slot is no longer needed the Network Monitor is informed by the station so that the frame slot can be released and made available to other stations. Let $C_i$ be the condition when a packet is received intact, $C_r$ be the condition when the slot is no longer needed, $C_{cp}$ be the condition when a carrier signal is present in the slot, and $C_c$ be the condition when a reservation confirmation is received by a station. The media-access protocol for static reservation of a frame slot during normal network operation is summarized as follows:

<u>Static Frame Slot Reservation Algorithm</u>

Step 1: CALL Random_Number_Generation Procedure

$N_f := N_r$

Step 2: GET $N_f$(MOD $N_x$)

Step 3: IF $N_f(C_{ef}, C_{sr}, C_{ns}, C_{pl})$ is false then

$N_f := N_f + 1$

GO TO Step 2,

ELSE

Listen to the channel at the beginning of the data field of Frame $N_f$,

IF $N_f$[Data_Field($C_{cp}$)] is true then

$N_f := N_f + 1$

GO TO Step 2,

ELSE

$N_{af}$ := $N_f$,

Step 4: GET Frame $N_{af}$ in the next frame cycle,

Step 5: Transmit a reservation request to the Network Monitor in Frame $N_{af}$ slot,

Step 6: IF Reservation_Request_Packet($C_i$) is false then

CALL Collision_Resolution Procedure,

GO TO Step 2,

ELSE

WHILE Reservation_Confirmation($C_c$) is false

Transmit a self-addressed packet in Frame $N_{af}$ slot in succeeding frame cycles,

ENDWHILE

Frame slot reservation request confirmed,

Station is free to transmit in Frame $N_{af}$ slot in succeeding frame cycles,

Step 7: IF $N_{af}(C_r)$ is true then

Transmit a disconnect message to the Network Monitor,

Frame $N_{af}$ slot is released.

**5.10.2 Subframe Reservation.** Subframes can be reserved statically both during network initialization, if the network's traffic patterns and load distribution are known, and during normal network operation. If the static channel capacity required is much less than a frame slot, then it is more efficient to reserve a subframe slot(s) rather than a super subframe slot. For example, in applications such as real-time voice communications, periodic data transfers, time-contrained applications, and applications with guaranteed access requirement, the static subframe slot reservation is ideal and the channel utilization can be optimized for this type of channel traffic. On the other hand, in electronic mailing, bulk data transfers, facsimile, image transfers, terminal-to-host communications, etc., where the traffic pattern is dynamic and bursty, the use of static subframe slot reservation may adversely affect the channel utilization.

Once the network service requirements, traffic patterns, and load distribution are determined, the subframe slots can be assigned statically through use of the centrally-located Network Monitor. The static reservation of subframe slots during network initialization differs considerably from that of normal network operation. In order to reserve subframe slots statically, the Network Operator first must determine the stations that fit the criteria for static subframe reservation and their needs as well as the traffic patterns and the channel's load distribution. The frame control field settings and the process of reserving subframe slots statically during the network initialization is summarized as follows:

- Determine the traffic patterns and the channel's load distribution,
- Identify the stations with continuous and periodic traffic, but short packets,
- Set the Data/Network Status field for data transmission, "0",
- Set the Data/Pre-Empted Frame field for data transmission, "0",
- Set the Slotted/Non-Slotted Frame field for slotted frame format, "1",
- Set the Static/Dynamic Reservation for static reservation, "0",
- Set the Frame Service Priority Level to "111",
- Assign a certain number of frames for static subframe slot reservation and insert these numbers into the subframe reservation map,
- Store the subframe reservation map in the memory of the Network Monitor,
- Reset the frame check sequence field of the frames allocated for static subframe slot reservation, and download the assigned frame headers to the Segment Headend for cyclic transmission,
- Send messages containing the frame and subframe slot numbers to stations to which static subframe slots were asssigned.

Once the network is initialized and the reservation confirmation messages are sent by the Network Monitor to the assigned stations, the statically reserved subframes slots can be used by the assigned network stations. The frame header format of statically reserved subframe slots is shown in Figure 5.32.

| | | | | |
|---|---|---|---|---|
| PREAMBLE | SD | 00100111 | FRAME NUMBER | FCS |

Figure 5.32. Frame Header Format of Statically Reserved Subframe Slots

Stations are not permitted to transmit messages in the reserved frames until they receive their reservation confirmations. These reservation confirmation messages may be sent in the assigned subframe slots or in the subframe slots assigned for dynamic channel-time reservations. If the frame header is configured for a slotted and statically reserved frame, network stations are not permitted to contend for the subframe slots even though some slots may go by empty. In other words, the statically reserved subframe slots can only be used by the stations which have received a confirmation to use a designated slot(s). When a subframe slot is statically reserved by a station, there is no requirement to transmit dummy or real messages in the designated slot(s) in every frame cycle, thus slots may go by empty and no other station will attempt to transmit in that slot. Note that the list of the reserved subframe slots is kept at the Network Monitor for the purpose of network restart and crash recovery.

During the network initialization process a certain number of frames are reserved for static subframe slot reservations. These frame numbers are entered in the frame reservation map stored at the Network Monitor and then downloaded to the Segment Headend for cyclic transmission in the forward channel. Network stations may reserve subframe slots statically while the network is in operation. When a user requests a statically reserved subframe slot(s), it executes the Random Number Generation procedure to obtain a randomly distributed frame number, $N_r$. The station then monitors the channel to find the frame ($N_f := N_r$) whose number matches the number obtained via the random number generator. When the desired frame $N_f$ is found then the station checks the frame $N_f$ for the following criteria:

1) Error-free frame header, denoted as Condition $C_{ef}$,
2) Statically-reserved frame, Condition $C_{sr}$,
3) Slotted frame format, Condition $C_{sf}$,
4) Frame service priority level value of "000", Condition $C_{pl}$.

If the criteria stated above are not met, the station searches for a frame $N_{f+1}$ that meets the criteria. This search starts just after the frame $N_f$ that was checked, and continues until a frame that fits the criteria is found. If no frame is available in a given time interval, at a minimum one frame cycle, then the user is notified of the condition.

If a frame that fits the criteria is found, then the station uses the Slotted CSMA/CD scheme to request a subframe slot(s) for static reservation. First, the station listens to the channel by sensing the carrier at the beginning of each subframe slot in the desired frame period. If the subframe slot checked is busy, carrier sensed, and some other station is already requesting the slot, then the station examines the next slot, and continues this process until all sixteen subframe slots are tested. If none of these subframe slots is available then it examines the subsequent available frames until at least one subframe slot is located in the channel.

If an empty subframe slot is detected the station records frame $N_{af}$ and subframe slot number(s) $N_{as}$, and waits for the next frame cycle. In the next frame cycle the station looks for frame $N_{af}$, and if frame $N_{af}$ still meets the criteria set forth then it sends a reservation request packet, containing the requesting station's address, the frame and subframe slot numbers, $N_{af}$ and $N_{as}$ respectively, etc., to the Network Segment Monitor in the subframe slot(s) being requested. The station listens to the forward channel and looks for its request packet. If the reservation request packet is not received intact by the sending station, the reservation request packet has collided with other stations' packets. This means that other stations are also requesting the same subframe slot(s). In this case, the requesting stations use a collision resolution algorithm, the same backoff algorithm used for all types of access modes, to prevent repeated collisions in the same subframe slots. The reservation request packets are scheduled for retransmission at different times in the channel. The station continues sending its reservation request packet until it is successful.

If the reservation request packet is received intact by the sending station, this means that the Network Monitor has received the reservation request free of errors. While the reservation request is being processed, the requesting station transmits self-addressed packets in the subframe slot(s) being requested in each frame cycle so that no other station attempts to transmit in the subframe slot(s) requested. The station continues to transmit self-addressed packets until it receives a reservation confirmation from the Network Monitor. Meanwhile, the Network Monitor processes the request by marking the reserved subframe slot(s) in the frame reservation map and stores this information in its memory, and then sends a confirmation message, containing the reserved frame and subframe slot numbers, $N_{af}$ and $N_{as}$ respectively, to the corresponding station in the subframe slots available for dynamic subframe reservations and datagrams. As described earlier, deleting and inserting datagrams from and to the channel is the function of the Segment Headend. If no empty frame or subframe slots are available, one of the dynamically reserved frames is pre-empted by setting the Data/Pre-Empted

Frame field of the frame control byte for the pre-empted frame condition for only one frame cycle to send the reservation confirmation messages. The stations using the pre-empted frame will suspend their operation for only one frame cycle, and they will resume operation immediately after that.

When the station receives its reservation confirmation it can send its messages in the next frame cycle. To prevent other stations from interfering with the reserved subframe slot(s) the station sends a dummy or real message in the designated slot(s) in each frame cycle. If a station desires to reserve more than one subframe slot, it monitors the channel for the required number of statically reserved consecutive slots. When the station finds a sufficient number of consecutive subframe slots it sends a reservation request packet, occupying the consecutive slots being requested, to the Network Monitor. The request packet includes the number of subframe slots requested as well as the frame and the first available subframe slot number. Next, the station follows the same procedures used for static reservation of a single subframe slot. For the purpose of improving the channel utilization the decision to request a single subframe slot or multiple subframe slots should be based on the availability of channel bandwidth, traffic loading, and the number of stations attached to the network. For example, if the number of stations attached to the network is large a single subframe slot must be reserved instead of multiple slots. Alternatively, if the channel bandwidth is large and the channel loading is light then multiple subframe slots may be reserved. This information will be downloaded to all network stations during network initialization. If the subframe slots are no longer needed, the station informs the Network Monitor to release the subframe slots so that other stations may contend for the freed channel bandwidth in the coming frame cycles. Let $C_{mr}$ be the condition when an actual message is ready for transmission. The technique for reserving a subframe slot(s) statically during normal network operation is summarized as follows:

<u>Static Subframe Slot(s) Reservation Algorithm</u>

Step 1: CALL Random_Number_Generation Procedure,
    $N_f := N_r$

Step 2: GET $N_f$(MOD $N_x$),
    $N_s := 1$

Step 3: IF $N_f(C_{ef}, C_{sr}, C_{sf}, C_{pl})$ is false then
        $N_f := N_f + 1$
        GO TO Step 2,
    ELSE
        Listen to the channel at the beginning of
        each subframe slot in Frame $N_f$,

Step 4:     IF $N_s$<= 16 then
            IF $N_s(C_{cp})$ is true then
                $N_s := N_s + 1$
                GO TO Step 4,
            ELSE

GO TO Step 5,

ELSE

$N_f := N_f + 1$

GO TO Step 2,

Step 5: $N_{af} := N_f$

$N_{as} := N_s$

Step 6: Transmit a reservation request to the Network Monitor in the designated subframe slot(s) $N_{as}$ in the succeeding frame cycle,

Step 7: IF Reservation_Request_Packet($C_i$) is false then

CALL Collision_Resolution Procedure,

GO TO Step 2,

ELSE

WHILE Reservation_Confirmation is false

Transmit a self-addressed packet in the designated subframe slot(s) $N_{as}$ in succeeding frame cycles,

ENDWHILE

Subframe slot(s) reservation request confirmed,

Station is free to transmit in the Subframe $N_{as}$ slots,

Step 8: IF $N_{as}(C_{mr})$ is false then

Transmit a self-addressed dummy packet in succeeding frame cycles,

ELSE

Transmit an actual message in Subframe $N_{as}$ slots, Step 9: IF $N_{as}(C_r)$ is true then

Transmit a disconnect message to the Network Monitor,

Subframe $N_{as}$ slots released.

### 5.11 Dynamic Channel-Time Reservation

The proposed demand-adaptive media-access protocol employs a distributed technique to allow stations with bursty and sporadic traffic loads to reserve channel time dynamically. This technique reserves channel time on a demand basis, and adapts to changing traffic loading conditions. For example, in applications such as conventional data communications, electronic mailing, digitized/compressed video image transfer, bulk data transfer, file transfer, etc., the channel capacity is only required sporadically and probably for a short period of time. In some applications such as image transfers a significant portion of the channel capacity may be required for a very short period. In order to accommodate various types of traffic loading conditions the dynamic channel-time reservation technique is proposed so that the channel may adapt to changing traffic loading conditions.

The essence of the dynamic channel-time reservation technique is the ability to easily adapt to load fluctuations in real-time while providing guaranteed channel access for time-constrained applications as well as high channel throughput and low access delay under light and heavy traffic conditions. This technique relies on the distributed intelligence provided at each network station rather than the Network Monitor. The channel can be operated totally in either static or dynamic reservation mode or both simultaneously. In other words, the channel time can be partitioned into two channel reservation modes: 1) static reservation, and 2) dynamic reservation. Each frame can be operated independently in either of two modes, one frame can be assigned for static reservation while the next one is assigned for dynamic reservation. The ratio of the dynamically and statically reserved frames is a function of the traffic patterns, the channel's load distribution, and the number of stations attached to the network. For example, for high-speed image transfers the channel may be partitioned into a small number of dynamically reserved frames, i.e., with a 10 MBPS signalling speed the total channel capacity can be divided into ten fixed size frames, allowing a data transfer rate of 1 MBPS. Alternatively, the channel can be divided into 256 fixed size dynamically reserved frames for bursty terminal type of traffic. For real-time voice communications the channel time may be partitioned into 256 frames and then the frames can be further subdivided into sixteen subframe slots to accommodate digitized voice packets. The channel partitioning and the assignment of frames for dynamic or static reservation depend on the channel requirements.

The communication network can be configured for various types of channel traffic patterns and load distributions.

Since it can be operated simultaneously both in dynamic and static channel reservation modes, integrated services ranging from data and digitized/packetized voice to digitized/compressed video communications can easily be supported. The multi-mode media-access protocol operation is made possible by using the frame header to configure each frame independently while allowing a Slotted CSMA/CD protocol to operate simultaneously in the channel.

The Static/Dynamic Reservation field in the frame control byte determines the mode of operation, static or dynamic. When this field is set to "1" the channel time in the associated frame is dynamically reserved via a distributed channel arbitration algorithm. The dynamic reservation of the channel can be applied to frame and subframe slots as in the static reservation mode. The Slotted/Non-Slotted Frame field in the frame control byte determines whether or not the frame is slotted. If the frame is non-slotted the channel is dynamically reserved via the dynamic frame slot reservation technique. If the frame is partitioned into fixed size subframe slots then the dynamic subframe slot(s) reservation technique is used.

**5.11.1 Frame Reservation.** The frames can be reserved dynamically during normal network operation through the use of a distributed arbitration mechanism which uses a similar technique to the Slotted CSMA/CD protocol. When a high volume of information is to be transmitted in a given time period, i.e., a large file transfer, it is more efficient and effective to reserve a frame dynamically for the duration of transmission rather than to reserve a frame statically. Also, in session-oriented applications such as terminal-to-host communications where a continuous communication link may be required for the duration of a session, and in real-time voice communications in which connections are made for the duration of a conversation, a frame should be reserved on a demand basis. The dynamic channel-time reservation technique provides the means to reserve a frame or subframe slot(s) dynamically on a demand basis without compromising the channel throughput and access delay.

The channel is operated in the dynamic frame reservation mode by configuring the frame headers before network initialization and then by downloading them to the Segment Headend for cyclic transmission in the forward channel. The frame control field of the frame headers configured for dynamic frame reservation mode should be initialized as follows:

- Set the Data/Network Status Frame field for data transmission, "0",
- Set the Data/Pre-Empted Frame field for data transmission, "0",
- Set the Slotted/Non-Slotted Frame field for non-slotted frame format, "0",
- Set the Static/Dynamic Reservation field for dynamic reservation, "1",

- Set the Data/Installation Frame field for data transmission, "0",
- Set the Frame Service Priority Level accordingly, depending on the type of traffic expected in the channel.

The selection of the Frame Service Priority Level should be based on several factors such as network traffic patterns, the channel's load distribution, the number of stations attached to the network, etc. Furthermore, the number of dynamically reserved frame slots in a channel is determined by the Network Operator according to these same factors. The frame header format of a dynamically reserved frame is illustrated in Figure 5.33. Once the frame headers are properly initialized by the Network Monitor and downloaded to the Segment Headend for cyclic transmission in the forward channel, network stations can contend for the frame slots assigned for dynamic reservation.

| | | | | | |
|---|---|---|---|---|---|
| PREAMBLE | SD | 00010XXX | FRAME NUMBER | FCS | SUPER SUBFRAME |

Figure 5.33. Frame Header Format of Dynamically Reserved Frame Slot

The network is initialized as follows:

1) The frame headers are transmitted cyclically by the Segment Headend,

2) Stations adjust their propagation delay latency to synchronize to the global time-base reference,

3) Network and station status information messages are exchanged,

4) The channel is made available with the transmission of a message signalling the availability of channel access.

When a station receives a request for a dynamically reserved frame slot from its user, it first executes the Random Number Generation procedure to obtain a randomly distributed frame number, $N_r$. After that the station monitors the channel and looks for the frame ($N_f := N_r$) whose number matches the one obtained from the random number generator. When the station locates frame $N_f$ it checks the corresponding frame header for the following criteria:

1) Error-free frame header, Condition $C_{ef}$,

2) Dynamically-reserved frame, Condition $C_{dr}$,

3) Non-slotted frame, Condition $C_{ns}$.

If the criteria above are met then the station invokes the Priority Differentiation Procedure to determine whether or not the priority criterion can be met.

If frame $N_f$ meets both the criteria regarding the frame header and the priority level, then the station listens to the channel, sensing the carrier, at the beginning of the data field of frame $N_f$. If a carrier is detected, meaning that some other station is already using the frame slot, then the station skips that frame and examines frame $N_{f+1}$. This process continues until a frame, referred to as $N_{af}$, which can meet both criteria, can be located and its data field is not being used by another station. If no free frame slot can be located within a predetermined interval, the station informs its user of the condition.

If no carrier is detected in the data field of frame $N_f$, the station records the frame number, $N_{af}$. The station looks for the same frame number in the next frame cycle, and contends for the frame slot along with other network stations. When the desired frame $N_{af}$ is located by the station, the criteria are again verified for frame $N_{af}$, and a self-addressed packet is transmitted in the data field of frame $N_{af}$. If the self-addressed packet is not received intact by the sending station, this indicates that a collision has occured due to packet transmissions in the same frame slot by other stations. Thus, all contending stations will know that more than one station is trying to reserve the same frame slot dynamically. In this case, the stations will have to reschedule their retransmissions for a later time. Packet collisions are resolved by a retransmission backoff algorithm. The backoff algorithm which is used to resolve the collisions is the same for all media-access modes and will be described later in this chapter. Once the station retransmits its self-addressed packet, it then listens for its own packet, and continues this process until no collision occurs in that frame slot. This technique is an extension of the Slotted CSMA/CD protocol and is used to reserve a frame slot dynamically.

If the station receives its own self-addressed packet intact, meaning that it was the only station requesting the frame slot, it records the dynamically reserved frame number, $N_{af}$. Once the frame slot is successfully used to transmit a self-addressed packet, the station is free to use the slot in the succeeding frame cycles. Once reserved, the station must send a real or dummy packet in the designated frame $N_{af}$ slot in each frame cycle as long as it desires to keep the frame slot reserved. If a station fails to transmit a packet in the designated frame $N_{af}$ slot, it may lose its exclusive right to access the frame slot, and it will have to contend for the same or another frame slot in the same manner described above.

This technique is fairly simple in terms of making reservations and keeping the reserved slot as long as it is desired. The technique to reserve frame slots dynamically uses a Slotted CSMA/CD protocol for frame slot reservations. The assigned station is required to send a packet in the designated frame slot in each frame cycle for the duration of the information transfer or session, thus informing other network stations that the frame slot is being used and is not available for contention. This technique can be used in conjunction with the static reservation scheme because each frame can be operated independently in either of two access modes.

In the case of dynamic frame slot pre-emption by the Segment Headend or the Network Monitor for the purpose of transmitting network status, emergency network management messages, datagrams from other MAN Segments, etc., the station whose frame slot is pre-empted refrains from transmitting in the pre-empted frame slot only for one frame cycle. It is suggested that a frame slot not be pre-empted for more than one frame cycle. Furthermore, the frames with the highest service priority level should not be pre-empted so that periodic traffic is not interrupted. Let $C_{ca}$ be the condition when a station desires to continue using the reserved slot. The technique to reserve a frame slot dynamically is summarized as follows:

<u>Dynamic Frame Slot Reservation Algorithm</u>

```
Step 1:   CALL Random_Number_Generation Procedure,
          Nf := Nr
Step 2:   GET Nf(MOD Nx)
Step 3:   IF Nf(Cef, Cdr, Cns) is false then
              Nf := Nf + 1
              GO TO Step 2,
          ELSE
              CALL Priority_Differentiation Procedure,
              Listen to the channel at the beginning of
              the data field of Frame Nf,
              IF Nf[Data_Field(Ccp)] is true then
                  Nf := Nf + 1
```

```
                GO TO Step 2,
            ELSE
                N_af := N_f,
Step 4:     GET Frame N_af in the next frame cycle,
Step 5:     Transmit a self-addressed packet in Frame N_af
            slot,
Step 6:     IF Self_Addressed_Packet(C_i) is false then
                CALL Collision_Resolution Procedure,
                GO TO Step 2,
            ELSE
                Frame N_af slot reserved,
                Station is free to transmit in Frame N_af slot
                in succeeding frame cycles,
Step 7:     IF N_af(C_ac) is false then
                Don't transmit in Frame N_af slot,
                Frame slot released,
            ELSE
                IF N_af(C_mr) is true then
                    Transmit an actual message in Frame N_af
                    slot,
                ELSE
                    Transmit a dummy message in Frame N_af
                    slot.
```

**5.11.2 Subframe Reservation.** Subframe slots can be reserved dynamically during normal network operation through the use of a distributed arbitration algorithm similar to that of the dynamic frame reservation. When the amount of information to be transmitted is much less than a frame period, channel utilization can be improved significantly by dynamically reserving a subframe slot(s) rather than a frame slot. This technique is ideal for those applications in which the traffic pattern is periodic or bursty, but the message sizes are small. For example, in session-oriented terminal-to-host communications the message size is relatively small and the traffic is fairly bursty. Alternatively, in real-time packetized voice communications the traffic pattern is periodic but the message size is fixed and small. Therefore, the dynamic subframe slot reservation technique can be effectively and efficiently used in real-time voice communications, connection and session-oriented applications, etc., and the channel throughput can be maximized while the access delay is maintained at a minimum. For time-constrained applications some of which require guaranteed channel access, the dynamic subframe slot reservation technique provides the best network performance while allowing bursty traffic to coexist with both periodic and continuous traffic loads. Briefly, this technique adapts to load fluctuations instantly and provides the means to reserve channel time on a demand basis. The technique of dynamically reserving a subframe slot can also be expanded to allow multiple and consecutive subframe slots to be reserved by a single station.

To operate the channel in the dynamic subframe slot reservation mode, the frame headers are created by the Network Monitor and downloaded to the Segment Headend for cyclical transmission in the forward channel. The frame control field is initialized as follows:

- Set the Data/Network Status Frame field for data transmission, "0",
- Set the Data/Pre-Empted Frame field for data transmission, "0",
- Set the Slotted/Non-Slotted Frame field for slotted frame format, "1",
- Set the Dynamic/Static Reservation field for dynamic reservation, "1",
- Set the Data/Installation Frame field for data transmission, "0",
- Set the Frame Service Priority Level accordingly.

The number of frames to be allocated for subframe slot reservations and the priority level to be assigned to the frame slots are determined by: 1) traffic patterns in the channel, 2) the traffic load distribution, and 3) the number of stations attached to the network. The frame header format of a dynamically reserved subframe slot(s) is illustrated in Figure 5.34. Once the frame headers are properly initialized by the Network Monitor and downloaded to the Segment Headend for cyclical transmission in the forward channel, network stations can then contend for the subframe slots.

| PREAMBLE | SD | 00110XXX | FRAME NUMBER | FCS | SUBFRAME # 1 | --- | SUBFRAME # 16 |
|---|---|---|---|---|---|---|---|

Figure 5.34. Frame Header Format of Dynamically Reserved Subframe Slots

When a station receives a dynamic subframe slot(s) reservation request from its user, it first determines how many subframe slots will be required to serve its user. A station is allowed to contend for more than one subframe slot. However, the subframe slots must be contiguous in a frame slot. The station invokes the Random Number Generation procedure to obtain a randomly distributed frame and subframe slot numbers, $N_{rf}$ and $N_{rs}$ respectively, and listens to the channel to locate this frame, now referred to as $N_f$. When frame $N_f$ is located in the channel, it is checked for the following criteria:

1) Error-free frame header, Condition $C_{ef}$,

2) Dynamically-reserved frame, Condition $C_{dr}$,

3) Slotted frame format, Condition $C_{sf}$.

If the criteria above are not met then the station examines frame $N_{f+1}$, and continues this process until it locates an available frame in the channel. If the criteria are met then the station invokes the Priority

Differentiation Procedure to determine whether or not the priority criterion can be met.

When a frame that meets the frame header criteria and its priority level is equal or less than that of the message is located, the station listens to the channel at the beginning of subframe $N_{rs}$ slot, now referred to as $N_s$. Listening to the channel is done by carrier-sensing. If no subframe slots are available, carrier sensed in each subframe slot following subframe $N_s$ slot, the station examines the subsequent frame $N_f+1$. If more than one subframe slot is required, consecutive slots must be checked for carrier presence, if one of the consecutive subframe slots is busy the station must check the subsequent frame $N_f+1$.

If no carrier is detected in a subframe slot(s), meaning that no other station is using the slot(s), the station records the frame and subframe slot numbers, $N_{af}$ and $N_{as}$ respectively. The station then transmits a self-addressed packet in the subframe slot(s) being requested in the next frame cycle, and listens to the forward channel to receive its own packet.

If the self-addressed packet is not received intact by the sending station, this indicates that other station(s) was trying to reserve the same subframe slot(s). The packets transmitted by the requesting station(s) are lost and this condition will be detected by all stations. In this case, a retransmission backoff algorithm is used by the contending stations to resolve the packet collisions. The retransmission backoff algorithm ensures that only one station acquires the slot in question. Briefly, a Slotted CSMA/CD protocol with two types of backoff algorithms is used to dynamically reserve a subframe slot(s). The channel throughput will not be adversely influenced due to packet collisions during slot reservations since the subframe slot reservation period is fairly short and collisions are controlled by using the proposed retransmission backoff algorithms.

If the self-addressed packet is received by the sending station without errors, meaning that no other station has transmitted a packet in the same subframe slot(s) and no collision has occured, the station records the frame and subframe slot numbers, $N_{af}$ and $N_{as}$ respectively. Generally, collision detection in broadband networks is accomplished by comparing the address fields of a packet and checking the subframe check sequence as well as by using the technique of signature analysis of the packet header. If the address fields of the transmitted packet match those of the received packet and the subframe check sequence is correct, no packet collision has occurred. Notice that the packet header also contains the number of subframe slots being occupied. When the station receives its self-addressed packet intact, it knows that the subframe slot reservation was successful and it is free to use the subframe slots to transmit its message. However, the station must transmit an actual or dummy packet in the reserved subframe slot(s) in each frame cycle regardless of whether it has any message to transmit or not. Failing to transmit a packet in the reserved subframe slot(s) releases the slot(s) for contention by all network stations. If a station fails to transmit a packet in the reserved subframe slot(s) it may lose it and must contend for a subframe slot(s) under equal conditions with other stations. Once the subframe slot(s) is dynamically reserved by a station, other stations will refrain from using the reserved slots, ensuring guaranteed access to the channel. The technique to dynamically reserve a subframe slot(s) is summarized as follows:

Dynamic Subframe Slot(s) Reservation Algorithm

```
Step 1:   CALL Random_Number_Generation Procedure,
          N_f := N_rf
          N_s := N_rs
Step 2:   GET Frame N_f(MOD N_x) and Subframe N_s(MOD 16),
Step 3:   IF N_f(C_ef, C_dr, C_sf) is false then
              N_f := N_f + 1
              GO TO Step 2,
          ELSE
              CALL Priority_Differentiation Procedure,
              Listen to the channel at the beginning of
              Subframe N_s slot located in Frame N_f,
Step 4:   IF N_s <= 16 then
              IF N_s(C_cp) is true then
                  N_s := N_s + 1
                  GO TO Step 4,
```

```
        ELSE
            GO TO Step 5,
    ELSE
        Nf := Nf + 1
        GO TO Step 2,
Step 5: Naf := Nf
        Nas := Ns
Step 6: GET Frame Naf and Subframe Nas in succeeding
        frame cycle,
Step 7: Transmit a self-addressed packet in Subframe Nas
        slot located in Frame Naf,
Step 8: IF Self_Addressed_Packet(Ci) is false then
            CALL Collision_Resolution Procedure,
            GO TO Step 2,
        ELSE
            Subframe Nas slot(s) reserved,
            Station is free to transmit in Subframe Nas
            slot in succeeding frame cycles,
Step 9: IF Nas(Cac) is false then
            Don't transmit in Subframe Nas slot(s),
            Subframe slots released,
        ELSE
            IF Nas(Cmr) is true then
                Transmit an actual message in Subframe
                Nas slot(s),
            ELSE
                Transmit a dummy message in Subframe Nas
                slot(s).
```

## 5.12 Datagram Transmission without Reservation

The channel time can also be used to transmit datagrams without requiring slot reservations. In low-duty cycle conventional data communications where the traffic pattern is extremely bursty, there is no need for dynamic channel reservation because the message length will not exceed a frame slot in a predetermined interval. Rather the messages will arrive very sporadically and connections will not be necessary. If the dynamic frame and subframe slot reservation techniques are employed for these types of applications the channel throughput will be affected adversely due to increased overhead caused by unnecessary slot reservations. The mechanism which allows datagrams to be transmitted without making reservations employs the same technique used for making dynamic frame and subframe slot(s) reservations, Slotted CSMA/CD using the proposed backoff algorithms. The datagrams can only be transmitted in dynamically reserved frame subframe slot(s) with the priority level of 3 or lower, thus not interfering with higher priority reservation requests. The datagrams are actually treated as reservation requests used in making dynamic channel-time reservations.

When a station is requested to transmit a single message, then it determines whether or not it can use the datagram transmission protocol. It then decides on the packet size, a super subframe or variable length subframe. Typically, a datagram is treated as a single, addressed entity and no acknowledgment is required. Once the channel time requirement, a frame or subframe slot(s), is determined, the station invokes the Random Number Generation procedure to obtain a randomly distributed frame and/or subframe number(s), designated as $N_{rf}$ and $N_{rs}$ respectively, and listens to the forward channel to locate frame $N_{rf}$, now referred to as $N_f$. When the frame is located in the channel it is checked for the following criteria:

1) Error-free frame header, Condition $C_{ef}$,

2) Dynamically-reserved frame, Condition $C_{dr}$,

3) Slotted frame format, Condition $C_{sf}$, for transmitting in a subframe slot(s), and non-slotted frame format for transmitting in a frame slot, Condition $C_{ns}$,

4) Frame service priority level of 3 or less, Condition $C_{p3}$.

If the criteria are not met then the station examines frame $N_f+1$, and continues this process until it locates an available frame in the channel.

If the criteria above are met, the station listens to the forward channel at the data field of frame $N_f$ or at the subframe $N_s$ slot. Similarly, listening to the channel is done by carrier sensing. If the carrier is detected in the requested frame $N_f$ or subframe $N_s$ slot(s), the station examines the subsequent frame $N_f+1$ and/or subframe $N_s+1$ slot, and continues this process until an empty slot is located.

If no carrier is detected in frame $N_f$ or subframe $N_s$ slot(s), meaning that no other station is using the slots, the station registers the frame and/or subframe slot numbers, $N_{af}$ and $N_{as}$ respectively, and transmits its packet, a datagram, in the designated slot(s) in subsequent frame cycles. After transmitting its packet the station listens to the forward channel to receive its own packet.

If the self-addressed packet is not received intact by the sending station, this indicates that other station(s) also attempted to transmit a packet in the same slot, and a packet collision has occurred. The packets transmitted by the requesting stations will be lost and this condition will be detected by the contending stations. In this case, the station will reschedule its packet retransmission by invoking the Collision Resolution procedure. Once a datagram is transmitted in a frame or subframe slot(s), all network stations are allowed to contend for the slot(s) in question in succeeding frame cycles. Let $C_{fs}$ be the condition when a station needs a frame slot and $C_{ss}$ be the condition when a station needs a subframe slot(s) for datagram service. The technique to transmit datagrams without making reservations is summarized as follows:

<u>Datagram Transmission Algorithm</u>

Step 1: CALL Random_Number_Generation Procedure,

$N_f := N_{rf}$ and/or $N_s := N_{rs}$

Step 2: GET Frame $N_f$ and/or Subframe $N_s$,

```
Step 3:   IF Nf(Cef, Cdr, {Csf or Cns}, Cp3) is false then
              Nf := Nf + 1
              GO TO Step 2,
          ELSE
              Listen to the channel at the beginning of the
              data field of Frame Nf or Subframe Ns,
              IF Datagram(Cfs) is true then
                  IF Nf(Ccp) is true then
                      Nf := Nf + 1
                      GO TO Step 2,
                  ELSE
                      GO TO Step 5,
              ELSE
Step 4:           IF Ns <= 16 then
                      IF Ns(Ccp) is true then
                          Ns := Ns + 1
                          GO TO Step 4,
                      ELSE
                          GO TO Step 5,
                  ELSE
                      Nf := Nf + 1
                      GO TO Step 2,
Step 5:   IF Datagram(Cfs) is true then
              Naf := Nf
          ELSE
              Nas := Ns,
Step 6:   GET Frame Naf and/or Subframe Nas,
```

Step 7: Transmit the datagram in Frame $N_{af}$ and/or Subframe $N_{as}$ slot(s) in subsequent frame cycle,

Step 8: IF Transmitted_Datagram($C_i$) is false then

CALL Collision_Resolution Procedure,

GO TO Step 2,

ELSE

Datagram successfully transmitted,

Frame $N_{af}$ and/or Subframe $N_{as}$ slot(s) are available in succeeding frame cycle.

## 5.13 Retransmission Backoff Algorithms

During the frame and subframe slot reservation process as well as datagram transmissions packet collisions may occur because two or more stations may be competing for the same slot. A retransmission backoff algorithm is required for the purpose of resolving packet collisions occurring in the frame and subframe slots. The retransmission backoff algorithms generally used by the most common contention-based protocols, i.e., the binary exponential backoff algorithm, the tree-search algorithm, etc., are not appropriate for the proposed demand-adaptive media-access protocol. The need for a new retransmission backoff algorithm stemmed from the inapplicability of the current algorithms. The current backoff algorithms are mainly intended for one or two hundred stations whereas the proposed media-access protocol must handle a larger number of stations scattered in a large geographical area. If the current algorithms are employed the resolution time might be too long and the overhead will eventually increase due to excessive collisions, in turn resulting in lower channel throughput.

The retransmission backoff algorithms required for the proposed demand-adaptive media-access protocol must be capable of handling a large number of stations and large signal propagations delays. Another requirement is that the algorithms must be adaptable to channel time-slotting and frame and subframe slot structures. In this dissertation, two new retransmission algorithms are being proposed for adaptation to the demand-adaptive media-access protocol. Both of these algorithms should work well for static and dynamic reservation techniques as well as datagram transmissions.

**5.13.1 Priority-Oriented Collision Arbitration Algortihm.** In order to support prioritized services in the proposed media-access protocol, each frame is assigned one of eight priority levels. When a station receives a request from its user, it includes a priority level for the service being requested. A station may compete for frame and subframe slot(s) if and only if the service's priority level is equal or greater than that of the frames in question. This limits the number of frame slots that can be contended for, consequently reducing competition in certain frame slots. A priority class is defined as a set of priority levels. The highest priority level in a class will be referred to as "High Priority", and the rest of the priority levels as "Low Priority". For example, if a station has a message with a priority level of 5 then it may compete for channel time in frames with priority level 5 or below but not higher. In this example, the priority level 5 represents the "High Priority", and the levels 0, 1, 2, 3, and 4 are referred to as the "Low Priority". The "High and Low Priority" definitions are required to describe the priority-oriented collision arbitration algorithm.

For use in retransmission backoff algorithms an 18-bit random number generator is employed, refer to Figure 5.35. The station address is used to construct an 18-bit seed so that the random numbers generated are geometrically distributed. The random number generator is constructed as follows: bits 3 and 18 are exclusive-ored and the 18-bit shift register using the resultant as a carry bit is shifted to the left for one bit, implementation of the polynomial $x^{18}+x^3+1$. This polynomial is used because it is simple to implement and provides a geometrical distribution of random numbers [SIRA 84]. During station initialization process the seed is loaded into the generator, and several operations are performed before the random number is used by the media-access protocol. When a random number is required to determine packet transmission time, the least significant sixteen bits are extracted from the 18-bit shift register, and the least and most significant bytes are exclusive-ored. As illustrated in Figure 5.36, the resultant is used as a random frame number, $N_{rf}$. Furthermore, the least and most

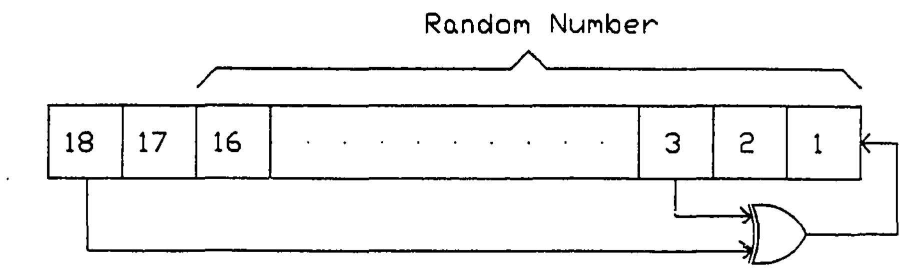



Figure 5.35. Random Number Generator

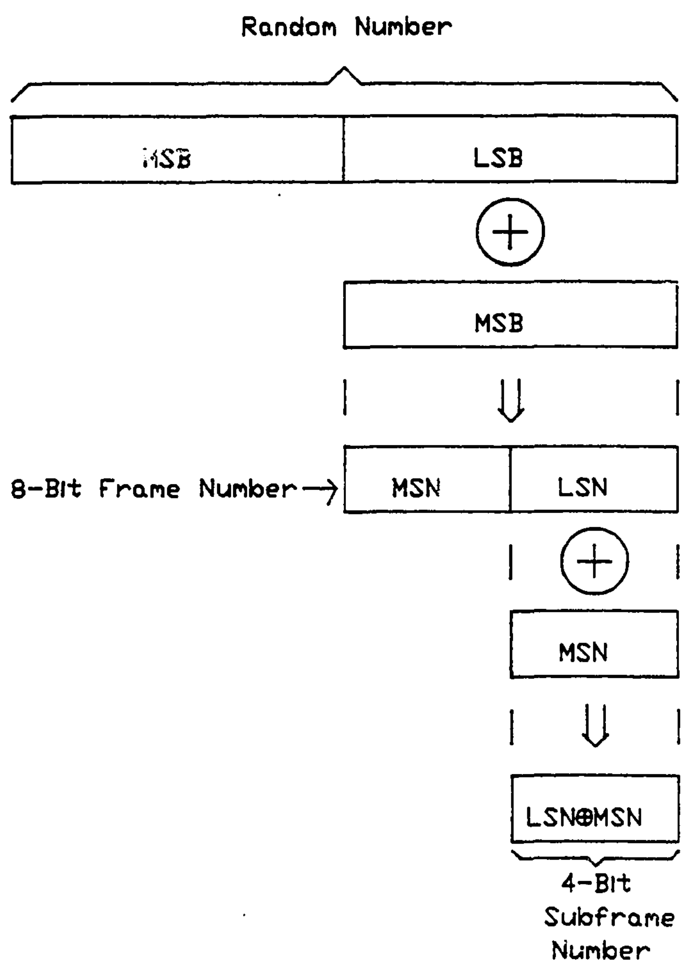



Figure 5.36. Random Frame and Subframe Number Generation

significant nibbles of the resultant byte are also exclusive-ored, and subsequent result is used as a random subframe slot number, $N_{rs}$. Once a number is obtained, this process is repeated so that the next random number is geometrically randomized. This process is called "Random Number Generation", and is used by both static and dynamic reservation techniques as well as datagram transmission.

In the demand-adaptive media-access protocol, when a station receives a request from its user for packet transmission, it first determines its message priority level, $P_u$, and then invokes the Random Number Generation procedure to randomly point to a frame and/or subframe slot. Once the frame and/or subframe slot is located, its availability is checked by comparing the frame service priority level, denoted as $P_f$, with the message priority level. The "Logical Priority Level" ($P_x$) is defined to provide a means to differentiate the stations' priority levels as "High Priority" and "Low Priority", and initially assumes the value of the frame service priority level which is determined at the reception of the first available frame. At the start of this process, if the station's message priority level is greater than the frame service priority level, it is assigned "High Priority" $P_h$, otherwise "Low Priority" $P_l$. If the station has a datagram to transmit it is assigned "Low Priority". If the frame or subframe slot is available a Slotted CSMA/CD protocol is employed to reserve the slot in question. If only one station attempted to transmit in the slot randomly selected, no collisions occur in the designated slot, the reservation as well as datagram transmission processes are successfully completed. Conversely, if a packet collision occurs in the slot being requested, the priority-oriented collision arbitration algorithm is employed to resolve collisions.

In this technique, when a collision occurs the stations assigned with the "Low Priority" level are forbidden to compete in the frame or subframe slot(s) being requested, and these stations compete for channel time by restarting the priority-oriented collision arbitration process. Alternatively, to reduce collisions among the stations assigned with "High Priority" level, the "Logical Priority" level is incremented by one.

The stations with service priority level greater than the new "Logical Priority" level are assigned to "High Priority", and the rest to "Low Priority". When the new priority assignment process is completed, the stations transmit their packets in the designated slot(s) in the succeeding frame cycle. If collision still occurs then the priority class assignment process is repeated as long as the "Logical Priority" level does not exceed the highest priority level 7.

If a collision still occurs at the highest priority level, then the colliding stations employ the well-known technique of binary search by using station address to resolve collisions among stations with the highest priority level 7. Even though many steps may be required to resolve collisions, the priority-oriented collision arbitration algorithm can be implemented for both static and dynamic channel-time reservation techniques as well as datagram service. This algorithm is capable of handling a large number of stations because of its ability to reduce competition in the channel to a single station by use of the binary search mechanism and its use of Slotted CSMA/CD protocol in a fraction of the channel capacity, a frame or subframe slot(s), rather than over the entire channel. Let i point to one of the station address bits and $B_i$ be the logical value of the address bit being pointed to. The priority-oriented collision arbitration algorithm is summarized as follows:

<u>Priority-Oriented Collision Arbitration Algorithm</u>

Step 1: CALL Random_Number_Generation Procedure,
$N_f := N_{rf}$ or $N_s := N_{rs}$, $i := 1$

Step 2: GET Frame $N_f$ or Subframe $N_s$,
$P_x := P_f$

Step 3: IF $P_u >= P_x$ then
Stations are assigned "High Priority" level $P_h$, and transmit their packet in Frame $N_f$ or Subframe $N_s$ slot in succeeding frame cycle,
ELSE
Stations are assigned "Low Priority" level $P_l$ and refrain from transmission,
GO TO Step 1,

Step 4: IF Transmitted_Packet($C_i$) is true then
Transmission is successful,

```
          ELSE
              IF Px < 7 then
                  Px := Px + 1
                  GO TO Step 3,
              ELSE
Step 5:           IF Bi is even then
                      Stations with even Bi address bit
                      transmit their packet in Frame Nf or
                      Subframe Ns slot in succeeding frame
                      cycle,

                  ELSE
                      Stations with odd Bi address bit
                      refrain from transmission,
                      GO TO Step 1,
Step 6:   IF Transmitted_Packet(Ci) is true then
              Transmission is successful,
          ELSE
              i := i + 1
              GO TO Step 5.
```

A variation of this scheme is also proposed by defining a different packet size for each service priority level. However, this scheme can only be used by static and dynamic channel reservation techniques to reserve frame and subframe slot(s), but not for datagram service. In this technique, stations transmit their reservation request packets varied in size depending on their service priority level, the higher the priority the longer the packet (see Figure 5.24). If no collision occurs in the slot, this indicates that only one station successfully attempted to reserve the slot in question. On the other hand, if collision is detected in the slot, the requesting stations continue to listen to the channel to determine whether or not other network stations with a higher priority level are requesting the same slot. If a station detects a carrier in the slot beyond its reservation packet, then it knows that another station with higher priority level is requesting the same slot. The packets transmitted by both stations will be lost, but the stations with lower priority level will refrain from transmitting in the designated slot, and the other stations will be free to compete for the same slot in the next frame cycle. This process is repeated until all collisions in the designated slot are resolved. If collisions still occur among stations with the highest priority level then the binary search mechanism is employed as described previously.

The priority-oriented collision arbitration algorithm allows services with different priorities to be offered in the same channel. Furthermore, this algorithm provides hierarchical independence of performance, fairness within each priority class, and robust operation.

**5.13.2 Randomly Distributed Retry Algorithm.** The frame and subframe numbering mechanism being used in the demand-adaptive media-access protocol provides a means for establishing a simple retransmission backoff algorithm. In a partitioned and synchronized channel the packet retransmission rescheduling can be done in a randomly distributed fashion. This scheme coupled with frame and subframe numbering structure is applicable to both static and dynamic channel reservation techniques as well as datagram transmission.

When a packet must be scheduled for retransmission due to collisions in the slot(s) or channel noise, a station invokes the Random Number Generation procedure to obtain a randomly distributed frame and/or subframe number. As illustrated in Figure 5.37, the frame and/or subframe numbers obtained through the Random Number Generation process point to the frame and subframe slot(s) in the channel respectively.

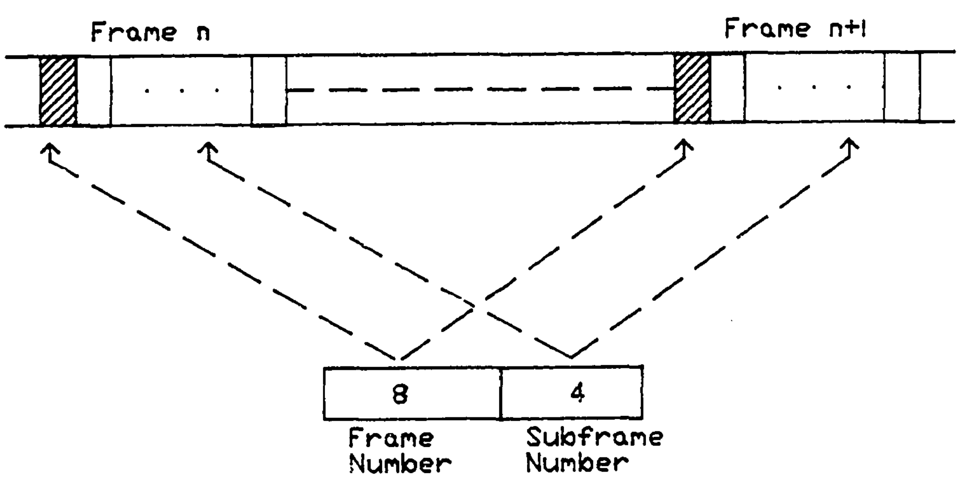



Figure 5.37. Randomly Distributed Retry Algorithm

When the frame and/or subframe slots are located in the channel, the slot availability is checked by using Slotted

CSMA/CD protocol. If the frame or subframe slot being checked is not available, the station examines the subsequent frame or subframe slot, and it continues to do so until an available slot is located. If no carrier is detected in the slot, the station transmits its packet in the designated slot in the succeeding frame cycle.

In the static and dynamic channel reservation modes, if no collision occurs in the transmitted slot(s), then the slot(s) is successfully reserved. If a collision occurs the station invokes the Random Number Generation procedure to obtain a new randomly distributed frame and/or subframe slot(s) number. If the number of packet retransmissions exceeds a certain limit, then the retransmission window is further widened by using the random frame number obtained through the Random Number Generation procedure as a frame cycle number as well as a frame number. In this case, the station first locates its retransmission frame cycle, and then searches for the frame and/or subframe slot. Once the slot in question is located, the station employs a Slotted CSMA/CD protocol to transmit its packet. This process continues until retransmission is successfully completed.

The randomly distributed retry algorithm, which is very simple to implement, provides a fairly robust operation and distributes the retries randomly and geometrically. However, the overhead caused by collisions is higher than the priority-oriented collision arbitration algorithm. This algorithm is capable of handling large numbers of stations because it evenly distributes the retries across frame and subframe slots, and it can further reduce competition in the channel by widening the retransmission window through use of frame cycles as part of retransmission process. This retransmission backoff algorithm is ideal for datagram transmission, but it is also applicable to both static and dynamic channel reservation techniques.

### 5.14 Multi-Mode Protocol State Diagram

The demand-adaptive media-access protocol can operate in different media-access modes determined by the service request received from the station's user. Considering that a MAN would support a wide range of integrated services encompassing data, digitized/packetized voice, and digitized/compressed video services, the channel capacity requirement for each class of service will be different. In order to allocate channel capacity on a demand basis and still provide high channel utilization and low access delay, the channel must be operated in different media-access modes determined by the service requirements. For example, under bursty traffic conditions the channel time required is fairly short and transmission requests are sporadically distributed. Alternatively, under continuous and periodic traffic loads, as the case in real-time voice communications, the channel capacity will be needed at regular intervals, and load distribution and concentration will equally be balanced.

For the purpose of allowing the diversity and volume of applications to be served in a MAN, the service requests must first be sorted out so that the appropriate media-access mode can be determined. As discussed earlier, the demand-adaptive media-access protocol will be able to handle the required data rates over four orders of magnitude and operate in various access modes such as static channel-time reservation, dynamic channel-time reservation, and datagram transmission.

A station may operate in the following media-access modes: 1) "Slotted CSMA/CD", 2) "Synchronous Time-Division Multiple Access", STDMA, namely Static Channel-Time Reservation, and 3) "Asynchronous Time-Division Multiple Access", ATDMA, namely Dynamic Channel-Time Reservation. When a network station is turned on, its initialization process includes a global time-base reference synchronization, propagation delay latency adjustment, diagnostics, etc. After initialization, the station enters the "Dormant State", that is the state in which the station is ready to accept a service request from its user. In "Dormant State", stations listen to the channel constantly, and maintain proper station operation while waiting for service requests. The service requests are tagged with service priority level, channel capacity requirement, type of service, destination station address, etc. After the network is initialized all stations are assumed to be in "Dormant State" and listening to the channel for network management messages.

A station may enter any one of the media-access modes from the "Dormant State" at any time, but transitions between media-access modes are not allowed excepting that Slotted CSMA/CD is used for making slot reservations. The multi-mode media-access protocol state diagram is illustrated in Figure 5.38.

The Slotted CSMA/CD media-access mode is used for datagram transmissions as well as transmission of reservation requests generated in other media-access modes, ATDMA and STDMA. The ATDMA mode is used to reserve channel-time, in units of frame or subframe slots, dynamically whereas the STDMA mode is employed to reserve channel-time statically.

In the "Dormant State", the station idles and waits for a service request. When a service request is received the station determines the media-access mode in which it is going to operate. After determining the media-access mode, the station decides on the service priority level and the packet length which may range from as small as a subframe to as large as a super subframe.

If the service request is for a datagram transmission the station enters the Slotted CSMA/CD access mode. Once the datagram is successfully transmitted, the station reverts back to the "Dormant State" for another service request.

When a station, while in the "Dormant State", receives a dynamic channel-time reservation request from its user, it immediately enters the ATDMA access mode. The station first decides on the packet length according to the service request requirements. The packet length may be as small as a subframe, but may not exceed a frame slot. In the ATDMA

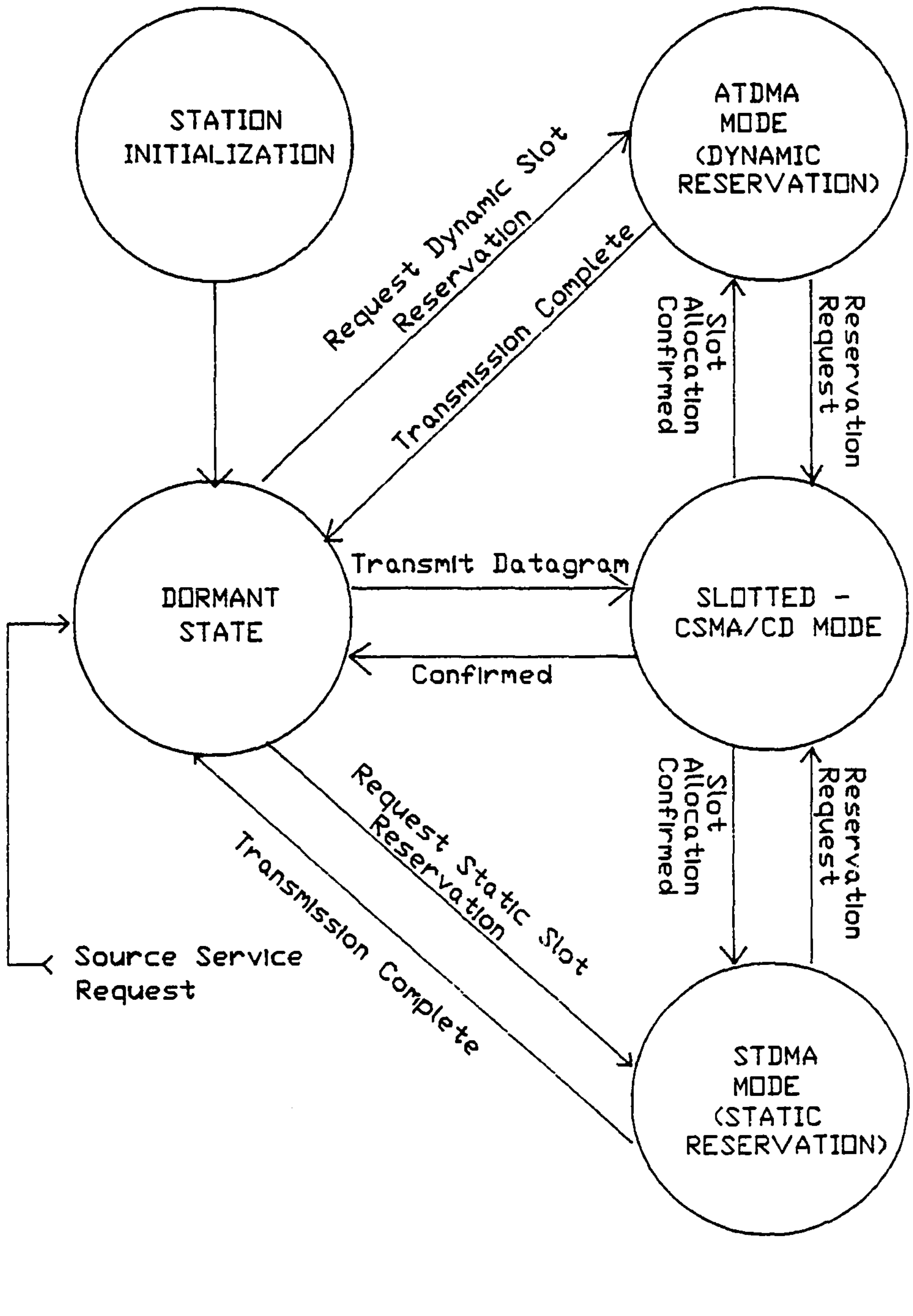



Figure 5.38. Multi-Mode Media-Access Protocol State Diagram

access mode, the station enters the Slotted CSMA/CD access mode to make slot(s) reservation. After successful confirmation of the slot reservation, the station reverts back to the ATDMA access mode. When the slot is no longer needed by the station, it is released and the station re-enters the "Dormant State" for another request.

If the service request received from the station's user is for static channel-time reservation, the station first determines the channel access parameters such as packet length, service priority level, etc., and enters the STDMA access mode. While in this mode, the station enters the Slotted CSMA/CD access mode to make a slot reservation. While in the Slotted CSMA/CD mode, the station sends a reservation request to the Network Monitor. Upon the confirmation of the reservation request, the station reverts back to the STDMA access mode, and is allowed to transmit its packet in the designated slot(s) in succeeding frame cycles. When the statically reserved slot(s) is no longer needed, the slot is released and the station goes back to the "Dormant State".

As stated earlier in this dissertation, a single media-access mode of operation will not be sufficient and effective in sharing a channel for a variety of integrated services in a MAN. The proposed demand-adaptive media-access protocol coupled with its multi-mode operation provides the means to allow stations to share an ubiquitous channel effectively and efficiently to support integrated services in a MAN.

## 5.15 Estimated Effective Channel Utilization

There are many papers in the recent literature regarding LANs, radio and satellite networks, etc., and their channel throughput under various conditions [ABRA 77, ARTH 82, BALA 79, BUX 81, CARP 84, CHLA 80, KLEI 75b, LAM 79, METC 76, MEIS 82, NUTT 82, STUC 83, TOBA 78]. In the proposed media-access protocol the channel can be operated in various access modes ranging from static and dynamic reservation to datagram transmission. Therefore, the estimated effective channel utilization of the demand-adaptive media-access protocol involves the analysis of the following access modes: 1) static and dynamic channel-time reservation protocols, and 2) Slotted CSMA/CD protocol for making reservations and transmitting datagrams.

In the static and dynamic channel-time reservation modes, the upper bound of the channel utilization represents a fraction of the channel bandwidth required for making reservations. This efficiency can, in general, be increased at the expense of larger access delay, as is the case in a Slotted CSMA/CD protocol, since the larger data frame and subframe time associated with more efficient operation means that random message arrivals must wait longer on the average. In the static and dynamic channel-time reservation modes, lower access delays are achieved under varying traffic load conditions by the use of Slotted CSMA/CD protocol for making reservations.

For a fixed channel data rate, the increased delay for a larger number of stations with static and dynamic channel-time reservation protocols applies only to the reservation portion of the channel using a Slotted CSMA/CD protocol. The use of static reservation versus dynamic reservation involves a number of performance tradeoffs including channel utilization, reservation delay variance, stability, etc.

When high traffic concentrations and nonvarying loads exist at most or all stations in a network, a static channel-time reservation protocol will, in general, give the best overall performance. Where bursty station traffic or variable loading prevails, the dynamic channel-time reservation protocol will provide significant improvements in both channel efficiency and access delay. Where very small message sizes are sent by a large number of very bursty stations, datagram service without reservations using Slotted CSMA/CD, provides significant performance improvements relative to the static and dynamic reservation protocols.

Since the proposed media-access protocol operates in various access modes, channel utilization must be calculated separately for each access mode. The overall effective channel utilization should also include the overhead required to operate the channel in multi-access modes.

The channel utilization of a reservation protocol may reach the ideal level, maximum attainable throughput, assuming that the reservation process introduces a negligible amount of overhead. However, the effect of the reservation process should also be included in the calculation of the estimated effective channel utilization.

Once a slot is reserved the overhead associated with making reservations is totally eliminated. In static and dynamic channel-time reservation techniques, a Slotted CSMA/CD protocol is used to make reservations as well as to transmit datagrams. The total overhead in the channel consists of three parts: 1) the overhead incurred due to time-slotting and synchronization, the frame header overhead, 2) the overhead incurred due to subframe synchronization and delimitation, the subframe header overhead, and 3) the overhead caused by the packet collisions in frame and subframe slots.

Let "d" be the data rate in bits per second, "p" be the preamble size in bits, "q" be the guard band size in bits, and "w" be the frame header size in bits. Define $B_{fh}$ as the the frame header overhead.

$$B_{fh} = (N_f * w) / d$$

$$w = p + q + 32$$

$$B_{fh} = (N_f * (p + q + 32)) / d$$

For example, with a data rate of 10 MBPS per channel, 256 frames in a frame cycle, a preamble of 8 bits, and a guard band of 8 bits, the frame header overhead is less than 0.05 percent of the total channel capacity. As the number of frames in a frame cycle decreases, the overhead will further decrease, and it can be neglected in the channel utilization calculations.

Let X represent the percentage of the slotted frames in a channel. The subframe header overhead $B_{sh}$ is given by:

$$B_{sh} = (X * N_f * 16 * q) / d$$

The total overhead caused by collisions consists of two parts: 1) the overhead due to collisions in frame slots, 2) the overhead due to collisions in subframe slots. Define $B_{cs}$ as the total overhead caused by collisions, $B_{cfs}$ as the overhead due to collisions in frame slots, and $B_{css}$ as the overhead due to collisions in subframe slots.

$$B_{cs} = B_{cfs} + B_{css}$$

The fundamentals of CSMA/CD protocols are well known and the channel utilization of a CSMA/CD network using a simple but very useful model has been calculated [METC 76]. It is assumed that there are always M stations seeking to transmit packets. The channel utilization $S_c$ of the Slotted CSMA/CD protocol, as proposed in this dissertation, is calculated as follows:

$$S_c = P / (P + d * T * F(M))$$

where P is the number of data bits transmitted, T is the slot time (frame or subframe slot), and F(M) is a slowly-varying function of the number of stations. F(M) varies only from 1.44 when M=5 to 1.717 when M reaches infinity. Let $T_f$ be the frame slot and $T_s$ be the subframe slot.

$$B_{cfs} = (1 - X) * d * T_f * F(M)$$

$$B_{css} = X * d * T_s * F(M)$$

$$B_{cs} = ((1 - X) * T_f + X * T_s) * d * F(M)$$

The overall channel overhead is given by:

$$B = B_{fh} + B_{sh} + B_{cs}$$

The estimated effective channel utilization is defined as the ratio of the data bits transmitted to the data bits transmitted plus the overall overhead.

$$S = P / (P + B)$$

where S is the estimated effective channel utilization.

$$S = P / (P + B_{fh} + B_{sh} + B_{cs})$$

When the channel is operated exclusively in static and dynamic reservation modes with a non-slotted frame format, assuming that the reservation process is complete and the overhead associated with subframe headers and collisions is eliminated. In this case, the effective channel utilization $S_1$ is given by:

$$S_1 = P / (P + B_{fh})$$

Since $B_{fh}$ is relatively low and can be neglected, the channel utilization is at maximum, close to the ideal level, when the channel slots are totally reserved.

If the channel is operated exclusively in static and dynamic reservation modes with slotted frame format, the channel utilization $S_2$ is given by:

$$S_2 = P / (P + B_{fh} + B_{sh})$$

In this mode of operation, the effective channel utilization may also reach the ideal level since the overhead associated with frame and subframe headers is negligible.

During datagram transmissions and the reservation process, the channel is operated with a Slotted CSMA/CD protocol. This represents the worst case condition for the proposed protocol, and the effective channel utilization will mainly be determined by the overhead caused by collisions. However, the overall channel utilization will still be higher than that with the CSMA/CD protocol since the channel is slotted and collisions may only occur in frame and/or subframe slots. Even if the channel is operated with a Slotted CSMA/CD protocol, the estimated effective channel utilization will be higher than 85 percent [METC 76]. This means that the lower bound of the channel utilization equals that of Slotted CSMA/CD protocol, and the upper bound reaches the ideal level.

In summary, the proposed media-access protocol, which combines the best attributes of reservation and contention protocols, provides significant performance improvements under varying load conditions created by the integrated services. The estimated effective channel utilization could be as low as that of the Slotted CSMA/CD protocol and as high as that of the ideal TDMA protocol.

# CHAPTER VI

## CONCLUSIONS AND SUGGESTIONS FOR FUTURE RESEARCH

Trends in the growth of communication networks seem to indicate that the next generation of networks will have to support integrated digital services. The present media-access protocols, however, are not directly applicable to very large networks which are assumed to support a large number of users scattered over a large geographical area for data, packetized voice, and digitized/compressed video communications services provided in a single shared channel. New media-access protocols are needed which can provide cost effective, high-speed communications for a large population of users spread over a wide geographical area. Under these circumstances, we feel that Metropolitan Area Networks provide attractive solutions to the design of integrated services communication networks. Specifically, with the availability of ultra high-speed transmission capabilities over coaxial-based Institutional Networks and CATV Facilities, the delivery of integrated services to the business and home markets has become economically and technically feasible.

The objective of this research was to develop a new media-access protocol and network topology which can be applied to MANs over which thousands of users scattered over a wide geographical area can share network resources to interchange information encompassing data, voice and video.

The emphasis is on a coaxial-based MAN since there exists an abundance of bandwidth in CATV Facilities which are currently being used mainly for video entertainment services. Major contributions of this research may be classified in four categories:

- o Definition and characterization of Metropolitan Area Networks including network properties, topologies, and service requirements,
- o Comprehensive study and analysis of several media-access protocols intended for Satellite, Radio, and Local Area Networks as applied to MANs. Statement of the deficiencies of the present protocols. Definition of the functional requirements of a media-access protocol to support integrated services in a MAN environment,
- o Definition and development of a new hierarchical communication network topology and architecture using segmentation structure,
- o Definition and development of a new media-access protocol that meets MAN service requirements.

We believe that the proposed demand-adaptive media-access protocol is an optimum scheme to provide integrated services encompassing data, packetized voice, and digitized/compressed video communications in a MAN environment. We note that because of its flexibility, architecture, and its adaptability to varying traffic load conditions the proposed protocol may be tailored and effectively used for specialized services. For example, a Private Branch Exchange (PBX) can be implemented by using the new media-access protocol to offer multiplexed voice and data services simultaneously. Alternatively, the proposed protocol will allow the allocation of a large portion of the total channel capacity for high-speed communications in which the channel capacity demand is high for only a period of time. In another application, a MAN may be constructed by providing a high-speed communication channel, a voice channel, a data communication channel, a video communication channel, and a combination of the above over a single medium such as coaxial-cable, fiber optices, etc.

The demand-adaptive media-access protocol in conjunction with the hierarchical network topology, as proposed in this dissertation, uses a unique multi-mode access mechanism which allows integrated services to be offered to a large number of users in a shared channel environment while providing the flexibility to be tailored for specific applications. This protocol, which is independent of the signalling speed and medium being employed, is ideal for MANs as well as LANs, Satellite and Radio Networks, PBXs, etc, to integrate data, voice and video services on a ubiquitous channel.

Extensions to this research

The analytical models and simulation studies of the most commonly known media-access protocols are widely available [ARTH 82, BRUT 83, CHLA 79, STUC 83]. These include the applications of queueing theory, scaled down computer simulations, protocol verification models, performance analysis with testbed facilities, etc. As of now, most efforts in performance evaluation have been concentrated on the CSMA/CD, Token-Passing, Polling, and Reservation protocols. Little attention has been given to multi-mode media-access protocols in the performance studies. As the integration of data. voice and video services on a single medium became a necessity in users' applications, the need for a multi-mode media-access protocol has been recognized and we expect that considerable research efforts will be devoted in this direction in coming years. We hope that this research will lead the way in the evolution of multi-mode media-access protocols.

The proposed media-access protocol uses a combination of the CSMA/CD and reservation protocols modified for simultaneous operation. We recommend that the proposed protocol be specified with one of the upcoming protocol verification and specification languages so that its correctness can be proven and any holes in the access modes can be eliminated. It is also suggested that an analytical model of the proposed protocol be constructed and a performance analysis encompassing channel throughput and access delay be performed analytically. Using the analytical model a computer simulation should be performed to verify the correctness of the protocol and the performance results obtained in the analytical study. Because of computing power and memory limitations we may not be able to simulate the protocol for a large population of users. Therefore, a scaled down version, i.e., several hundreds of users instead of thousands, may be sufficient to obtain useful results. In analytical studies and computer simulations, a traffic model should include data, voice and video traffic patterns. We suggest that a performance analysis be performed for the following cases:

1) Data traffic,
2) Voice traffic,
3) Video traffic,
4) Voice and data traffic,
5) Data, voice, and video traffic.

We recommend that the performance tests be repeated for the following channel access modes:

1) Static channel-time reservation,
2) Dynamic channel-time reservation,
3) Slotted CSMA/CD,
4) Static channel-time reservation and Slotted CSMA/CD,
5) Dynamic channel-time reservation and Slotted CSMA/CD,
6) Static and dynamic channel-time reservation and Slotted CSMA/CD.

Another research area is the verification of the correctness of the proposed protocol and the performance results obtained with analytical study and computer simulation by using a reconfigurable network facility. The performance analysis should be repeated for the same criteria suggested in computer simulations. The results obtained from analytical studies, computer simulations, and real-time testbed studies should be correlated to satisfactorily verify the correctness of the proposed media-access protocol.

As a part of a complete communication network system and a complementary extension to the proposed protocol, we also suggest that network and transport layers be studied as to how they interface to the demand-adaptive media-access protocol and use the services of the protocol to support data, packetized voice and video communications. As a final extension to this research, we believe that the demand-adaptive media-access protocol can be expanded and proposed to IEEE 802.6 Metropolitan Area Networks Standards Committee to be considered as a MAN standard protocol since there is a lack of standards for MANs supporting integrated services.

In this dissertation, we proposed and developed a new multi-mode media-access protocol to be used in the next generation of communication networks to support integrated services. This research should generate interest in multi-mode media-access protocols and may lead to several implementations not only for MANs, but also for Satellite and Radio Networks as well as data/voice LANs. This research may also give a new dimension in the integration of data, packetized voice, and digitized/compressed video services in MANs and Global Networks. Major effort should be devoted to propose the demand-adaptive media-access protocol for standarization and practical implementations. We strongly believe that a communication pipeline carrying data, voice and video simultaneously has become a reality in the information age we are living in, and integrated services can be offered to businesses and homes at economical cost levels.

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