Performance Analysis of Star Architecture Packet-Switched VSAT Network Using Roll-call Polling Multiple Access Scheme

by

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(ABSTRACT)

Data link control, multiple access, and flow control for data communication have at last advanced to the state that it is possible for applications that require combining these techniques to be carried out. Therefore, research efforts are now beginning to focus on the performance of these applications, rather than the previous trend of carrying out the performance of each scheme separately.

This study analyses the performance of a Very Small Aperture Terminal (VSAT) satellite star network. The network uses roll-call polling as its multiple access scheme, High-level Data Link Control or HDLC, and go-back-N Automatic Repeat Request, for error control. The network, is a VSAT packet-switched network, that carries out its communications task on a Time Division Multiplexing (TDM) satellite channel. This research consists of three major parts. First, the performance analysis of the single-hop star architecture network is carried out. This includes the study of a polling communication system for the inbound, VSAT-to-Hub line. The time delay of a packet using the inbound line is evaluated. Secondly, the performance analysis of the TDM outbound Hub-to-VSAT line is represented. The throughput of the multiplexed system for the outbound Hub-to-VSAT line, as well as the average time delay of a packet are determined. Thirdly, both the analysis of the inbound, and outbound lines are combined to provide the performance of the double-hop architecture of the network. The time delay of a VSAT-to-VSAT packet is found.

Acknowledgements

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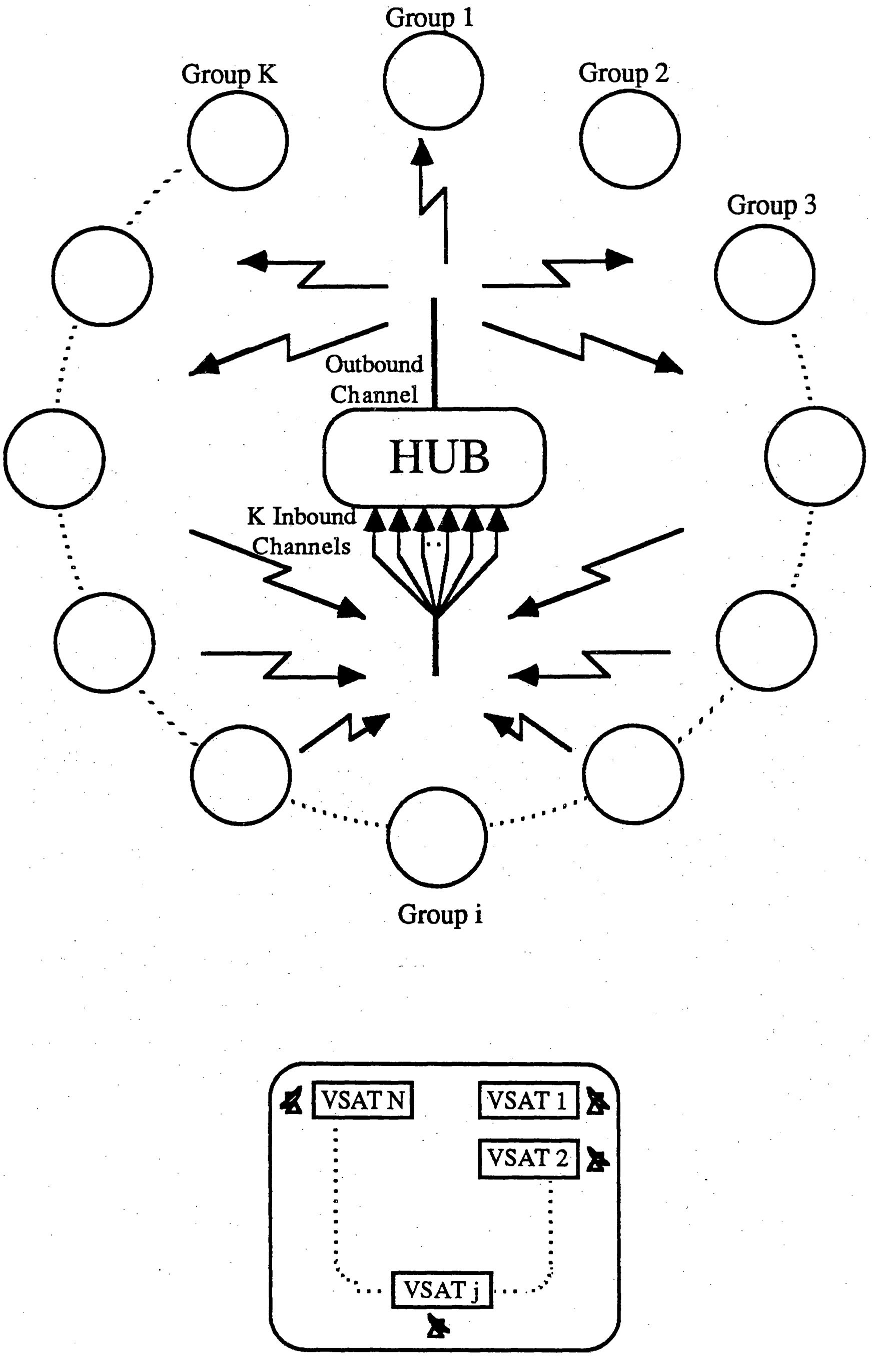
RESEARCH DESCRIPTION

The system considered is depicted in Figure 1. As can be seen, the system is a star VSAT network composed of a Hub station (primary station) and many remote VSATs (secondary stations). The Hub station is connected to host computers via terrestrial lines. Each remote VSAT is also connected to user's terminal devices. The network is composed of K VSAT groups, each group consists of N VSATs, as shown in Figure 1. The VSAT groups are connected to the Hub station via (K+1) satellite channels. Each VSAT group transmits packets to the Hub station via a distinct digital channel. This channel has a known bit rate R_b, and a known bit error rate P_b. The K group channels form the inbound lines. The Hub station, on the other hand, sends packets to all VSATs in the network using one single Asynchronous Time Division Multiplexing channel (ATDM). This ATDM channel is called the outbound line. It also has a known bit rate R'_b, and a bit error rate P'_b . In this study, two configurations (or architectures) will be considered: the single-hop star network, in which the VSATs communicate with the Hub station, and the double-hop star network, in which VSATs communicate with one another with routing being done via the Hub station.



Group detail

Figure 1. VSAT Star Network.

LIST OF SYMBOLS

$AK =$ Transmission time of an ACK frame sent by a VSAT to the Hub station.

$D_i =$ Poisson arrival distribution of outbound packets.

$\varepsilon =$ Error factor of an inbound packet.

$E(D_{II}) =$ Time access delay of an inbound packet.

$E(W_c) =$ Queueing delay of an ACK frame at the Hub station.

$E(W_{II}) =$ Queueing delay of an inbound polled packet station to a VSAT.

$E(W_{pc}) =$ Queueing delay of a polling command.

$K =$ Number of groups in the network.

$I =$ Packet length (Information field in an HDLC frame).

$I' =$ Overhead (control field in HDLC frame).

$L_i =$ Queue length distribution of outbound packets.

$\lambda_C =$ Arrival rate of ACK and control traffic at the Hub station.

$\lambda_{II} =$ Arrival rate of inbound traffic at one VSAT.

$\lambda_{IO} =$ Arrival rate of outbound traffic at the Hub station.

$M =$ Sequence number modulus $= 128$ (of satellite channel).

$N =$ Number of VSATS in one group.

$N_i =$ Retransmission distribution of outbound packets.

$P_b =$ Bit error rate of an inbound line.

$P'_b =$ Bit error rate of the outbound line.

$P_{II} =$ Probability that an inbound information frame is in error.

$P_{IO} =$ Probability that an outbound information frame is in error.

$P_{SI} =$ Probability that an inbound supervisory frame is in error.

$P_{SO} =$ Probability that an outbound supervisory frame is in error.

$R_b =$ Channel bit rate of an inbound line.

$R'_b =$ Channel bit rate of the outbound line.

$\rho_c =$ ACK and control frames traffic intensity at the Hub station.

$\rho_{II} =$ Inbound packets traffic intensity.

$\rho_{IO} =$ Outbound packets traffic intensity.

$T_{ack} =$ Transmission time of an ACK frame sent by the Hub station to a VSAT.

$T_{II} =$ Inbound information frame length, in seconds (s).

$T_{IO} =$ Outbound information frame length, in seconds (s).

$T_{oc} =$ Timeout interval of a polling command.

$T_{op} =$ Timeout interval of an inbound polled packet.

$T_p =$ One way propagation delay (VSAT-satellite-Hub) = 0.25 s.

$T_{SI} =$ Inbound supervisory frame length, in seconds (s).

$T_{SO} =$ Outbound supervisory frame length, in seconds (s).

$\bar{T}_c =$ Average cycle time of the polling system.

$\bar{T}_i =$ Average transmission time of packets waiting at VSATs in level i.

$\bar{T}_p =$ Transmission time of an inbound polled packet.

$T_{HV} =$ Time delay of a packet transmitted from the Hub station to a VSAT.

$T_{VH} =$ Time delay of a packet transmitted from a VSAT to the Hub station.

$T_{VV} =$ Time delay of a packet transmitted from a VSAT to another VSAT.

$\bar{W}_i =$ Average walk time to level i.

I. BACKGROUND AND LITERATURE REVIEW

1-1 *Introduction*

In recent years, a great deal of attention has been directed towards combining the study of data link controls and multiple access techniques for data communications. On one hand, data link controls constitute a layer of the international Open Systems Interconnection (OSI) reference model which is gradually being accepted as a standard in data communications. On the other hand, multiple access schemes are being used in the network layer of local area networks, cable networks, packet radio networks, and satellite networks. Traditionally, it is possible to carry out the study of these techniques separately, given the fact that Forward Error Correction, or FEC, is used for error recovery. FEC, however, presents several disadvantages when compared to other currently available techniques. These disadvantages are mainly an increase in the required bandwidth, a trade-off between reliability and simplicity, thus an apparent increase in cost. Other schemes, however, require that both studies be carried at the same time given the strong correlation between the two.

In this thesis, we combine a very popular data link control protocol, HDLC, with a multiple access scheme, roll-call polling. The error recovery technique used is go-back-N

Automatic Repeat Request (ARQ), and the transmission is achieved via a broadcasting Time Division Multiplexing channel.

To carry out the study of combining these techniques, a brief focus on each one of them separately is required. In this first chapter we provide a literature review for this research. A description and a formal definition of the various technical terms mentioned above will presented.

1-2 Time Division Multiplexing (TDM)

Several categories of multiaccess procedures for local area networks (LAN) and wide area networks (WAN) have been developed. The major ones are fixed assignment, random assignment, demand assignment and adaptive assignment. Fixed assignment methods are illustrated by frequency division multiple access (FDMA) and time division multiple access. The former uses frequency division multiplexing (FDM), hence subdivides the available bandwidth, and assigns each subdivision to one user. The latter, uses time division multiplexing, where the channel time is divided into slots which are organized into frames. The entire bandwidth is available to one user but, only during a designated time slot in consecutive frames. The assignment of the channel time to users has resulted in two types of TDM. The first type, carries out the sharing of the channel by making preassigned, sequential allocation of time slots to each user. This is a fixed assignment strategy, called synchronous time division multiplexing (STDM). The drawback of such a technique is the potential waste of channel time, when assigned to a user that has no input data, while there could be congestion of input data on others. The second type of TDM technique mainly provided to overcome the disadvantage of STDM is asynchronous time division multiplexing (ATDM), often referred to as statistical multiplexing. This method is tailored to accomodate input that fluctuates randomly between users [1]. For such input, utilization of the channel is significantly increased.

In systems employing ATDM, as shown in Figure 2, all users are multiplexed onto the same transmission line. A separate line from each user input is terminated in a line buffer that stores packets as they arrive. All line buffers, carrying the total input of the system are polled by a scanner that immediately transfers the existing packets in the line buffers into a central buffer where they join a queue for the shared channel. Since packets belonging to different users are interleaved in the queue and in the channel, they must contain some form of identification. This is done by including some overhead data that carries the packet destination. Unlike STDM, packets in ATDM do not need to be of the same length. ATDM is based on allocating the channel to a user that demands it, that is allocation is made only when a packet in the buffer is waiting to be transmitted. The allocation of the channel bandwidth is managed by the scanner or the concentrator. Its operation is equivalent to a single server queue. The channel is the server and the queue consists of the packets generated by the user's lines sharing the common transmission line. ATDM provides an efficient use of the channel even when every user line generates bursty traffic individually. The superiority of ATDM is based upon the phenomon of scaling in queueing theory. The average throughput of a single high-speed channel is better than that of multiple low-speed channels, even though the total bandwidth is the same in each case [2].

1-3 High-level Data Link Control (HDLC)

One of the main tasks of the data link control layer of any communication architecture is the correct and orderly delivery of packets between neighboring nodes in a network. For this purpose several protocols have been developed, and the most popular one is the High-level Data Link Control protocol (HDLC). Developed by the International Standards Organization,

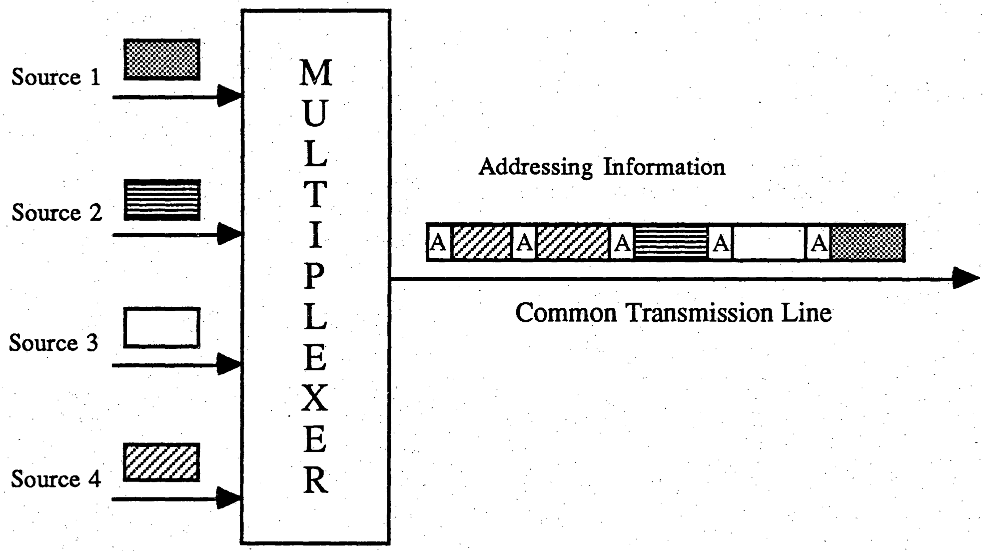


Figure 2. Asychronous Time Division Multiplexing (ATDM)

HDLC has become the international standard for data link control. As is the case for any link protocol, HDLC performs three major tasks: mainly establishing a connection (connect phase), correctly transferring data (data-transfer phase), and releasing the connection when data transfer is over (disconnect phase). For our purposes, we will focus on the data transfer phase.

In communications noisy channels often cause data to be received incorrectly at the receiving node. Several error recovery techniques have been devised for dealing with these problems. These techniques fall into two main categories: error detection, and error correction. HDLC is a bit-oriented protocol by design; it fits closely the go-back-N Automatic Repeat Request error correction scheme to be described later. In addition to error recovery, HDLC has to carry out several other functions such as synchronization and packet identification. For these purposes, several overhead control bits are added to each data packet. The combination of the overhead bits and the data packet is commonly referred to as a frame, and is what actually gets transmitted between neighboring nodes. The format of the HDLC frame is depicted in Figure 3. As can be seen in the figure, each frame consists of 48 overhead bits and a variable number of information bits, varying from zero to several thousand, hence the bit-oriented nature of the HDLC protocol. The flag bit pattern consists of a sequence of six ones, and this sequence can appear nowhere else in the frame. Consequently, a zero is inserted by the transmitter every time five consecutive ones are encountered in the frame bit stream. These zeros are automatically removed by the receiver. As shown by Figure 4, three types of frames are available to handle flow of information. These are I (information), S (supervisory) and U (unnumbered) frames. We shall limit our interest to information and supervisory frames. A description of the field assignments, and definitions will be provided shortly with the study of go-back-N error recovery scheme.

It is essential to note that the HDLC protocol can operate in any one of the following modes [3]:

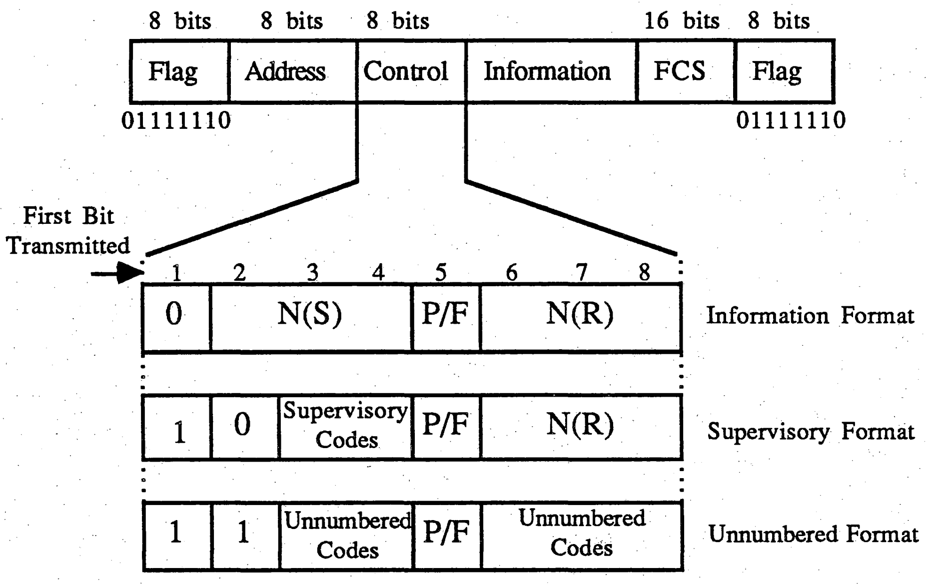


Figure 3. HDLC Frame Format.

	Control Field Bit Encoding								
Format	1	2	3	4	5	6	7	8	Commands/Responses
Information	0	-- N(S) --			*	-- N(R) --			I - Information
Supervisory	1	0	0	0	*	-- N(R) --			RR - Receive Ready
	1	0	0	1	*	-- N(R) --			REJ - Reject
	1	0	1	0	*	-- N(R) --			RNR - Receive Not Ready
	1	0	1	1	*	-- N(R) --			SREJ - Selective Reject

Figure 4. **HDLC Command/Responses.**

- Normal Response Mode (NRM). In this mode, a single primary station communicates with several secondary stations, using some multiple access scheme such as polling. The secondary stations are not allowed to initiate transmission unless commanded by the primary station.

- Asynchronous Response Mode (ARM). This mode is very similar to NRM with the exeption that a secondary station does not need the permission of the secondary station in order to initiate transmission.

- Asynchronous Balanced Mode (ABM). This mode is used for point-to-point transmission only. Both stations communicating are considered as equal parteners.

Given the nature of our network, we will be considering the NRM mode.

1-4 Go-Back-N Automatic Repeat Request (ARQ)

As mentioned earlier, two main approaches are available for handling errors that may occur during the transfer of information across a channel. The first method performs error correction and detection at the receiver side. To do so, several check or parity bits have to be added to the information bit stream to ensure a required level of error correction. An example of this technique is Forward Error Correction or FEC. This technique usually requires a higher channel bandwidth, complex hardwares to perform the correction tasks, and results in a strong dependence on the type of errors that the channel may experience. The second approach relies on the fact that the receiving side is able to report the failure or success of a transmission of a frame over a return (feedback) channel. This method of error detection with request for retransmission, also referred to as ARQ (Automatic Repeat Request), has been universally adopted for very high data integrity.

ARQ utilizes acknowledgements as its primary tool. Correctly received frames may be acknowledged by a special acknowledgement frame (ACK), while a negative acknowledgement frame (NAK) may be used to report an error condition. The acknowledgement can also be embedded in the control field of an information frame. Additionally, timeouts are used to avoid situations where the transmitting side is indefinitely awaiting a lost acknowledgement. Timeouts are usually treated as negative acknowledgements. Within the ARQ protocol, there are three procedures that are commonly used. These are the Stop-and-Wait protocol, the go-back-N protocol and the Selective Repeat protocol. We shall deal only with the go-back-N protocol since we will be using it in this study. An important concept used in go-back-N or continuous transmission is that of sliding windows [4]. Once a connection is established between a transmitting and a receiving station, a window of a fixed size M is set up at both sides. The window size is usually 8 for a terrestrial HDLC link and 128 for a satellite HDLC link. The application of the continuous ARQ error recovery scheme is as follow: the tramsnitting station has to maitain a "send state variable" or counter, while the receiving station keeps track of a "receive state variable". The sending station continuously transmits up to M-1 information frames without interruption. The sequence number of each frame which is the value of the "send state variable",is encoded in the N(S) portion of the HDLC frame control field. The "send state variable" is incremented after each frame is sent. Upon receipt of a frame, the receiving station compares the frame's sequence number with the value of the "receive state variable". If the frame is received correctly and both sequence numbers match, the "receive state variable" is incremented by one. It is placed in the N(R) field of the frame carrying the acknowledgement. This frame must include an RR command and be sent back to the transmitting station to complete transmission. If an error is detected, however, the current receive state counter value is encoded in the N(R) field of a negative acknowledgement frame. This time, the frame carries a REJ command and it is sent back to the transmitting station to indicate that the frame sent was received in error.

In practice, the receiving station does not acknowledge each individual frame separately, but waits instead until either the window is exhausted and all frames sent have been correctly received, in which case it replies with an acknowledgement frame, indicating that it is ready to receive the first frame in the next window, or until a frame is received incorrectly. In this latter case, a negative acknowledgement frame is returned, with the N(R) field set to the sequence number of the incorrectly received frame. Upon receipt of a positive acknowledgement frame, the transmitting station releases all window positions prior to the acknowledged sequence number. These positions can then be filled with new packets awaiting transmission. The transmission of new frames is, however, not disturbed and proceeds in a usual manner. When a negative acknowledgement frame is received, the preceding positions are again released for use by new packets. The "send state variable" in this case is reset to the sequence number of the rejected frame and all frames from that point on are retransmitted. The sliding window nomenclature comes from the fact that the windows are circular and continuously revolving. The various bit values of the HDLC frame will be discussed in a following section, after polling is introduced.

1-5 Roll-call Polling

The heart of any network is the protocol that permits the sharing of a common transmission medium among a number of users. The standard techniques used include random access, such as Aloha, and controlled access, mainly polling. A polling network is one where a central computer, usually acting as a master, performs the task of polling each station in the network in some kind of a predetermined order to provide access to the channel. Consequently, a station is not allowed access until polled, at which time it can use the full data rate of the available channel for transmission. In this context, we identify two modes of operation, roll-call polling and Hub polling. In Hub polling, the central computer initiates a

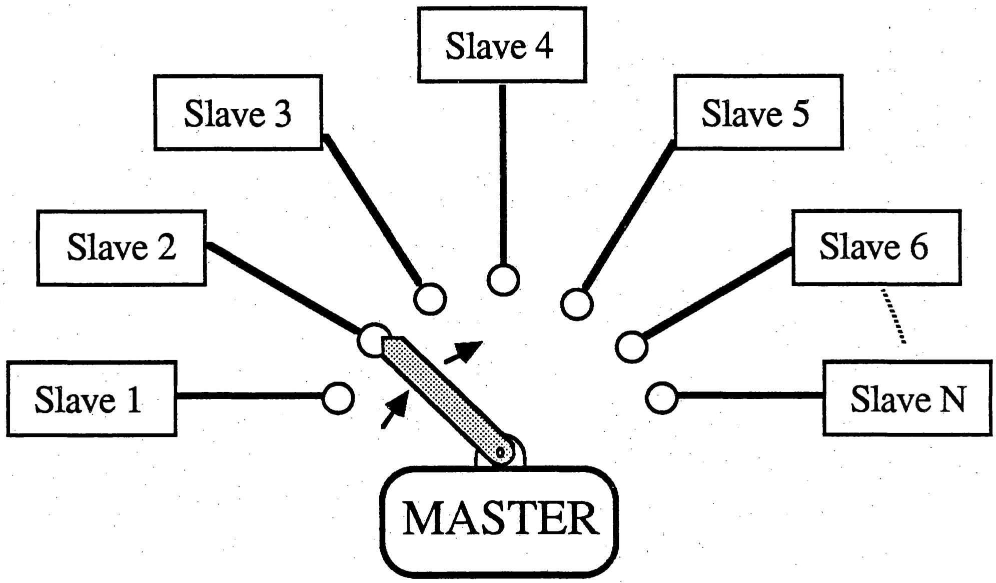


Figure 5. Roll-call Polling.

cycle by polling the first station. Once the station has completed its transmission, it issues a go-ahead information to the next station, which in turn passes the go-ahead information to the next station until all stations have been polled, and the cycle is repeated. Roll-call polling, as its name implies, requires the master station to sequentially poll every slave station in a preset order. The master grants permission to one slave station to transmit, and allows it a random time duration to sent their waiting packets. Once the polled station has completed transmission, the master moves on to polling the next secondary station, on its ordered list. This is repeated until all secondary stations have been given permission to transmit. This concept is illustrated by Figure 5.

Roll-call Polling in HDLC

HDLC is a bit-oriented protocol that provides the use of the same procedures among different applications. Applications that require different modes of operations and use different set of commands and responses to perform their activtities. Such applications include multipoint systems which differ from point-to-point systems. For multipoint configurations such as that of Figure 5, HDLC operates in the normal response mode, NRM. In this mode, as mentioned earlier, the primary station is in total control of the multipoint link.It is also responsible for maintaining a separate session with each station attached to the link.

The class of procedures defined for use with NRM, the unbalanced normal class (UNC), provides for the transmission of I frames, as well as ready to receive (RR) and not ready to receive (RNR) S frames, both as commands, from the primary station and responses from the secondary stations [3]. The concept of command performed in HDLC appears in the use of the address field, and the P/F bit, both shown in Figure 3. In NRM, recognizing only one primary station, and one or more secondary stations, the address of any transmitted frame is always that of the secondary station. To poll a station, the master transmits either an information or a supervisory frame with the address field set to that of the station being polled and the P bit set to 1. This is done over a common broadcasting channel. The corresponding station, which should be as for all slaves watching for this poll, responds once it recognizes its address by starting transmission of information frames. The F bit of the control field of the last information sent should be set to 1 to signal completion of transmission. A station granted permission to transmit can send up to M-1 frames. A polled station with nothing to transmit should simply answer back with a supervisory frame with the F bit set to 1. Once the master senses the F = 1 condition, it moves on to the next station The error recovery schemes with NRM are provided with two options. The first one includes the use of SREJ supervisory frames. The second one, used for continuous ARQ is the checkpointing mechanism, available with I, RR and RNR frames. A transmission of any of the above type of frames with the P bit set to 1 indicates which information frames have been acknowledged and where in the numbering sequence retransmission of I frames begin [5].

II. PERFORMANCE ANALYSIS OF THE NETWORK

2-1 Introduction

The previous chapter provided a general foundation for undrestanding the schemes and protocols utilized by the network components to communicate with each other. The goal of this chapter is to develop the method to analyse the given network, based on its topology.

The chapter starts with a general description of the communication components of the network. They are the outbound line and the inbound lines. A focus is then made on the utilization of these components in the actual application of the method developed for the network analysis. The chapter is concluded by providing a list of the characteristics of the model used throughout the rest of the thesis. These characteristics are needed to determine the analytic formulas that describe the performance of the network.

2-2 Communication Components

In order to develop the approach used for the analysis of the network, we need to take a closer look at the components provided to carry out the application of such a method. The characteristics of these components affect the design. The following section provides a formal definition of the communication components used, their major features and the functions they will perform. We start with the outbound line then move on to the inbound lines. The section is concluded by introducing a terminology "level" that will be used throughout the analysis.

2-2-1 The Outbound Line

The outbound line of the Hub station is a satellite broadcast time division multiplexing channel. First, being a satellite channel, the outbound line has a long propagation delay which strongly affects the data link control used. Then, the broadcast nature of the transmission allows many VSATs to hear the transmission in progress at the same time. Finally, the TDM characteristic of the channel permits the multiplexing of subchannels. Each subchannel will be serving a different VSAT.

The outbound line carries on two main functions. First, it is responsible for transmitting packets going from the Hub station to different VSATs in the network. Secondly, it controls the access to the inbound lines and acknowledge the inbound packets. The outbound data flow is illustrated by Figure 6.

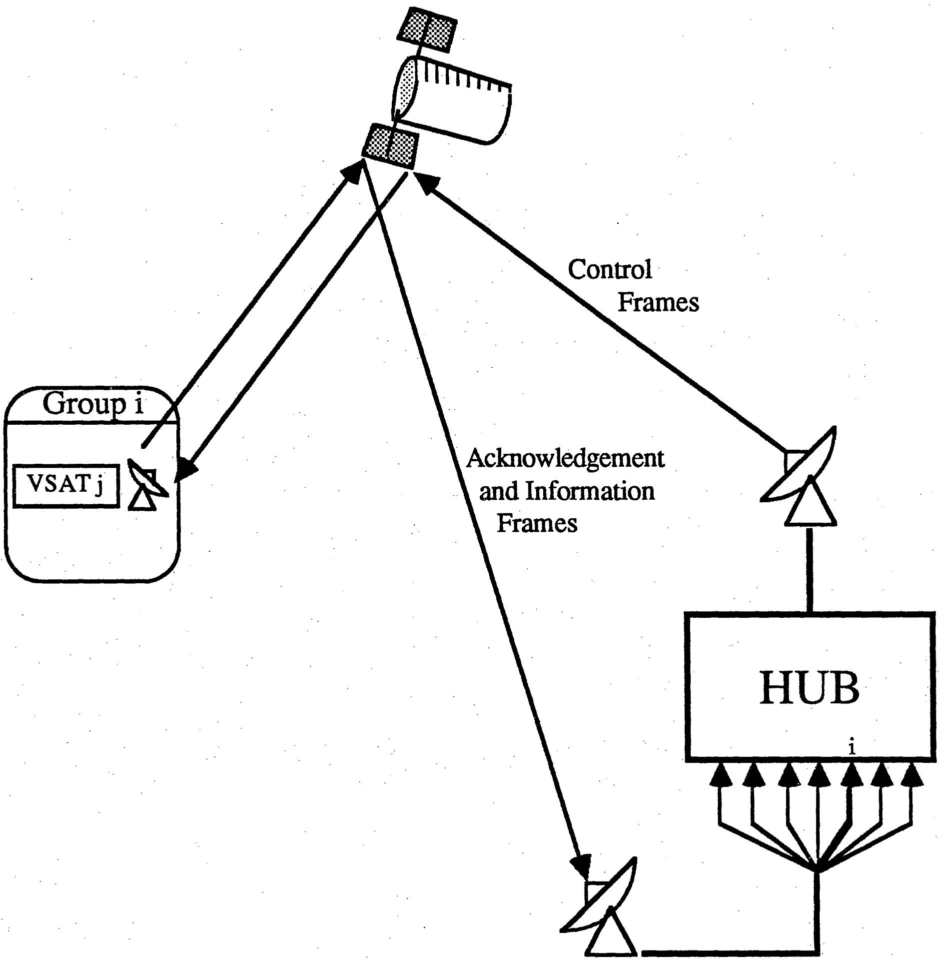


Figure 6. Outbound Traffic Data Flow.

2-2-2 The Inbound Lines

An inbound line is a digital channel that is shared by N VSATs of the same group. Performing in full-duplex operation where data flow is in both directions, an inbound line is responsible for two tasks. The first one is the transmission of the forward traffic, that is packets going from group VSATs to the Hub station. The second one, is the acknowledgement of the reverse traffic, that is packets going from the Hub station to group VSATs. The inbound data flow is depicted in Figure 7. Using roll call polling, every inbound line will be cycling among the N VSATs of its corresponding group. The VSATs of each group are served in a sequential order, starting with VSAT #1 and finishing with VSAT #N.

The Level Concept

We now introduce the term "level". We assume N levels in the network: Level# 1 through Level# N. The notation VSAT(i,j), $i=1....N$; $j=1....K$, denotes the i^{th} # VSAT of the j^{th} # group.

LEVEL # i = {VSAT(i,1),VSAT(i,2),.........,VSAT(i,K)}

The level concept is depicted in Figure 8.

2-3 Approach Description

The proposed protocols and the communication components just described have been lashed together to provide the approach derived for the analysis of the network. The method developed is based on two main factors: fairness, to enforce equal utilization of the netwok resources, that is, the use of the outbound line and the inbound lines; and performance: to obtain a high throughput and minimize the average time delay that a packet spends in the

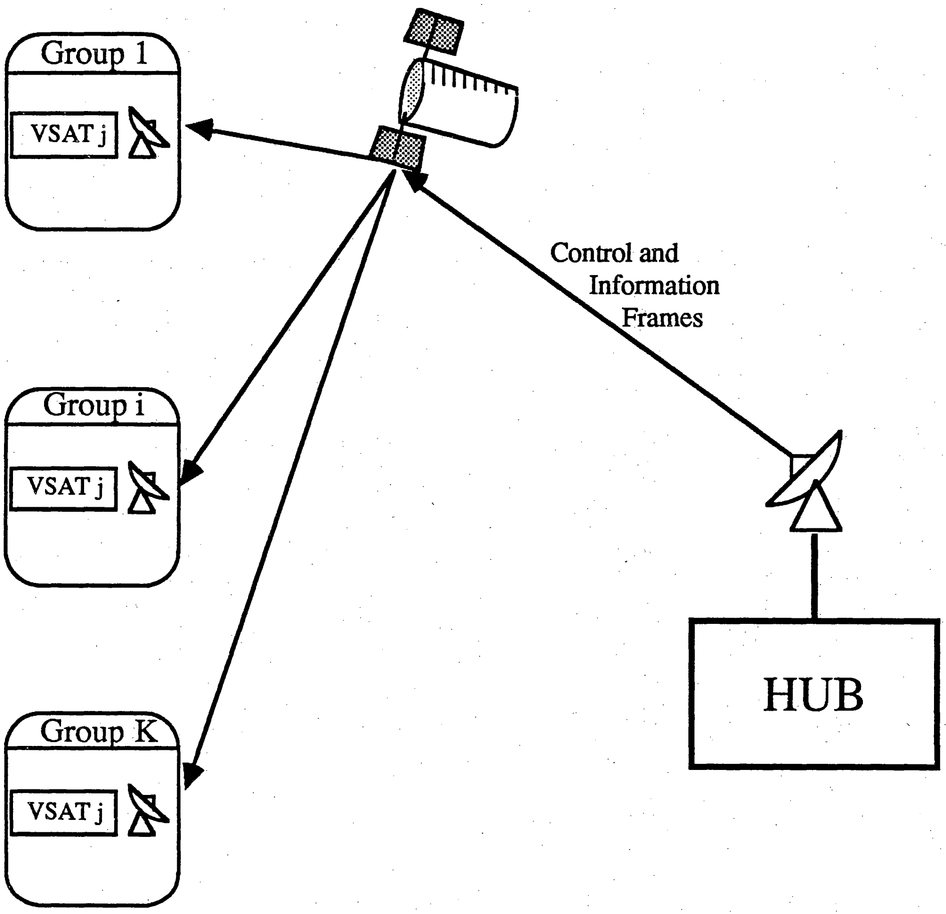


Figure 7. Inbound Traffic Data Flow.

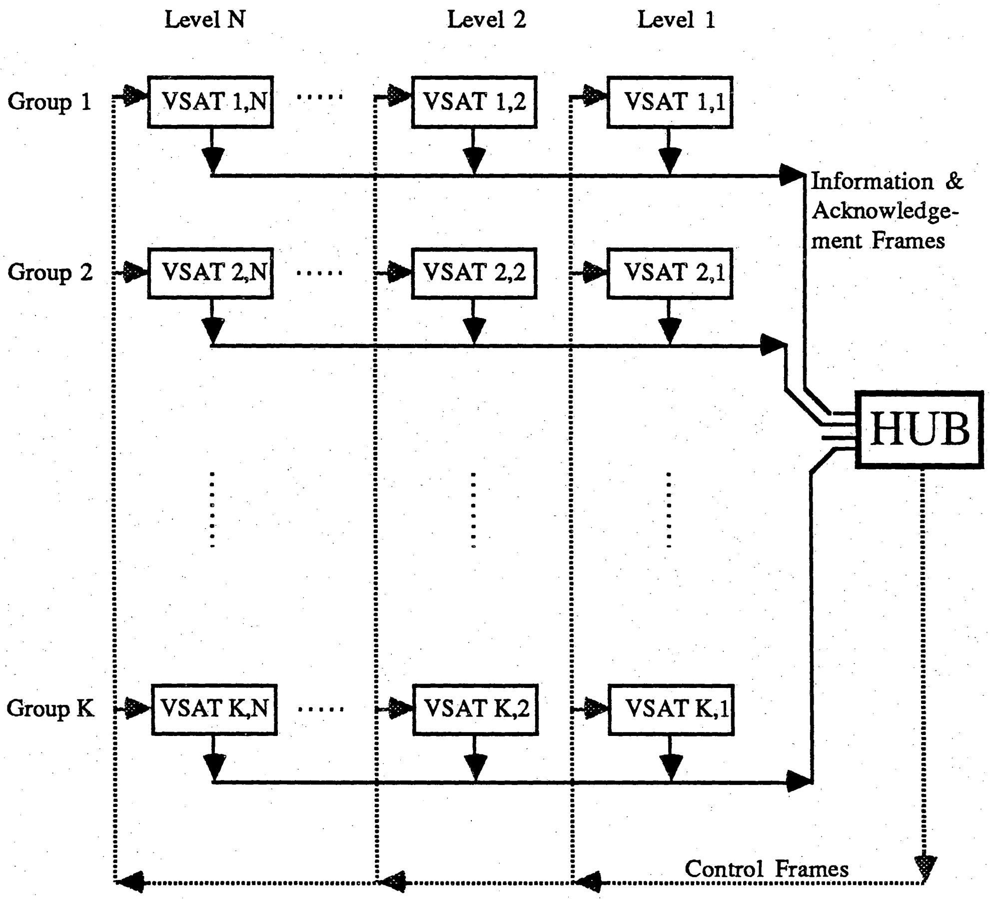


Figure 8. Level Concept.

system. Two types of transmission could take place. A packet may be transmitted from VSAT-to-Hub in a single-hop architecture in which case the average time delay of this packet is equal to the inbound delay time. Or, a packet may be transmitted from VSAT-to-VSAT in a double-hop mode. In this case, with the routing being done through the Hub station, the average time delay of a transmitted packet is equal to the sum of the inbound delay time and the outbound delay time . In the following dicussion these two types of transmissions are described.

First, for the transmission of the outbound traffic, packets going to all VSATs are connected to a multiplexer. These packets are served in order of arrival, using the full capacity of the outbound common transmission line. In order to distinguish among packets going to different VSATs sharing the line, addressing information must accompany these packets. The address field of the HDLC information frame carrying the packet is that of the corresponding receiving VSAT. The acknowledgement of the outbound frames is contained in the frames sent by the VSATs, once granted access of the inbound lines. These acknowledgements may be carried in a separate HDLC supervisory frame, in which case a polled VSAT has received packets from the Hub station but has no data packets to send in return, or they could be piggy-backed in the HDLC information frames transmitted by a polled VSAT that has waiting packets to send.

For the transmission of the inbound traffic, the Hub station uses roll call polling scheme to provide access to VSATs. The strategy to perform this operation is based on two major facts. First, VSATs from different groups use different inbound lines to transmit their packets. This implies, with the "level" notation introduced earlier, that VSATs of one level may access the N available inbound lines at the same time without any collision between packets. Secondly, given the broadcast nature of the outbound line, a polling command addressing one level may be heard by all VSATs of that level. The application of the access scheme is as follows. The Hub station transmits a ready-to-receive (RR) HDLC supervisory frame with the polling bit P set to 1 and addresses it to one level. All VSATs of the addressed level recognize the polling command and respond to it simultaneously using different inbound lines. An interrogated VSAT with waiting packets to send is granted access of its group inbound line until it completes transmission and error recovery of its packets. The Hub station acknowledges the packets of each sending VSAT separately. The acknowledgement is carried out in RR HDLC supervisory frames. To acknowledge the packets of one VSAT, the Hub station transmits an acknowledgement frame and addresses it to the corresponding VSAT. The ACK frame has its P bit set to 1, giving permission to the addressed VSAT to transmit. Also, the ACK frame provides the sequence number of the next packet expected to be sent by the VSAT. The end of a transmission period is indicated by a correct transmission of an HDLC information frame with its F bit set to 1. Once all polled VSATs of the same level have completed transmission, the Hub station moves on to poll the next level. This is repeated until all levels of the network have been polled, in which case all VSATs would have been given permission to transmit their packets by then. Figure 9 illustrates how a VSAT operates in the network.

Finally, for the the transmission of both outbound and inbound traffic, we get a system where the outbound line operates as a server that cycles among queues of packets going to different VSATs. The server starts by serving packets going to VSAT (1,1) and terminates by serving packets going to VSAT (K,N). If a VSAT queue is empty, the outbound line moves on to serving the queue of the packets going to the next VSAT, without wasting the channel time. While cycling among the VSAT queues, the server is interrupted to control the access on the inbound lines and acknowledge their traffic. The command and acknowledgement frames have a nompreemptive priority over the outbound packets. As for the inbound lines, they will be carrying the packets and ACK frames transmitted by the VSATs of their assigned group. With the use of roll call polling, VSATs are served starting with VSATs in level 1 and finishing with VSATs in level N. The Hub Operations are depicted in Figures 10,11 and 12. The states of the outbound line and the inbound line are depicted in Figures 13, 14.

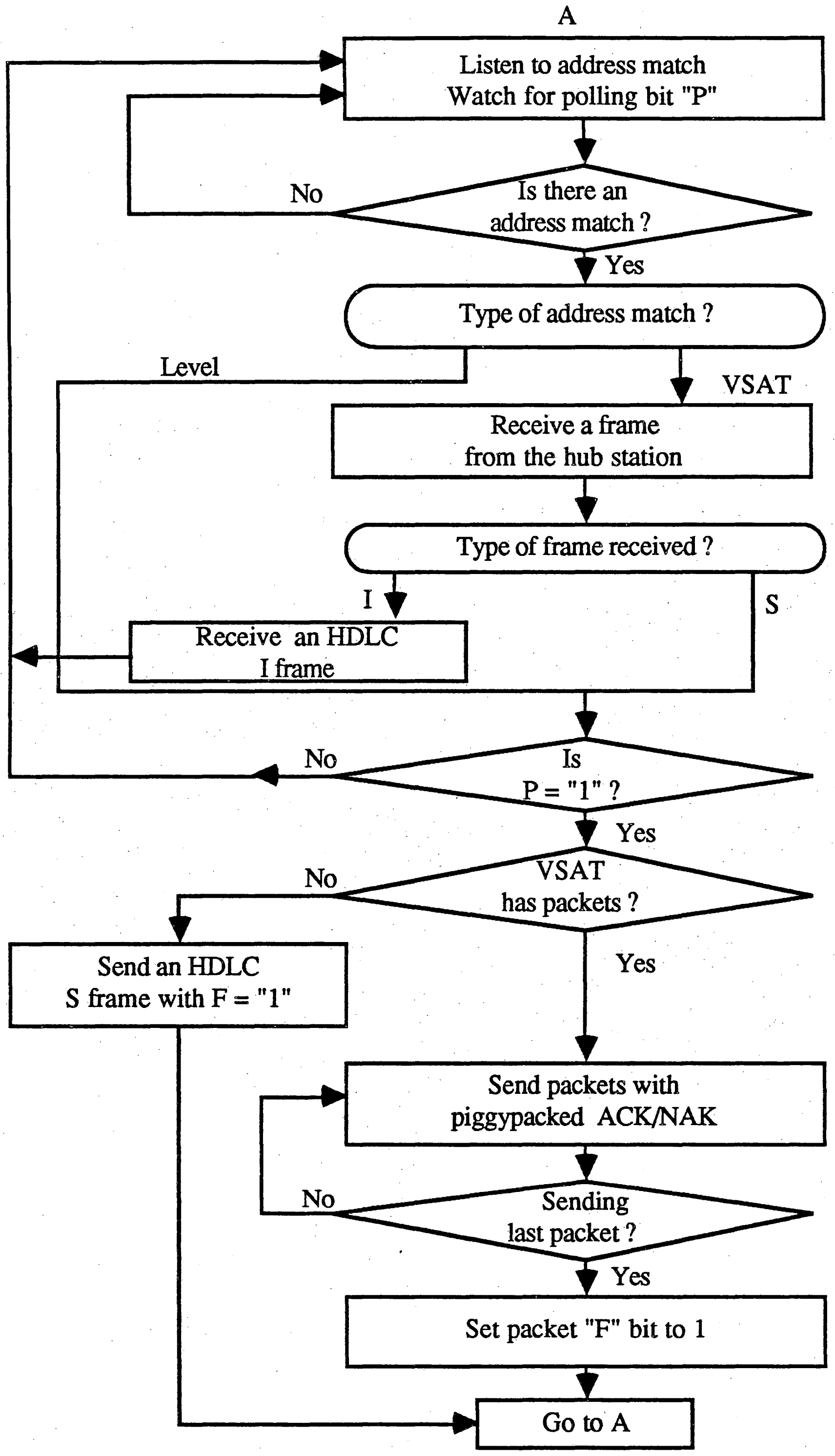


Figure 9. VSAT operations.

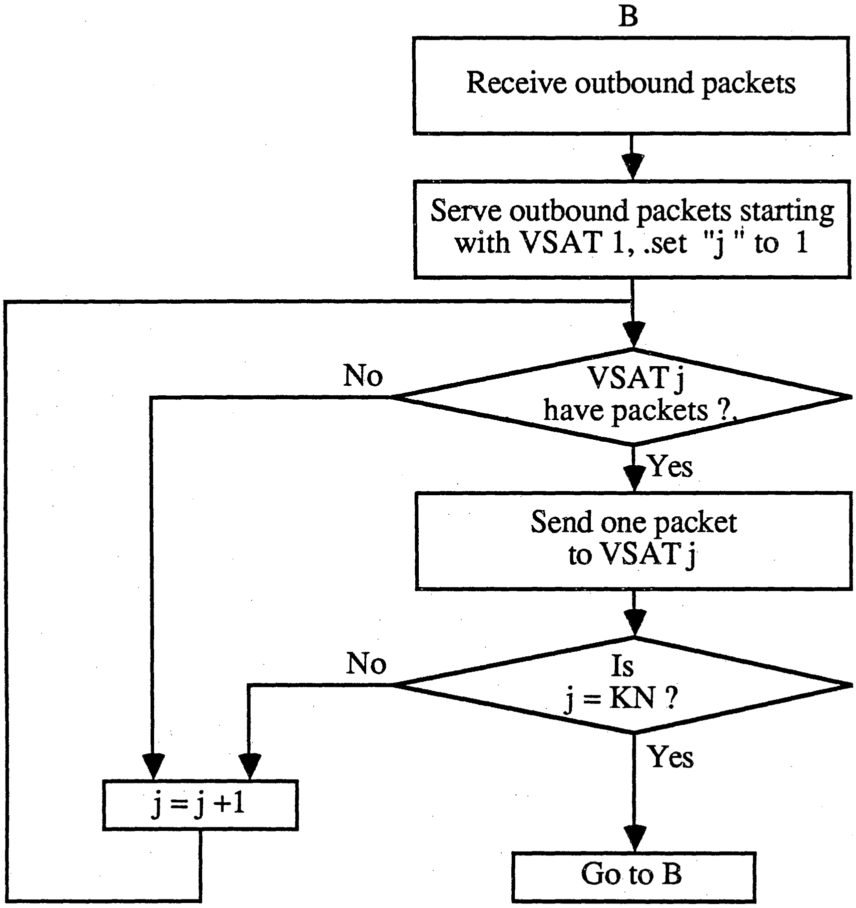


Figure 10. Hub Outbound Operations.

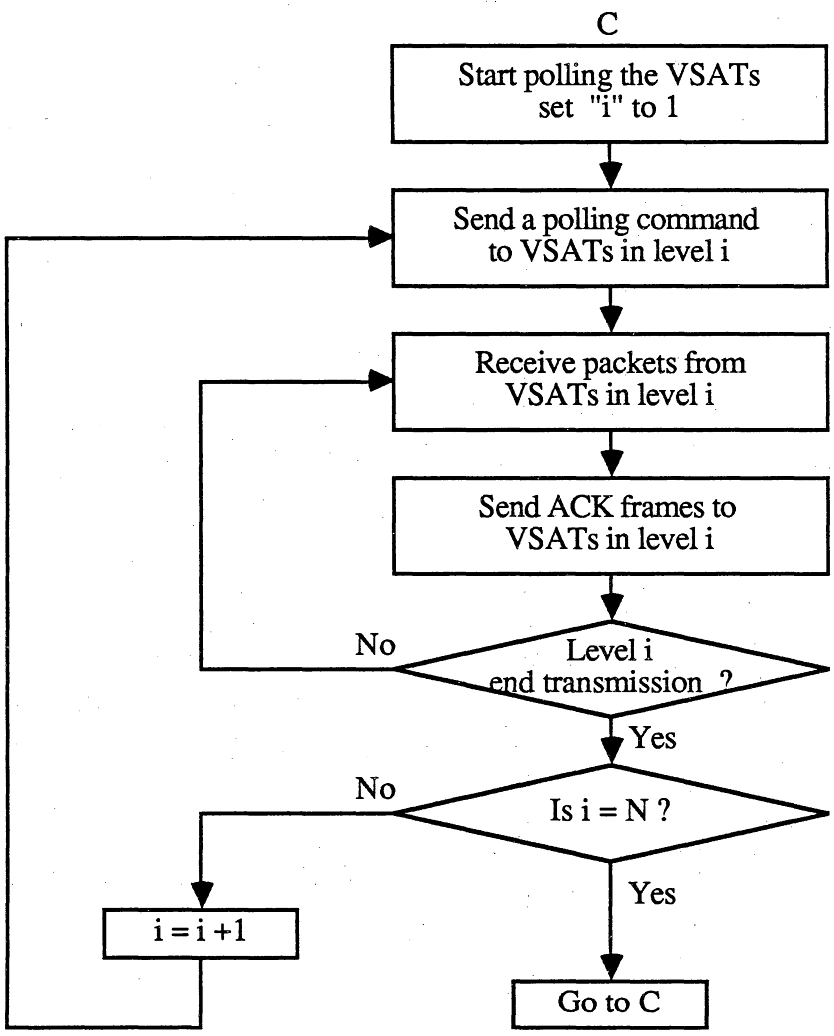


Figure 11. Hub Inbound Operations.

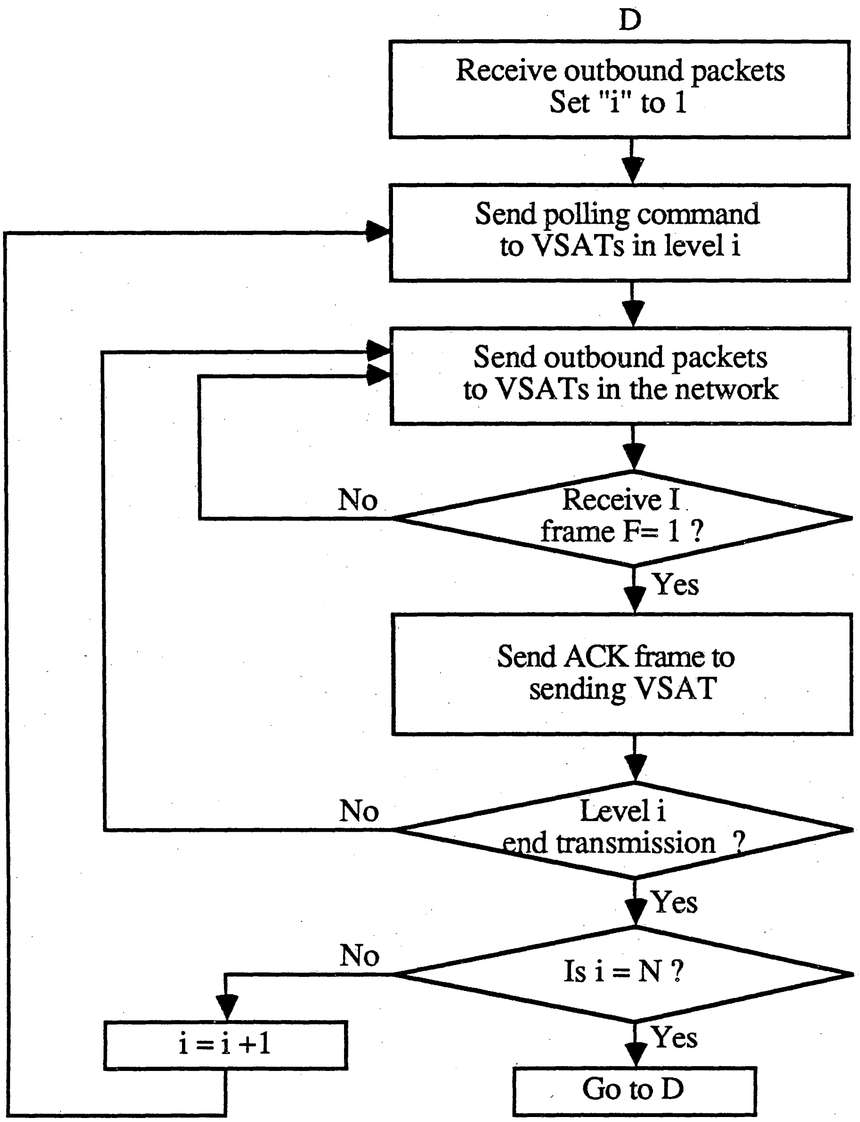


Figure 12. Hub combined Operations.

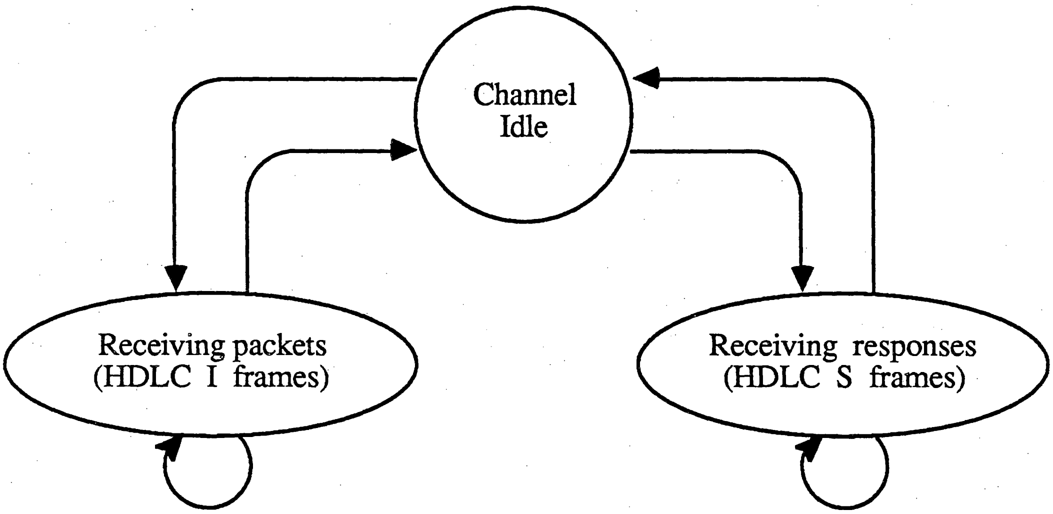


Figure 13. Inbound Channel Operations.

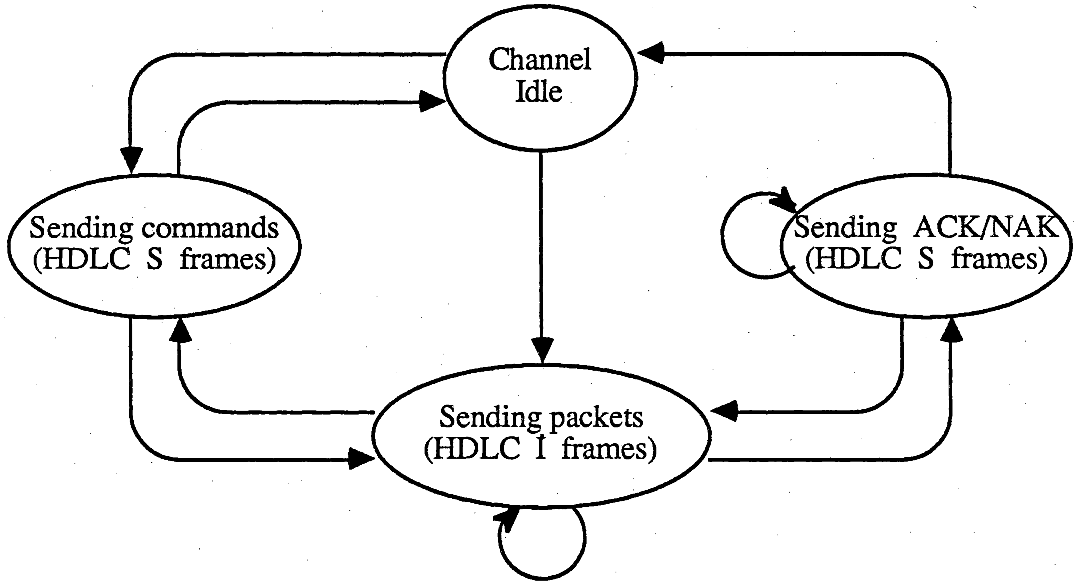


Figure 14. Outbound Channel Operations.

2-4 Characteristics of the Model

The following section provides a summary of the characteritics of the model utilized throughout the analysis.

- Two kinds of frames are used, an HDLC information frame of length $(l + l')$ bits and an HDLC supervisory frame of length l' bits.
- All data packets are of the same length. The terms packet and HDLC information frame are used interchangeably.
- We assume a Poisson arrival process of packets at every VSAT with an equal arrival rate λ_{II}.
- We assume an overall Poisson arrival process of packets at the Hub station with an arrival rate λ_{IO}.
- We assume a Poisson arrival process of ACK frames at the Hub station with an arrival rate λ_c.
- Two queues are being served by the outbound line. The first is a queue of first priority for control and ACK of the inbound traffic. The second is a multiplexed stream of data packets going to the VSATs.
- The polling commands as well as the ACK frames of the inbound traffic are both HDLC supervisory frames.

III. INBOUND LINE TIME DELAY

3-1 *Introduction*

The objective of this chapter is to determine the performance of the roll-call polling system of the network. As in any polling system, the emphasis is on the calculation of access delay as experienced by packets at each of the VSATs. This delay is measured from the time a packet arrives at one of the VSATs that share a common inbound line to the time the packet begins transmission. This time, as found by Schwartz [6], is comparable to the waiting time in queueing calculations. To find the inbound delay time, one must add the average transmission time of a packet to the access delay. The average transmission time of a packet is the virtual transmission time of an HDLC information frame using a go-back-N protocol for error recovery.

This study, as mentioned in the earlier chapter, is restricted to Poisson arrivals at the VSATs. The average access delay is then found to be made up of two terms. One term is precisely the M/D/1 wait time, since I-frames are assumed to have fixed length. The other, unique to polling systems, is given by one half the cycle time. The cycle time is the time required to complete a poll of all of the (KN) VSATs. In our case, this is equivalent to the time required to poll the N levels of the network. In every level poll, we complete a poll of K VSATs sharing different inbound lines.

At low traffic, the average time that an arriving packet has to wait before transmission is intuitively equal to half the cycle time. As traffic increases, additional time delay is incured due to the time required to serve packets that have arrived earlier. This is expressed by the M/D/1 waiting time.

An attempt to complete a time delay analysis for a polling system where the traffic is random is quite complex, since the delays at all VSATs are interdependent. Thus, the delay during a given polling cycle depends on the packets waiting to be transmitted at every VSAT. In the analysis of the time delay of this polling system, the average traffic case is assumed at each VSAT. The average access delay, thus the average inbound delay time is calculated. In the following sections we carry out the analysis of the average access time, starting with the average cycle time, then the M/D/1 wait time.

3-2 Cycle Time

The time required to cycle once among the N levels is made up of two components. One is the time required to transfer permission to transmit from VSATs in one level to VSATs in another level called "walk time". The other is the time to actually transmit packets once VSATs of a level have been given permission to do so. The first component is the time required for a correct transmission of the polling command to VSATs of each level, and the synchronization time of those VSATs listening on the outbound line to recognize their polling command and take action to start transmitting, and the propagation delay required by the polling signal to physically propagate to the VSATS in the polled level. In our case, this propagation delay will be a part of the virtual transmission time of establishing a poll. The second component is the transmission time of packets waiting to be served. That is equal to the total number of packets in the polled VSATs queues multiplied by the average transmission time of a packet. In summary, we have

$$\overline{T}_C = \sum_{i=1}^{N} \overline{W}_i + \sum_{i=1}^{N} \overline{T}_i \qquad (3-1)$$

where

$\overline{T}_c$ = average cycle time.

$\overline{T}_i$ = average transmission time of packets waiting at VSATs in level # i.

$\overline{W}_i$ = average walk time to level # i.

3-2-1 Walk Time

The walk time is made up of the transmission time of a polling command, its propagation delay, and the synchronization time of a VSAT to a poll. The propagation delay of the polling command is included in the average time delay of a correct transmission of such command. Given the long propagation delay of the outbound line, the synchronization time is therefore neglected. The polling command is an HDLC supervisory frame and hence of constant length. The average transmission time of such a frame to each of the VSATs in any level is the same. Thus the average walk time of the VSATs in any level is the same, and is equal to $\overline{W}$. Thus the total walk time L is given by

$$L = \sum_{i=1}^{N} \overline{W} \qquad (3-2)$$

To find $\overline{W}$, a description of the polling command and its transmission is provided, as follows. For any VSAT to start sending its packets, during its corresponding transmission time, the polling command must be correctly received by all VSATs of the level polled. If the command frame is received in error by any addressed VSAT, it has to be resent to all VSATs of that level. Any packet sent as a response to a polling command that has not been received correctly by all VSATs of the polled level is ignored. This implies that the polling command is retransmitted until it is positively acknowledged by all polled VSATs. The retransmission of the polling command is driven by a timeout function. This timeout function is activated at the Hub station every time it sends out a polling frame. The polling frame is retransmitted if the corresponding timeout expires before its acknowledgements from all polled VSATs arrive. The choice for the duration of the timeout is based on the type of response a polled VSAT sends back as a reply to the polling command. Once polled, a VSAT with no waiting packets to transmit responds to the poll by sending a supervisory frame with its F bit set to 1, indicating an empty queue of data packets. On the other hand, a polled VSAT that has packets to send replies by transmitting the first packet waiting in the queue. With the assumption that for the first time during a polling cycle, a polled VSAT has at least one packet waiting to be transmitted, a reply to a correct polling command frame is indicated by a transmission of an HDLC information frame. We define the timeout period to be equal to the propagation delay of the polling command (Hub-to-VSAT delay), plus the frame length in units of time of an inbound packet, plus the propagation delay of the packet (VSAT-to-Hub delay).

$$T_{oc} = 2T_p + T_{II} \qquad (3-3)$$

Where

$$T_{II} = \frac{I + I'}{R_b} \qquad (3-4)$$

Transmission Time of a Polling Command

In this section we provide a description of the type of transmission for a polling command. Then, we compute the duration of this transmission. We start by listing the following "facts". First, in order to start polling VSATs in the current level, packets sent by all VSATs of the previous polled level had to be acknowledged. Then, in order to generate the polling command for the next level, the Hub station must wait for VSATs in the current level to finish transmission. Thus, the ACK and control queue is empty prior to the arrival of the polling command frame. As mentioned earlier, a polling command frame has the nonpreemptive first priority over the data packets being served by the outbound line. Assuming that the outbound line always transmits a frame, independently of the queueing discipline under consideration, then according to [3] the residual transmission time of any arriving frame must be the same as if frames of both classes, commands and information, are transmitted with the same priority. This residual time is equal to the average wait time of an M/G/1 queue with service time equal to the length of the frame being transmitted. Having only the frames of the second class present when the polling command frame arrives at the outbound line, the M/G/1 queue reduces to an M/D/1 queue with service time equal to the outbound information frame length, in units of time, T_{IO}. The case considered here is then that of a queue of "fixed service time" , with an arrival rate of λ_{IO}. The average wait time of a polling command, $E(W_{pc})$, is the mean residual time of the outbound data traffic given by [7]

$$\begin{aligned} E(W_{pc}) &= \frac{\lambda_{IO}}{2}\overline{T^2_{IO}} \\ &= \frac{\lambda_{IO}}{2}T^2_{IO} \end{aligned} \tag{3-5}$$

where

$$T_{IO} = \frac{l' + l}{R'_b} \tag{3-6}$$

In the following discussion, the computation of the polling command transmission time is represented. The transmission time of a polling command is that of an HDLC supervisory frame, of length T_{SO} . This frame is sent to K VSATs. It is transmitted on an ATDM outbound channel of error bit rate P'_b. A positive acknowledgement of this frame is equivalent to a receipt of an information frame of length T_{II} from each polled VSAT. Every information frame is transmitted on a different inbound channel with error bit rate P_b. Finally, the average waiting time of a polling command frame, is given by Eq. (3-5).

The probability that both the polling command frame and the reply information frame are correct, provided that the two events are independent, is equal to :

$$P_p = (1 - P_b)^{(l+l')}(1 - P'_b)^{(l')} \qquad (3-7)$$

where

$(1 - P_b)^{(l+l')}$ = The probability that the reply I- frame is correct.

and

$(1 - P'_b)^{(l')}$ = The probability that the polling S-frame is correct.

The probability that all K polled VSATs reply positively to the command is

$$P_{pK} = P_p^K \qquad (3-8)$$

Assume that the outbound line is able to serve the polling command in exactly m attempts. If the first (m-1) attempts are unsuccesful and the mth is succesful [8], then the probability of this event is:

$$P(m) = P_{pk}[(1 - P_{pk})^{m-1}] \qquad (3-9)$$

The time to complete transmission of the polling command in m attempts is:

$$T(m) = (m - 1)(T_{SO} + T_{oc} + E(W_{pc})) + (T_{SO} + T_p + E(W_{pc})) \qquad (3-10)$$

where

$$T_{SO} = \frac{l'}{R'_b} \qquad (3-11)$$

and T_{oc} , $E(W_{pc})$ are given by Eq. (3-3), and Eq. (3-5),respectively. $T_{so} + T_p + E(W_{pc})$ is the average time to transmit the polling command, under error free condition. This corredponds to the mth attempt. To evaluate the average time to poll one level, or the walk time level, one substitutes P(m) from Equation (3-9), averages T(m) and gets

$$\overline{W} = \sum_{m=1}^{\infty} P_{pk}[(1-P_{pk})^{m-1}][(m-1)(T_{oc}+T_{SO}+E(W_{pc}))+T_{SO}+T_p+E(W_{pc})] \qquad (3-12)$$

$\overline{W} = W_1 + W_2$, where

$$W_1 = \sum_{m=1}^{\infty} P_{pk}[(1-P_{pk})^{m-1}]((m-1)(T_{oc}+T_{SO}+E(W_{pc})))$$

At m = 1, $W_1 = 0$. Thus, one can write W_1 as

$$W_1 = \frac{1}{(1-P_{pk})} \sum_{m=2}^{\infty} P_{pk}[(1-P_{pk})^{m}]((m-1)(T_{oc}+T_{SO}+E(W_{pc})))$$

Since,

$$\sum_{m=2}^{\infty} P_{pk}[(1-P_{pk})^{m}] = \frac{(1-P_{pk})^2}{P_{pk}}$$

Then,

$$W_1 = \frac{(1-P_{pk})}{P_{pk}}(T_{oc}+T_{SO}+E(W_{pc}))$$

and

$$W_2 = \sum_{m=1}^{\infty} P_{pk}[(1 - P_{pk})^{m-1}](T_{SO} + T_p + E(W_{pc}))$$

Since,

$$\sum_{m=1}^{\infty}[(1 - P_{pk})^{m-1}] = (1 - P_{pk})^0 + (1 - P_{pk})^1 + (1 - P_{pk}) + \ldots$$

$$= \frac{1}{1 - (1 - P_{pk})} = \frac{1}{P_{pk}}$$

Then,

$$W_2 = (T_{SO} + T_p + E(W_{pc}))$$

Finally, one gets

$$\overline{W} = (1 - P_p^k)\,\frac{(T_{oc} + T_{SO} + E(W_{pc}))}{P_p^k} + (E(W_{pc}) + T_p + T_{SO}) \qquad (3-13)$$

Returning to Eq. (3-2) and substituting for $\overline{W}$, one finds the total walk time for the network to be

$$L = N(1 - P_p^k)\,\frac{(T_{oc} + T_{SO} + E(W_{pc}))}{P_p^k} + (E(W_{pc}) + T_p + T_{SO}) \qquad (3-14)$$

The transmission of a polling command is illustrated by Figure 15.

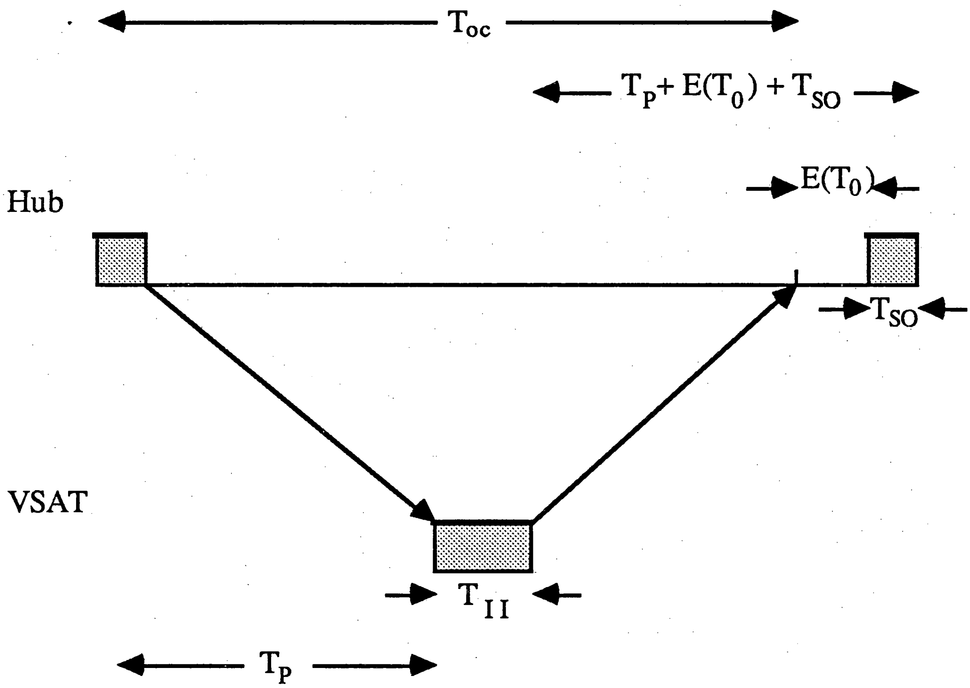


Figure 15. Transmission of a Polling Command.

3-2-2 Transmission Time

The average transmission time, $\overline{T}_I$, is the time to serve the packets waiting to be transmitted at the VSATs in level i once polled. In this analysis, the model considered is one in which buffers at all VSATs are infinite and all packets waiting are read out when the polling signal is received. All VSATs have the same arrival rate λ_{II}. The average wait time of a VSAT to begin transmission of its packets is given by the average cycle time of the network, $\overline{T}_c$. The average number of packets waiting to be transmitted at any VSAT when it is given permission to transmit is equal to $\lambda_{II}\overline{T}_c$. Now let $\overline{T}_p$ be the average transmission time of a polled packet. $\overline{T}_p$ is the virtual transmission time of an HDLC information frame using go-back-N ARQ for error recovery. The time required to transmit the waiting packets out on an inbound line is then

$$\overline{T}_I = \lambda_{II}\overline{T}_c\overline{T}_p \qquad (3-15)$$

Where $\overline{T}_c$ is given by Eq.(3-1) and $\overline{T}_p$ is computed in the following section.

Transmission time of a polled packet

The transmission time of a polled packet is equal to the virtual transmission time of an HDLC information frame using the go-back-N protocol for error recovery. In this protocol, with the P/F checkpointing mechanism available for HDLC-NRM operations, timeout strategy will always be used for determining an error occurence. With the polled VSAT being allowed to send only up to a maximum number of (M-1) packets, the sequence number N(S) of a polled information frame is limited by [9]

$$0 \quad \leq \quad N(S) \quad \leq \quad (M-1) \qquad (3-16)$$

Once exhausting its window size, a polled VSAT has to stop transmitting until it receives acknowledgement for the packets sent. A timeout function is activated at a VSAT after sending the first I-frame in the sequence. This timeout expires once a VSAT exhausts its window size. For an arbitrary polled packet, this takes place after a period of time

$$T_{op} = (M - 2)T_{II} \qquad (3-17)$$

Where T_{II} is given by Eq.(3-4) and $M = 128$.

In the go-back-N protocol, frames may be transmitted continuously without waiting for acknowledgements. This allows the interval between two successive transmissions to be equal to one frame length, in units of time. A frame may be retransmitted due to an error on the inbound line (I-frame), or an error on the outbound line (ACK frame of the packet). Due to the nature of the ARQ retransmission protocol, a frame may also be retransmitted due to an error that took place in a previous frame.

The probability that a polled packet is received in error is given by

$$P_{II} = 1 - (1 - P_b)^{(l+l')} \qquad (3-18)$$

According to Schwartz [3], the average time for a correct transmission to be received is

$$\bar{T}_p = T_{II} + (1 - P_{II}) \sum_{i=1}^{\infty} i P_{II}^{\,i} T_{Tp} \qquad (3-19)$$

where T_{Tp} is the time to transmit a repeat. This is illustrated by Figure 16.

Equation (3-19) indicates that to get to the i^{th} retry, a packet must have been retransmitted due to an error i times. The probability of receiving it correctly by the Hub is $(1 - P_{II})$. The probability that the the polled packet is retransmitted due to an error in its ACK frame will appear in T_{Tp} , where

$$T_{Tp} = T_{op} + T_{II} + T_{ack} \tag{3-20}$$

We will now derive T_{ack}. An acknowledgement frame is generated every time an HDLC information frame, with its F bit set to 1 is sent by any of the VSATs. This ACK frame joins a queue, with Poisson arrival rateλ_c, and is of first nonpreemptive priority, as mentioned earlier. Two major factors are to be taken into consideration during the transmission of an ACK frame: (1)the waiting time delay that an ACK frame has to experience for its transmissions and (2) since the reply to a correct ACK is indicated by a frame transmission that is exposed to a possible loss, a timeout function must be activated at the Hub station for each ACK frame sent out. This timeout function guarantees the retransmission of the ACK frame, every time its timeout period has expired.

The waiting time delay is the average wait time of the first-priority, control and ACK class of frames. In this class, the ACK frames queue up as in an M/G/1 system of a single class, seeing just themselves, only for the additional residual time of the frame being in service. This frame might be in their own class, or in the second class of the outbound data packets. This residual time is denoted by $E(T_c)$, and is equal to [7]

$$\begin{aligned} E(T_c) &= \frac{\lambda_{IO}}{2}\overline{T^2_{IO}} + \frac{\lambda_c}{2}\overline{T^2_{SO}} \\ &= \frac{\lambda_{IO}}{2}T^2_{IO} + \frac{\lambda_c}{2}T^2_{SO} \end{aligned} \tag{3-21}$$

Defining the intensity of the ACK, and control traffic to be

$$\rho_c = \lambda_c T_{SO} \tag{3-22}$$

The average wait time is then given by [7]

$$E(W_c) = \frac{E(T_c)}{(1-\rho_c)} \tag{3-23}$$

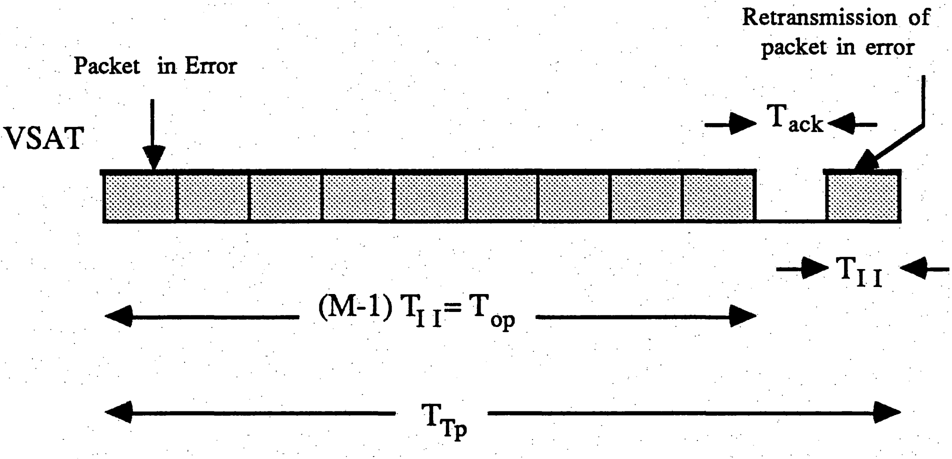


Figure 16. Transmission of Polled Packets.

As for the the timeout period, the same as that of a polling command, it is given by Eq. (3-3). The choice of this timeout duration is based on the knowledge that a positive receipt of the ACK frame is indicated by a transmission of an information frame if the polled VSAT has waiting packets to send, or by a supervisory frame with its F bit set to 1. In this latter case, the Hub station assumes that the VSAT polled has completed its transmission and error recovery of its packets. The transmission of an ACK frame is illustrated by Figure 17. The probability that an ACK frame is in error is

$$P_{SO} = 1 - (1 - P'_{b})^{l'} \qquad (3-24)$$

The average time of a correct transmission of an ACK frame is

$$\begin{aligned} T_{ack} &= T_{Tc} + (1 - P_{SO}) \sum_{i=1}^{\infty} i P_{SO}{}^{i} T_{Tc} \\ &= \frac{T_{Tc}}{(1 - P_{SO})} \end{aligned} \qquad (3-25)$$

with

$$T_{Tc} = T_{oc} + T_{SO} + E(W_c)$$

Substituting T_{oc}, and $E(W_c)$ from Eq. (3-3) and Eq. (3-23), respectively, one finds

$$T_{Tc} = T_{II} + 2T_p + T_{SO} + \frac{1}{(1 - \lambda_c T_{SO})} [\lambda_{IO} \frac{T_{IO}^2}{2} + \lambda_c \frac{T_{SO}^2}{2}] \qquad (3-26)$$

Returning to Eq. (3-19) and defining the parameter "a" as follow:

$$a = \frac{T_{Tp}}{T_{II}} = (M - 1) + (\frac{T_{Tc}}{T_{II}})(\frac{1}{1 - P_{SO}}) \qquad (3-27)$$

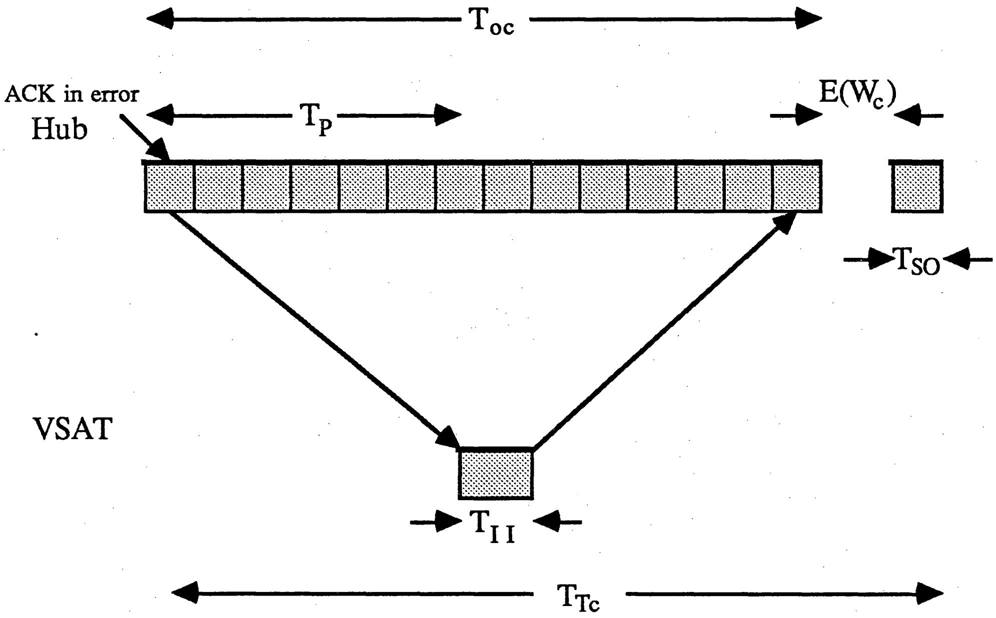


Figure 17. Transmission of Acknowledgement Frames.

where T_{Tc} is given by Eq. (3-26), T_{ll} is given by Eq. (3-4), and P_{so}, P_{ll} are given respectively, by Eq. (3-24), and Eq. (3-18), and M = 128. Then, the transmission time of a polled frame is

$$\overline{T}_p = T_{ll}\left[\frac{1+(a-1)P_{ll}}{(1-P_{ll})}\right] \tag{3-28}$$

We now define the traffic intensity of all VSATs

$$\rho_{ll} = N\lambda_{ll}T_{ll} \tag{3-29}$$

we also define the error factor to be

$$\varepsilon = \frac{1+(a-1)P_{ll}}{(1-P_{ll})} \tag{3-30}$$

With a given by Eq. (3-27), and P_{ll} given by Eq. (3-18), the total time to transmit all packets, waiting at the (KN) VSATs of the network, assuming that $\overline{T}_i$ is the same for all levels is then

$$\begin{aligned}\sum_{i=1}^{N}\overline{T}_i &= N\lambda_{ll}\overline{T}_c\overline{T}_p \\ &= N\lambda_{ll}\varepsilon T_{ll}\overline{T}_c \\ &= \rho_{ll}\varepsilon\overline{T}_c\end{aligned} \tag{3-31}$$

Inserting Eq. (3-1), and simplifying the resulting equation, one gets

$$\overline{T}_c = \frac{L}{(1-\rho_{ll}\varepsilon)} \tag{3-32}$$

Where L is given in Eq. (3-14), ρ_{ll}, and ε are given respectively by Eq. (3-29), and Eq. (3-30).

3-3 The M/D/1 Wait Time

The considered queue has a fixed service time equal to the transmission time of a polled information frame, $\overline{T}_p$. The total traffic of the polling system is given by $N\lambda_{\mathrm{II}}$. Referring to [7], the M/D/1 wait time of this system is given by

$$E(W_{\mathrm{II}}) = \frac{N\lambda_{\mathrm{II}}(\overline{T}_p)^2}{2 \times (1 - \rho_{\mathrm{II}}\varepsilon)}$$

$$= \frac{N\lambda_{\mathrm{II}}\varepsilon^2 T_{\mathrm{II}}^2}{2(1 - \rho_{\mathrm{II}}\varepsilon)}$$

$$= \frac{\rho_{\mathrm{II}}\varepsilon^2 T_{\mathrm{II}}}{2 \times (1 - \rho_{\mathrm{II}}\varepsilon)} \qquad (3-33)$$

3-4 VSAT-HUB Time Delay

The total access delay $E(D_{\mathrm{II}})$ for an arriving packet to a VSAT is the sum of a time delay due to polling, and an M/D/1 wait time due to packets already in the system prior to the packet arrival [2]

$$E(D_{\mathrm{II}}) = \frac{\overline{T}_c}{2}(1 - \frac{\rho_{\mathrm{II}}\varepsilon}{N}) + E(W_{\mathrm{II}}) \qquad (3-34)$$

Substituting $\overline{T}_c$, and $E(W_{\mathrm{II}})$ by Equations (3-32), and (3-34) one gets

$$E(D_{II}) = \frac{L(1 - \rho_{II}\frac{\varepsilon}{N})}{2(1 - \rho_{II}\varepsilon)} + \frac{\rho_{II}\varepsilon^2 T_{II}}{2(1 - \rho_{II}\varepsilon)} \qquad (3-35)$$

The total time delay, T_{VH}, of a packet being transmitted from a VSAT to the Hub station is the sum of the access delay and the transmission time of the packet sent.

$$T_{VH} = E(D_{II}) + \overline{T}_p \qquad (3-36)$$

$E(D_{II})$,and $\overline{T}_p$, are given by Equations (3-35), and (3-28), respectively.

IV. OUTBOUND LINE TIME DELAY

4-1 Introduction

The purpose of this chapter is to determine the performance of the outbound line for the network. This is expressed by the queueing time that a packet spends in the outbound line multiplexer (MUX) buffer before it can be purged from it, thus indicating that it has completed transmission. In the study of this multiplexed system for the outbound data traffic, packets going to different VSATs arrive from many unsynchronized peripherals and join a single packet stream to be transmitted on the common outbound line. Since the packets arrive in a bursty fashion, they have to wait in a queue before they can be transmitted. The total transmission time of a packet is determined by two major parameters. The first one is the queue length of the MUX buffer prior to the arrival of the packet. The second one is the number of retransmissions the packet experiences before it can actually complete its transmission.

In this chapter, we start the analysis of the outbound system by a study of the arrival process. The queue length distribution of the MUX is then determined using the known statistics of the arrival process. Later, the chapter continues on with a study of the retransmission distribution. Finally, the known statistics of both the queue length distribution and the retransmission process are combined to provide the characteristics of a MUX system using the go-back-N ARQ to cope with errors. The performance of such system is given by the time delay of the packets it is serving. This will then provide the outbound time delay of the network.

4-2 Model of the system

In this section, we list the characteristics of the mathematical model utilized throughout the analysis of the outbound channel time delay.

- The transmission rate of packets on the line out of the multiplexer buffer is $1/T_{IO}$ packet per second.
- The aggregate arrival process of packets to the multiplexer is a Poisson process.
- Time units of length T_{IO} on the outbound channel are referred to as slots.
- A packet requires exactly one slot for each transmission and the transmissions are synchronized to the occurrence of slots. Packets are transmitted on a First-Come-First-Serve (FCFS) basis. If the queue is not empty just prior to the beginning of a slot, the first packet in the queue is transmitted in that slot. If the queue is empty, no transmissions take place in that slot.
- An embedded Markov chain at the ends of slots is considered. We let L_i denote the number of packets buffered in the queue just prior to the start of the slot for $(i+1)^{th}$ slot.

- Individual packets are indexed by the subscript i. The random variable T_i refers to the time that packet i spends in the buffer. The waiting time is measured from the time of the arrival of the packet until the time it is removed from the MUX buffer.

- The queue has unlimited storage capacity. Thus, every packet can be stored prior to transmission.

- The delay due to control and ACK of the inbound traffic is expressed as an additional number of slots.

4-3 Preliminary Computations

4-3-1 Arrival Process:

The most frequently used process to model the behavior of queues is the Poisson process. Three basic statements are used to define the Poisson arrival process [3]. Consider a small time interval $\Delta t (\Delta t \rightarrow 0)$, separating times t and $t + \Delta t$. Then,

1. The probabilty of one arrival in the interval Δt is defined to be $\lambda \Delta t + 0(\Delta t)$, $\lambda \Delta t << 1$, where λ a specified proportionality constant.
2. The probability of zero arrivals in Δt is $1 - \lambda \Delta t + 0(\Delta t)$
3. Arrivals are memoryless; an arrival (event) in one time interval of length Δt is independent of events in previous or future intervals.

The mean time required to send a packet is T_{IO}. It corresponds to one time slot on the outbound line. Then, the probability generating function of the number of Poisson arrivals D_i during the packet transmission slot is given by [10]

$$G_D(Z) = E[Z^D] = \sum_{k=0}^{\infty} p_k Z^k$$

where

$$Pr\{D = k\} = p_k = \frac{\lambda_{IO}}{k} \exp(-\lambda_{IO}), \qquad k = 0,1.2... \qquad (4-1a)$$

Then,

$$G_D(Z) = \sum_{k=0}^{\infty} Z^k (\lambda_{IO})^k \frac{\exp(-\lambda_{IO})}{k}$$

$$G_D(Z) = \exp[-\lambda_{IO}(1-Z)] \qquad (4-1b)$$

The mean value, E[D] and the variance σ_D of the arrival process are given by [11]

$$E(D) = G'_D(Z)|_{Z=1} = \lambda_{IO} \qquad (4-1c)$$

and

$$\sigma_D^2 = G''_D(Z)|_{Z=1} + G'_D(Z)|_{Z=1} - (G'_D(Z)|_{Z=1})^2 = \lambda_{IO} \qquad (4-1d)$$

4-3-2 Queue Length Distribution

In this section we determine the statistics of the random variables L_i from the known statistics of the random variables D_i . A packet is removed in the i^{th} slot, provided that there is a packet present, when the queue is served. The remaining packets are joined by D_i new arrivals during the interval of time between slot (i-1) and slot i. The dynamic state of the MUX system is summurized as follows.

$$L_i = (L_{i-1} + D_i)$$

if there is a packet present at the i^{th} slot.

$$L_i = D_i$$

meaning that the queue was empty before the new arrival.

We define the unit step function as follow,

$$U(x) = 0 \quad x \leq \ 0 \qquad (4-2)$$

$$U(x) = 1 \quad x \geq \ 0$$

The fundamental equation for the MUX system queue length is then [12],

$$L_i = L_{i-1} - U(L_{i-1}) + D_i \qquad (4-3)$$

Assuming that the steady-state solution of the random variable L_i exists, then

$$\lim_{i \to \infty} E[L_{i+1}] = \lim_{i \to \infty} E[L_i] = E[L]$$

With $E[D_{i+1}]$ being the expected number of arrivals while a single packet is being transmitted, and $U(L_i)$ being the indicator function of the event that the number of packets in the system is greater than zero, we have

$$E[U(L_i)] = E[D_{i+1}] \tag{4-4}$$

Or, one also has

$$E[U(L_i)] = P[\text{ more than zero packets }] = 1 - P_0$$

where P_0 is the probabiliy that the system is empty. The probability generating function of the system state is then given by

$$P_i(Z) = E[Z^{L_i}] = \sum_{k=0}^{\infty} Z^k P[L_i = k]$$

Taking the expectations on both sides of (4-3) with i replaced by $i+1$ and evaluating $E[Z^{L_{i+1}}]$ we have

$$P_{i+1}(Z) = E[Z^{L_i - U(L_i) + D_{i+1}}] \tag{4-5}$$

The number of arrivals during a packet transmission time is independent of the number of packets in the queue, and one can write

$$\begin{aligned} P_{i+1}(Z) &= E[Z^{L_i - U(L_i)}]E[Z^{D_{i+1}}] \\ &= [Z^{L_i - U(L_i)}]G_D(Z) \end{aligned} \tag{4-6}$$

Since we have a stationary process we can suppress the dependence on i for D_i and, therefore, $E[Z^{D_{i+1}}] = E[Z^{D_i}] = G_D(Z)$. Now consider the term

$$
\begin{aligned}
E[Z^{L_i - U(L_i)}] &= [\sum_{k=0}^{\infty} Z^{k-U(k)}]P[L_i = k] \\
&= P_0 + \sum_{k=1}^{\infty} Z^{k-1} P[L_i = k] \\
&= P_0 + Z^{-1}[\sum_{k=0}^{\infty} P[L_i = k] - P_0] \\
&= P_0 + Z^{-1}[P_i(Z) - P_0]
\end{aligned}
\qquad (4-7)
$$

From equations (4-5), (4-6) and (4-7) we have

$$P_{i+1}(Z) = [P_0 + Z^{-1}(P_i(Z) - P_0)]G_D(Z) \qquad (4-8)$$

Assuming that a steady-state solution exists,

$$\lim_{i \to \infty} P_{i+1}(Z) = \lim_{i \to \infty} P_i(Z) = P(Z)$$

The traffic intensity for the Poisson arrival during one slot of time is

$$\rho_{IO} = \lambda_{IO}$$

The probability that there are no packets in the MUX buffer is

$$P_0 = 1 - \rho_{IO} \qquad (4-9)$$

Utilizing Equation (4-9) we get the probability generating function of the queue length

$$G_L(Z) = P(Z) = \frac{(1-\rho_{IO})(Z-1)G_D(Z)}{Z - G_D(Z)} \qquad (4-10)$$

The mean value of the queue length is obtained by differentiating $G_L(Z)$ at $Z = 1$.

The first derivative of Eq. (4-10), yields

$$G'_L(Z)[G_D(Z) - Z] + G_L(Z)[G'D(Z) - 1] = (1 - \rho_{IO})(-1)G_D(Z) + (1 - \rho_{IO})(1 - Z)G'_D(Z)$$

And the second derivative gives

$$G''_D(Z)[G_L(Z) - Z] + 2[G'_L(Z) - 1] + G'_L(Z)G''_D(Z) = 2(1 - \rho_{IO})(-1)G'_D(Z) + (1 - \rho_{IO})(1 - Z)G''_D(Z)$$

With $Z = 1$, $G_D(1) = G_L(1) = 1$, one finds

$$E[L] = \frac{(1 - \rho_{IO})G'_D(1)}{[1 - G'_D(1)]} + \frac{G''_D(1)}{[2(1 - G'_D(1))]}$$

Substituting ρ_{IO} by $G'_D(1)$ one gets

$$E[L] = G'_D(1) + \frac{G''_D(1)}{2[1 - G'_D(1)]} \qquad (4-11)$$

Therefore,

$$\begin{aligned} E[L] &= \frac{2G'_D(1) - 2(G'_D(1)^2) + G''_D(1)}{2[1 - G'_D(1)]} \\ &= \frac{G'_D(1) - (G'_D(1))^2 + (G''_D(1) + G'_D(1) - (G'_D(1))^2)}{2[1 - G'_D(1)]} \qquad (4-12) \\ &= \frac{E[D]}{2} + \frac{\sigma_D^2}{2[1 - E[D]]} \end{aligned}$$

and

$$G_L(Z) = \frac{(1 - E[D])(Z - 1)G_D(Z)}{Z - G_D(Z)} \qquad (4-13)$$

4-3-3 Retransmission of Packets

A packet may be retransmitted several times due to an error. The error can take place on the outbound line for a transmitted packet or on the inbound line for an ACK frame of that packet. This section determines the expected number of retransmissions of an outbound packet. A packet may be retransmitted in any of two cases: (1) the information frame carrying the packet is in error or (2) the information frame is correct however, its ACK frame is in error. Treating an ACK frame in error as a NAK, the event of retransmitting a packet is given by the the probability P,

$$\begin{aligned} P &= P_{IO} + (1 - P_{IO})P_{II} \\ &= P_{IO} + P_{II} - P_{IO}P_{II} \end{aligned} \qquad (4-14)$$

where P_{IO} = The probability that the information frame is in error,

$$P_{IO} = 1 - (1 - P'_b)^{l+l'} \qquad (4-15)$$

and P_{II} = The probability that the ACK frame is in error,

$$P_{II} = 1 - (1 - P_b)^{l+l'} \qquad (4-16)$$

A packet completes transmission if and only if it is received correctly and its ACK is also received correctly. The probability of this joint event equals the probability that the packet is not being retransmitted, i.e. (1 - P).

Let N denote the number of retransmissions of an outbound packet. The probability generating function of N is

$$G_N(Z) = E[Z^N] = \sum_{i=1}^{\infty} Z^i \Pr\{N = i\}$$

$$G_N(Z) = (1 - P)\sum_{i=1}^{\infty} Z^i P^{i-1} = \frac{(1 - P)Z}{(1 - PZ)} \tag{4-17a}$$

Thus

$$E[N] = G'_N(Z)|_{Z=1} = \frac{1}{(1 - P)} \tag{4-17b}$$

and

$$\sigma_N^2 = G''N(Z)|_{Z=1} + G'_N(Z)|_{Z=1} + (G'_N(Z)|_{Z=1})^2 = \frac{P}{(1 - P)^2} \tag{4-17c}$$

4-4 Analysis approach

In order to begin the analysis of the packet time distribution, we need to relate the computed parameters to this specific application. The system considered is one in which a transmitted packet cannot be purged from the outbound line buffer until it has been successfully received by its destination VSAT. The total time a packet spends in the queue depends on the number of retransmissions that took place due to an error. The strategy employed for coping with errors during a transmission is the go-back-N protocol. The application of this scheme is as follow. Consider an arbitrary packet i. ACK/NAK frames for packets (i+1), (i+2), ... are ignored until an ACK frame for packet i is received. If an ACK is received in error or a NAK is received, packet i is retransmitted in the next slot. All packets not yet purged from the buffer are then transmitted/retransmitted in sequence in the next available slots.

Assume that packet i has been transmitted in the time slot corresponding to the time interval $(i-1)T_{IO} \leq t \leq iT_{IO}$, the outbound line then serves all succeeding packets in the queue up to the maximum number of transmissions without receiving acknowledgement. Having a satellite link, with a sequence number modulus of 128, the maximum number of packets that can be received by a VSAT without exhausting its window, or buffer is (128 - 1). In the following study we will assume that the error recovery takes place before any of the destination VSATs receives a total of 127 packets. Now assume that AK is the total number of slots between the transmission of a packet and the receipt of its acknowledgement. The acknowledgement frame for the transmitted packet then occurs during the time interval

$$(i + AK - 1)T_{IO} \leq t \leq (i + AK)T_{IO}$$

.

The first transmission of a packet i takes only one slot, however, all subsequent retransmissions take (1 + AK) slots. We denote it by M the total number of slots required for a successful transmission of a packet. M forms a sequence of random variables that is related to the random variable N by the following equation

$$M = 1 + (N - 1)(1 + AK) = (N - 1)AK + N$$

The mean, variance, and generating function of M are

$$E[M] = (1 + AK)E[N] - AK, \qquad (4-18a)$$

$$\sigma_M^2 = (1 + AK)^2 \sigma_N^2, \qquad (4-18b)$$

$$G_M(Z) = \frac{G_N(Z^{1+AK})}{Z^{AK}}. \qquad (4-18c)$$

Before we determine the queue waiting time of the system, we need to determine the acknowledgement component AK.

4-4-1 Acknowledgement Delay

The acknowledgement delay is the interval of time between a transmission of a packet and the receipt of its acknowledgement in units of slots. Operating in normal response mode a receiving VSAT cannot acknowledge its packets unless it is in a transmission state. Since packets are transmitted in order of arrival, independent of the state of their destination VSATs, a receiving VSAT has to wait until it is polled to be able to send acknowledgement frames back to the Hub station. The minimum time for a packet i to be acknowledged consists of its one way propagation delay (Hub-to-VSAT), the transmission time of the inbound packet carrying the acknowledgement, and its propagation delay (VSAT-to-Hub). This event happens when packet i is received by a transmitting VSAT. On the other hand, the maximum time for an acknowledgement frame to arrive at the Hub station is equal to the time it takes to cycle among all VSATs, and give them permission to transmit. This takes place when packet i is sent to a VSAT that has just completed transmission.

In order to determine the total acknowledgement delay, one needs to find the delay introduced due to control and acknowledgement of the inbound traffic. The delay due to control and ACK traffic appears in two forms. First, it takes place when an outbound packet is being transmitted on the ATDM channel for the first time. This delay is introduced as an extra delay in the MUX queue waiting time. It is experienced by all packets waiting to be transmitted to their destination VSATs. Secondly, the delay takes place when the packet is being retransmitted due to an error. This appears as an extra delay in the packet waiting time to be merged from the MUX buffer. We make the following assumptions.

- The delay introduced in the first form is very small compared to the total transmission time of a packet, including its error recovery. Thus, one can assume that it is equal to zero.

- The delay due to control and ACK of inbound traffic introduced in the second form is the total delay considered during the analysis. It appears in the system as an additional number of slots in the acknowledgement delay. It is equal to the time spent by the outbound channel serving control and ACK frames between the transmission time of a packet and the arrival of its acknowledgement frame.

For the minimum acknowledgement time AK_{min}, the delay due to control and ACK of the inbound traffic delay is given by

$$\frac{(2T_P + T_{II})(\lambda_c T_{SO})}{T_{IO}}$$

then

$$AK_{min} = \frac{(2T_P + T_{II})(1 + \lambda_c T_{SO})}{T_{IO}} \qquad (4-19)$$

For the maximum acknowledgement time AK_{max}, the delay due to control and ACK of the inbound traffic is included in the polling access delay, thus

$$AK_{max} = \frac{(2T_P + E(D_{II}))}{T_{IO}} \qquad (4-20)$$

The average acknowledgement time AK, in slots of length T_{IO} , is limited by

$$AK_{min} \quad \leq \quad AK \quad \leq \quad AK_{max}$$

4-4-2 Queue Waiting Time

In order to find the total time delay that a packet has to spend in the system to complete transmission, we need to find its wait time in the queue. This delay is in turn equal to the total transmission time of the packets already in the queue prior to the packet arrival.

We start this analysis by considering a modified system where a packet i is thought to be removed from the queue AK slots prior to when the ACK frame is successfully received for that packet. That is, if an acknowledgement frame is received for packet i during slot i, we will assume that the packet was removed from the queue at the end of slot (i-AK). This describes a system where the total number of slots required for a successful transmission is $(1+AK)N$. The total delay time of a packet in our actual system is then the sum of the wait time in the modified system plus AK additional slots. The arrival process of packets is a standard Poisson process and every packet requires a random number of slots to complete transmission. If we suppose that every packet stored in the queue is assigned a fixed number of modified packets, this number is equal to the number of slots required to successfully transmit that packet. And, if we further assume that packets are converted to modified packets prior to their entrance to the queue, we will have a compound Poisson arrival of modified packets. Every packet has a random number of modified packets. If we suppose that the modified packets are stored in the queue, then the transmission delay of these modified packets is equal to the access delay of an arbitrary arriving packet.

Let α_i denote the number of modified packets stored in the queue just prior to the start of the slot for $(j+1)^{th}$ slot and let β_i denote the number of modified packets which arrived at the system between time interval of $(i-1)^{th}$ slot and i^{th} slot, The equation which relates these two quantities is

$$\alpha_i = \alpha_{i-1} - U(\alpha_{i-1}) + \beta_i \qquad (4-21)$$

This is the same fundamental equation as the queue length for the ATDM, but concerns modified packets. Thus, the mean and generating function for this random variable, as computed earlier, is given by

$$E[\alpha] = \frac{E[\beta]}{2} + \frac{\sigma_\beta^2}{2(1 - E[\beta])} \tag{4-22}$$

$$G_\alpha(Z) = \frac{(1 - E[\beta])(Z - 1)G_\beta(Z)}{Z - G_\beta(Z)} \tag{4-23}$$

We need to relate the parameters in these equations to the known parameters E[D], σ_D^2, and $G_D(Z)$ We refer to the following

Lemma [13]: Let $A = B_1 + B_2 + \ldots + B_K$ be a random sum of random variables, where B_1, B_2,.... are identically distributed, statistically independent random variables with mean μ_D,variance σ_D^2, and generating function $G_B(Z)$. Let K be the number of these random variables, K is itself a random variable, statistically independent of the B′s with mean μ_K, variance σ_K^2 , and generating function $G_K(Z)$. Then, the mean, variance and generating function of the random variable A are

$$\mu_A = \mu_K \mu_B \tag{4-24a}$$

$$\sigma_A^2 = \mu_K \sigma_B^2 + \mu_B^2 \sigma_K^2 \tag{4-24b}$$

and

$$G_A(Z) = G_K[G_B(Z)] \tag{4-24c}$$

Returning to our case, we see that β is the sum of D statistically independent, identically distributed random variables M, the mean, variance and generating function of D are then given by

$$E[\beta] = E[D]E[M] \tag{4-25a}$$

$$\sigma_\beta^2 = E[D]\sigma_M^2 + E[M]^2\sigma_D^2 \qquad (4-25b)$$

and

$$G_\beta(Z) = G_D[G_M(Z)] \qquad (4-25c)$$

Substituting these expressions into Equations (4-22) and (4-23), one obtains the mean and generating functions for α,

$$E(\alpha) = \frac{E[D]E[M]}{2} + \frac{E[D]\sigma_M^2 + E[M]^2\sigma_D^2}{2(1 - E[D]E[M])} \qquad (4-26)$$

and

$$G_\alpha(Z) = \frac{(1 - E[D]E[M])(Z-1)(G_D(G_M(Z)))}{[Z - G_D(G_M(Z))]} \qquad (4-27)$$

4-5 Time Delay Distributions:

Consider an arbitrary arriving packet i, the total time T_i it spends in the system before it completes its transmission consists of the following time delays:

1. T′, the time that the arriving packet must wait due to packets already in the queue before its arrival,
2. T″, the time that the packet must wait due to packets within its own arrival group which are placed ahead of it in the queue and
3. M the packet service time.

4-5-1 Computation of T′

T′ is the time that packet i has to wait due to packets waiting in the queue. The unfinished work in the queue at the beginning of the slot just prior to the arrival of an arbitrary packet has the generating function given by $G_\alpha(Z)$ [10]. If the unfinished work is positive, there will be one modified packet less at the time of arrival. In this case the generating function for T′ is

$$G_{T'}(Z) = \frac{G_\alpha(Z) + (Z-1)\{1 - E[D]E[M]\}}{Z}$$

Substituting $G_\alpha(Z)$ by Eq. (4-27), one gets

$$\begin{aligned} G_{T'}(Z) &= \frac{\{1 - E[D]E[M]\}(Z-1)[G_M(Z)]}{Z\{Z - G_D[G_M(Z)]\}} + \frac{\{1 - E[D]E[M]\}(1 - Z)}{Z} \\ &= \frac{\{1 - E[D]E[M]\}(Z-1)\{G_D[G_M(Z)]\}}{Z\{Z - G_D[G_M(Z)]\}} + \frac{\{1 - E[D]E[M]\}(Z-1)\{Z - G_D[G_M(Z)]\}}{Z\{Z - G_D[G_M(Z)]\}} \\ &= \frac{\{1 - E[D]E[M]\}(Z-1)}{Z\{Z - G_D[G_M(Z)]\}}\{G_D[G_M(Z)] + Z - G_D[G_M(Z)]\} \end{aligned}$$

Simplifying the last equation yields

$$G_{T'}(Z) = \frac{(1 - E[D]E[M])(Z-1)}{\{Z - G_D[G_M(Z)]\}} \qquad (4-28)$$

4-5-2 Computation of T″

T″ is the time that packet i has to wait due to packets within its own arrival group, which are placed ahead of it in the queue. Let Y be a random variable which denotes the size of the group in which the particular packet of interest arrives. The probability that this group is of size k is given by

$$P[Y = k] = \frac{kP[D = k]}{E[D]}, \; k = 1,2,... \qquad (4-29)$$

The distribution for the number of packets X which are served ahead of our arbitrary packet is given by [10]

$$P[X = l] = \sum_{k=l+1}^{\infty} \frac{P[D = k]}{E[D]}, \quad l = 0,1,2,.... \qquad (4-30)$$

The generating function of X is [11]

$$G_X(Z) = \frac{[1 - G_D(Z)]}{E[D](1 - Z)} \qquad (4-31)$$

The delay T″, an arriving packet observes is just the sum of a random number of independent instances of M, namely

$$T'' = M_1 + M_2 + ... + M_X$$

Hence, by the lemma, the generating function of T″ is

$$G_{T''}(Z) = G_X[G_M(Z)]$$

$$G_{T''}(Z) = \frac{1 - G_D(G_M(Z))}{E[D][1 - G_M(Z)]} \qquad (4-32)$$

4-5-3 Hub-VSAT Time Delay

The total delay time for the modified system, denoted by TM, is

$$TM = T' + T'' + M$$

where T', T'' and M are statistically independent random variables. The generating function of the total time delay of the modified system is the product of the generating functions of the three random variables [14]

$$\begin{aligned} G_{TM}(Z) &= G_{T'}(Z)G_{T''}(Z)G_M(Z) \\ &= \frac{\{1-E[D]E[M]\}(Z-1)}{\{Z-G_D[G_M(Z)]\}} \frac{\{1-G_D[G_M(Z)]\}}{\{E[D][1-G_M(Z)]\}} G_M(Z) \\ &= \{1-E[D]E[M]\}(Z-1) \frac{G_M(Z)\{1-G_D[G_M(Z)]\}}{E[D]\{1-G_M(Z)\}\{Z-G_D[G_M(Z)]\}} \end{aligned} \qquad (4-33)$$

From the moment generating properties of the generating function [14], the average delay E[TM] is found by differentiating $G_{TM}(Z)$ and evaluating at $Z = 1$

$$E[TM] = \frac{E[M]}{2} + \frac{E[D]^2\sigma_M^2 + E[M]\sigma_D^2}{2E[D]\{1-E[D]E[M]\}} \qquad (4-34)$$

The Hub-to-VSAT average time delay T_{HV} for a random packet i in slots of duration T_{IO} is

$$\begin{aligned} T_{HV} &= E[T] = E[TM] + AK \\ &= \frac{E[M]+2AK}{2} + \frac{E[D]^2\sigma_M^2 + E[M]\sigma_D^2}{2E[D]\{1-E[D]E[M]\}} \end{aligned} \qquad (4-35)$$

Substituting E[D], σ_D^2 and $G_D(Z)$, respectively by Equations (4-1c), (4-1d), and (4-1b). Also substituting E[M], σ_M^2 and $G_M(Z)$ by Equations (4-18a), (4-18b), and (4-18c) respectively one gets

$$T_{HV} = \frac{(1+AK \times P)+2AK(1-P)}{2(1-P)} + \lambda_{IO}\frac{P(1+AK)^2}{(1-P)^2} + \frac{[\frac{1+AK \times P}{1-P}]}{2(1-\lambda_{IO}\frac{(1+AK \times P)}{(1-P)})} \qquad (4-36)$$

And the probability generating function for the time delay is

$$\begin{aligned} G_T(Z) &= Z^{AK} G_{TM}(Z) \\ &= Z^{AK}\{1 - E[D]E[M]\}(Z-1) \times \frac{G_M(Z)\{1 - G_D[G_M(Z)]\}}{E[D]\{1 - G_M(Z)\}\{Z - G_D[G_M(Z)]\}} \end{aligned} \qquad (4-37)$$

4-5-3 VSAT-VSAT Time Delay

Operating in double-hop mode with routing being done through the Hub station, a packet sent from a VSAT to another VSAT must arrive at the Hub station first, then sent to the destination VSAT. The VSAT-to-VSAT packet delay is the sum of the inbound delay and the outbound delay.

$$T_{VV} = T_{VH} + T_{HV} \qquad (4-38)$$

V. NUMERICAL ANALYSIS

In this chapter we include some numerical results obtained using the formulas derived for evaluating the performance of the network. In these numerical results, the following values are used

$K =$ Number of groups in the network $= 5$

$M =$ Sequence number modulus $= 128$

$R_b =$ Channel bit rate of an inbound line $= 56\ 10^3$bits/sec

$R'_b =$ Channel bit rate of the outbound line $= 256\ 10^3$bits/sec

$I =$ Information field length $= 2048$ bits

$I' =$ Overhead $= 48$ bits

$T_p =$ One way propagation delay $= 250\ 10^{-3}$secs

$T_{II} =$ Inbound information frame length $= 37.428\ 10^{-3}$secs

$T_{IO} =$ Outbound information frame length $= 8.1875\ 10^{-3}$secs

$T_{SI} =$ Inbound supervisory frame length $= .8571\ 10^{-3}$secs

$T_{SO} =$ Outbound supervisory frame length $= .1875\ 10^{-3}$secs

$\lambda_C =$ Arrival rate of ACK and control traffic $= .3$ frames/sec

Tables 1 through 6 include the different time delays for the roll-call polling system developed for serving the inbound traffic. In these tables, assuming that the outbound arrival rate λ_{IO} is equal to .7 packets/sec, we observe the different effects of varying the inbound arrival rate, the number of VSATs in the network, and the outbound error bit rate on the load of the system as well as on the walk time, cycle time and access time. For very little traffic, the total walk time is effectively the cycle time of the polling system, and the minimum access delay is then one half the walk time. If the traffic increases to a load of $\rho_{II} = .5$, the average cycle time doubles. The access delay increases accordingly with both the one-half component of the cycle time, and the M/D/1 wait time. An increase in the number of VSATs results in an increase of all time delays of the system. The walk time and the transmission time, being dependent on this parameter, both go up, resulting in an additional delay in the cycle time. This delay is passed on to the access time delay. In Tables 4, 5 and 6 we observe the effect of the outbound bit error rate on the system delays. We see a significant increase in these delays. From Eq. (3-14), one can see that the walk time is a function of this component. A bit error rate P'_b, results in an increase in the number of retransmissions of the polling command frame. This leads to a greater transmission time of this frame, thus a longer system walk time. Also, an outbound bit error rate results in an increase in the transmission time of ACK frames of the polled packets resulting in an increase in the polling transmission time. The changes in each of these components is reflected in the system cycle time, thus in the system access delay.

Figure 18, shows T_{VH}, the time delay of a packet transmitted from a VSAT to the Hub station as a function of λ_{II} , the inbound arrival rate. As can be observed, the time delay VSAT-to-Hub goes up as the the arrival rate gets larger. The absolute effect on T_{VH} is expressed in the access delay. This depends on the one-half component of the cycle time and the M/D/1 wait time of the system. Both components are absolute positive functions of the inbound arrival rate λ_{II}. Figure 19 is a plot of T_{VH} as a function of the inbound bit error rate P_b , with the outbound bit error rate P'_b , being fixed to 10^{-5}. In this Figure, we observe the VSAT-to-Hub time delay increase with increasing P_b . An increase in the bit error rate results in an increase in the walk time (in the ACK frame of the polling command), it also results in a longer polled packet transmission time due to an increase in the number of transmissions of such a packet. Finally, because of P'_b, an additional delay in the M/D/1 wait time takes place. In Figures 18 and 19, we notice that the changes in T_{VH} are not so significant where the number of VSATs in the network is only 25. In this case, even at a high arrival rate, the load on the polling system is still small. When the number of VSATs is large, the load, and thus the time delay T_{VH}, experience a big delay increase as the arrival rate gets higher.

In Figures 21 and 22 we plot T_{HV}, the time delay of a packet transmitted from the Hub station to a VSAT as a function of P'_b, the outbound bit error rate. The acknowledgement time AK is taken to be equal to $\frac{(AK_{max} + AK_{min})}{2} T_{IO}$ secs . Both the inbound arrival rate and the outbound arrival rate are set to .7 packets/sec. In Figure 21 where the inbound bit error rate, P_b, is equal to 0, we see that the increase caused by changing P'_b is not as significant as in Figure 22 where P_b is equal to 10^{-6}. The ACK time AK, being longer in the second case because of the longer access delay, thus a larger AK_{max} that introduces an additional delay in T_{HV} . As expected, the Hub-to-VSAT time delay increases with increasing outbound bit error rate due to an increase in the number of retransmissions of a packet. In Figures 23, 24 and 25, we show T_{HV} as a function of the ACK time AK. This emphasises the strong correlation between serving the inbound traffic by the VSATs in a polling system and acknowledging the outbound traffic.

Table 1. Roll-call Polling Time Delays, N = 5

Load $[\rho_{II}]$	Walk Time $[L]$	Cycle Time $[T_c]$	Access Delay $[E(D_{II})]$
0.00	1.280	1.280	0.640
0.05	1.280	1.345	0.667
0.10	1.280	1.417	0.697
0.15	1.280	1.500	0.731
0.19	1.280	1.588	0.769

K*N = 25 VSATs $P_b = 10^{-6}$ $P'_b = 0$ All times in Secs.

Table 2. Roll-call Polling Time Delays, N = 10

Load $[\rho_{II}]$	Walk Time [L]	Cycle Time $[T_c]$	Access Delay $[E(D_{II})]$
0.00	2.560	2.560	1.280
0.10	2.560	2.834	1.406
0.19	2.560	3.176	1.562
0.29	2.560	3.610	1.763
0.39	2.560	4.183	2.026

K*N = 50 VSATs $P_b = 10^{-6}$ $P'_b = 0$ All times in Secs.

Table 3. Roll-call Polling Time Delays, N = 20.

Load $[\rho_{II}]$	Walk Time [L]	Cycle Time $[T_c]$	Access Delay $[E(D_{II})]$
0.00	5.119	5.119	2.559
0.19	5.119	6.351	3.151
0.39	5.119	8.366	4.117
0.58	5.119	12.251	5.981
0.77	5.119	22.875	11.077
K*N = 100 VSATs	$P_b = 10^{-6}$	$P_b' = 0$	All times in Secs.

Table 4. Effect of Outbound Bit Error Rate on the Polling Delays, $N = 5$.

Load [ρ_{II}]	Walk Time [L]	Cycle Time [T_c]	Access Delay [$E(D_{II})$]
0.00	1.286	1.286	0.643
0.05	1.286	1.352	0.671
0.10	1.286	1.424	0.701
0.15	1.286	1.505	0.735
0.19	1.286	1.596	0.773
$K*N = 25$ VSATs	$P_b = 10^{-6}$	$P'_b = 10^{-5}$	All times in Secs.

Table 5. Effect of Outbound Bit Error Rate on the Polling Delays, N = 10.

Load [ρ_{II}]	Walk Time [L]	Cycle Time [T_c]	Access Delay [$E(D_{II})$]
0.00	2.572	2.572	1.286
0.10	2.572	2.849	1.413
0.19	2.572	3.192	1.571
0.29	2.572	3.629	1.771
0.39	2.572	4.204	2.036
K*N = 50 VSATs	$P_b = 10^{-6}$	$P'_b = 10^{-5}$	All times in Secs.

Table 6. Effect of Outbound Bit Error Rate on the Polling Delays, N = 20.

Load [$\rho_{I\,I}$]	Walk Time [L]	Cycle Time [T_c]	Access Delay [$E(D_{I\,I})$]
0.00	5.020	5.020	2.510
0.15	5.020	5.935	2.948
0.30	5.020	7.257	3.581
0.46	5.020	9.338	4.578
0.62	5.020	13.092	6.370

K*N = 100 VSATs $P_b = 10^{-6}$ $P'_b = 10^{-5}$ All times in Secs.

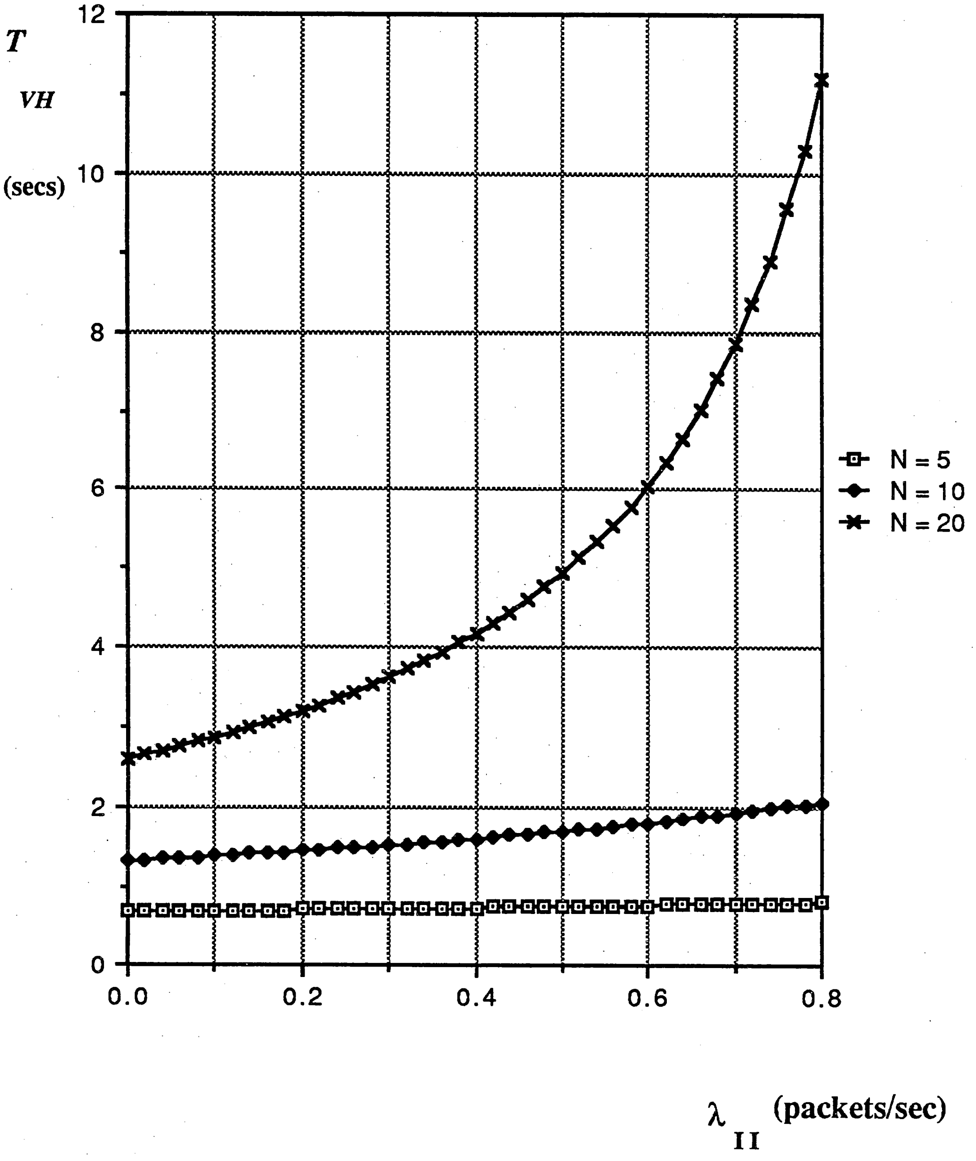


Figure 18. Inbound Time Delay vs. Inbound Arrival Rate.

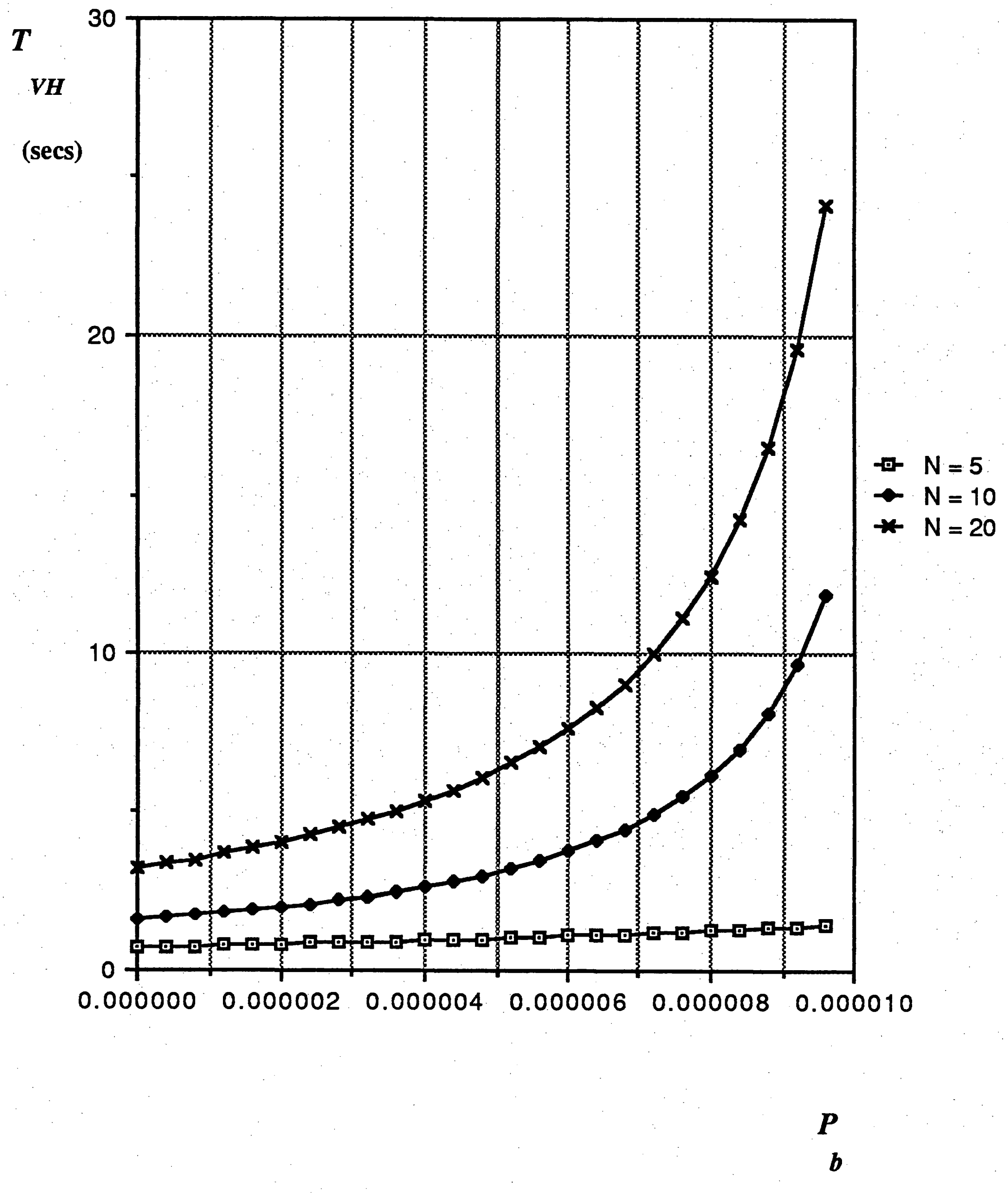


Figure 19. Inbound Time Delay vs. Bit Error Rate.

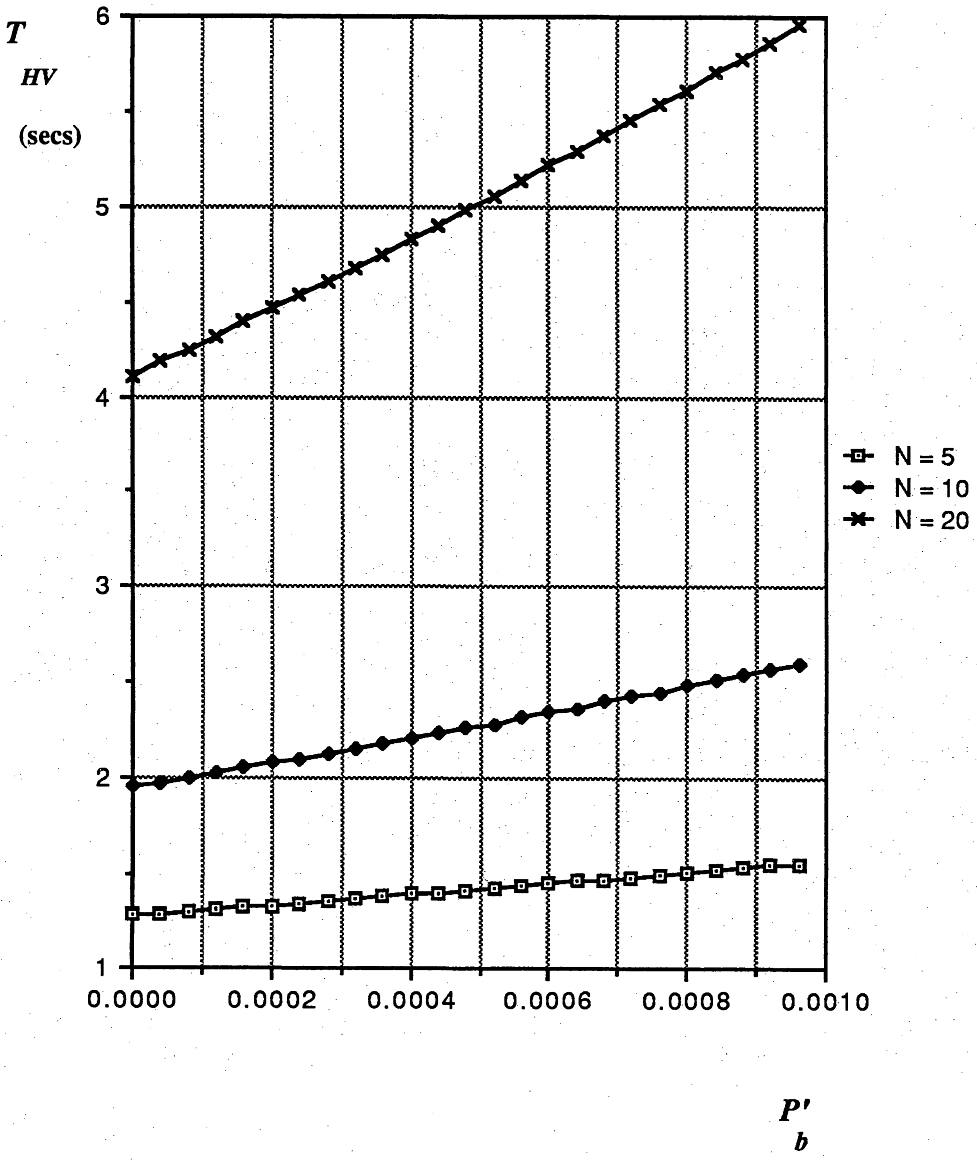


Figure 20. Outbound Time Delay vs. Bit Error Rate.

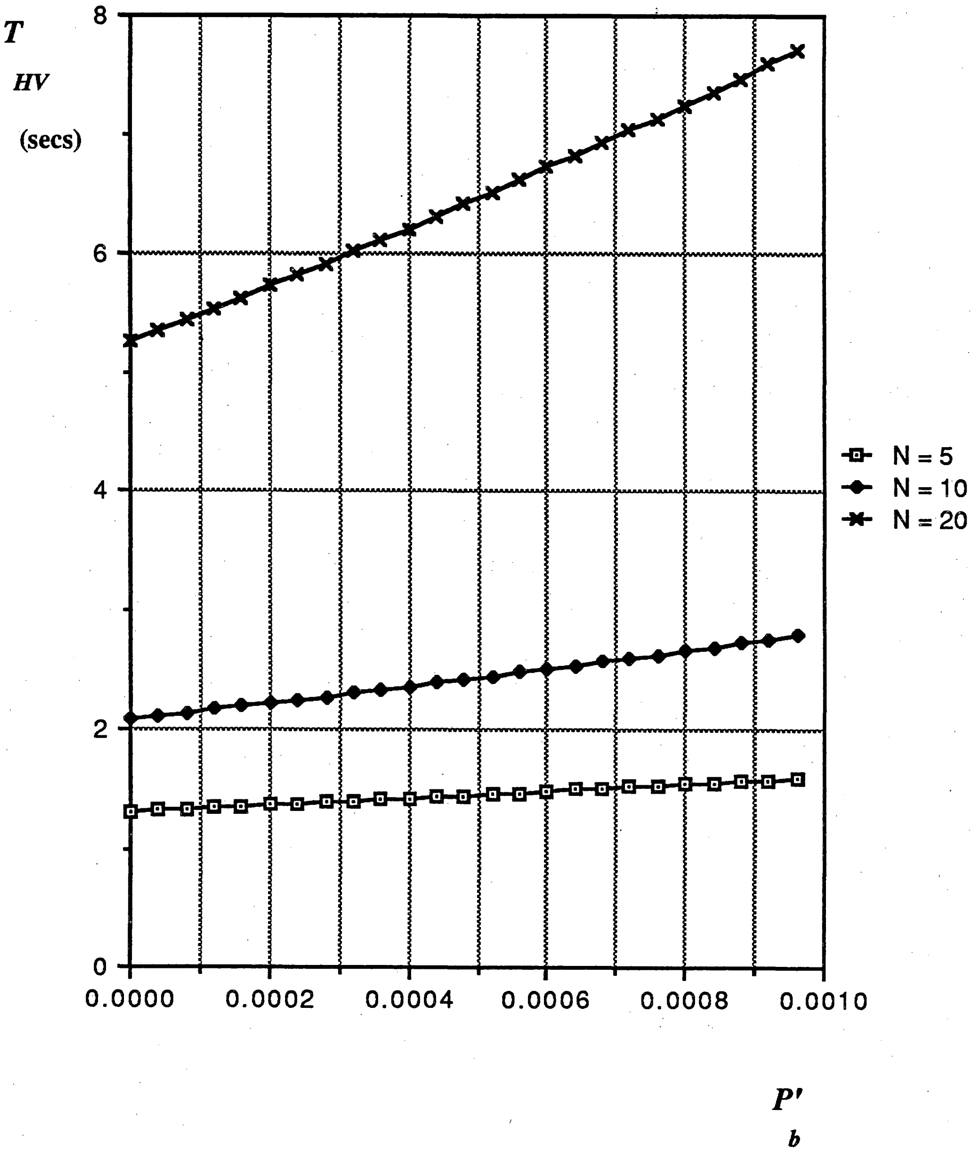


Figure 21. Outbound Time Delay vs. Bit Error Rate.

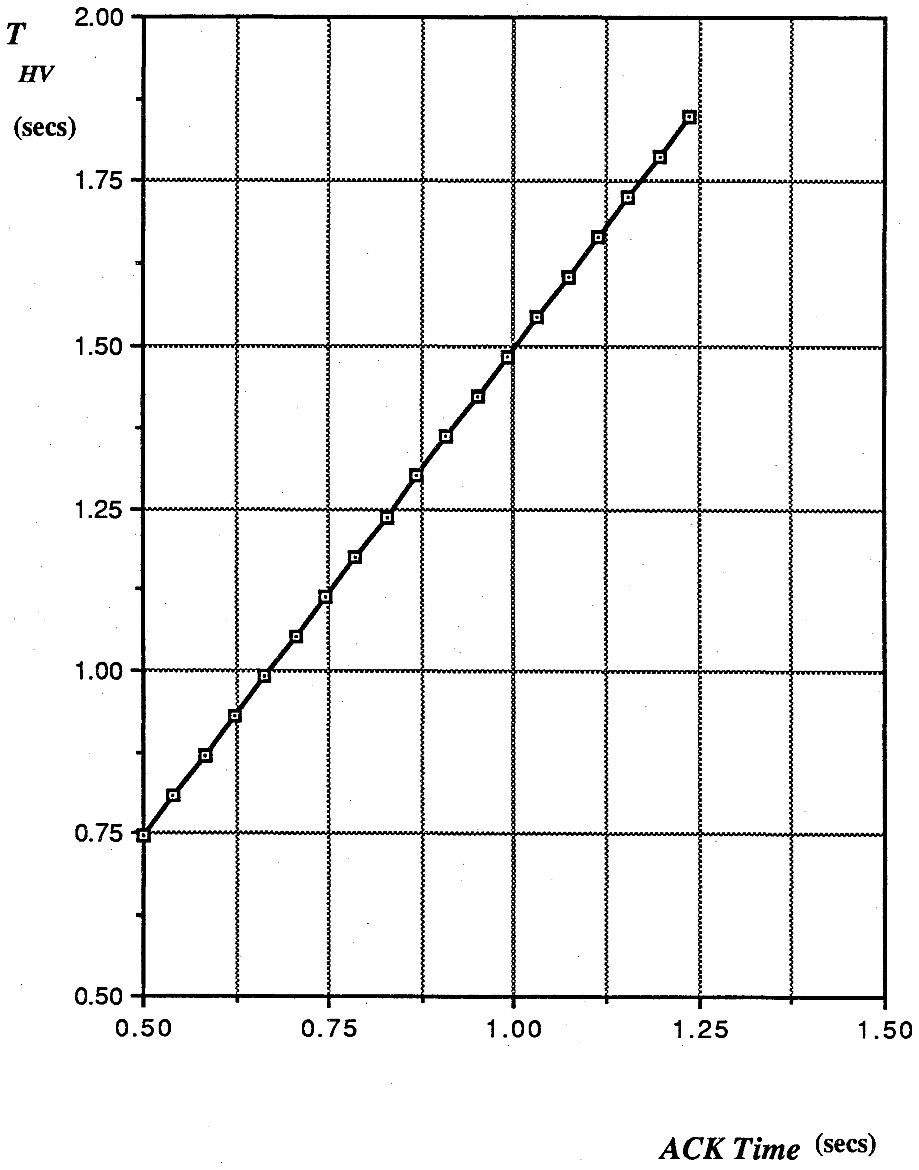


Figure 22. Outbound Time Delay vs. ACK Time, N = 5.

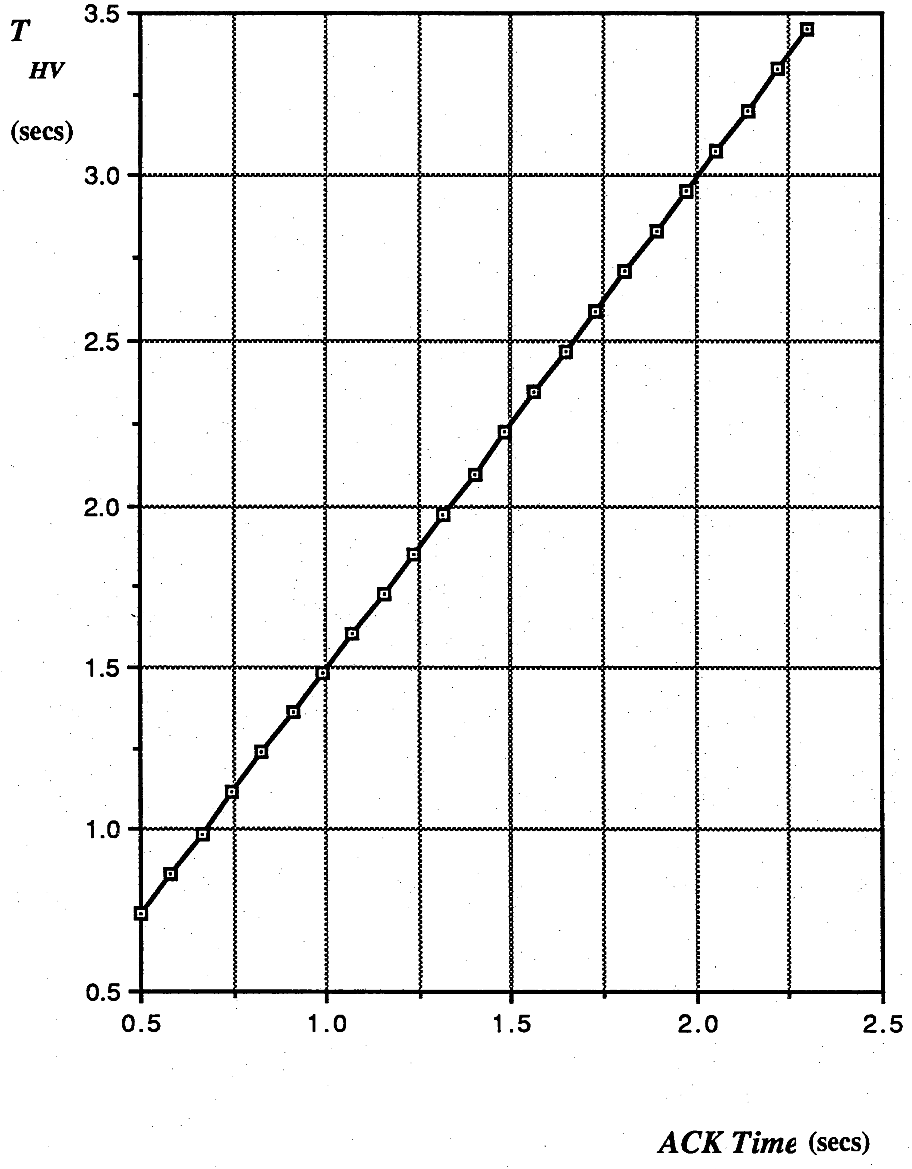


Figure 23. Outbound Time Delay vs. ACK Time, N = 10.

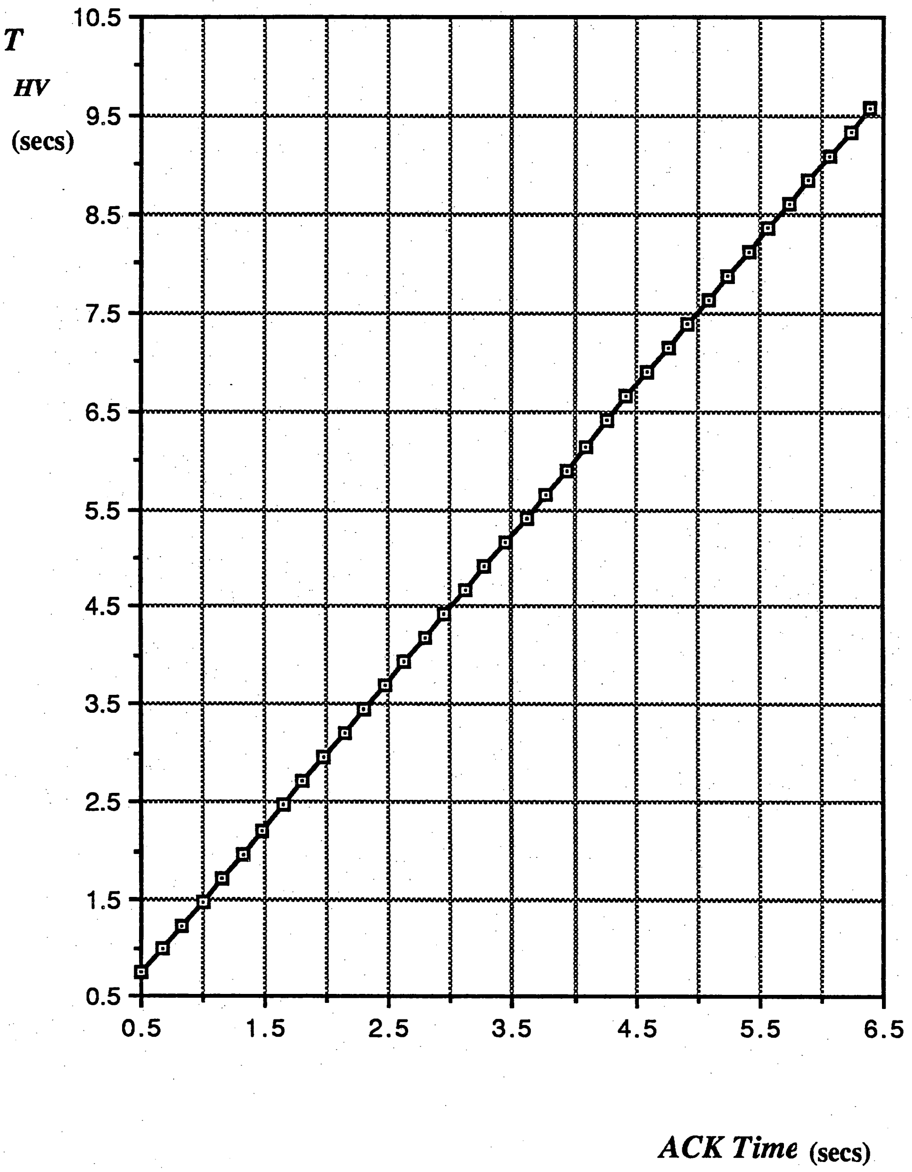


Figure 24. Outbound Time Delay vs. ACK Time, N = 20.

CONCLUSION

The performance of roll-call polling multiple access scheme in HDLC has been carried out in earlier studies. These studies however, restrict their models to a transmission media that is free of error. The primary contribution of this research is that it provides a mathematical formulation for directly computing the time delays of a roll-call polling system where we have specified the channel error bit rate.

Two tasks are carried out to provide the performance analysis of a VSAT star network. First, a polling model is developed for the VSAT-to-Hub communication. In this polling satellite communication system, VSATs from different groups take turns in transmitting data to the Hub station using their preassigned inbound multiaccess channels. The model is developed using a go-back-N ARQ error recovery scheme applied to an HDLC point-to-point link. Second, an ATDM model in which the Hub-to-VSAT communication is performed is provided. In this latter system, the Hub uses its satellite TDM broadcast outbound channel to send different signals to the VSATs of the network. These signals are of the control type, beamed back by the satellite to different VSATs of the network, or, they are of the information type, transmitted to a different VSAT each time. The model developed for serving the outbound traffic also uses go-back-N ARQ error recovery scheme. The error recovery technique developed for both models is applied to control possible errors experienced on the transmission channels. The case studied assumes independent error events which can take place during data and/or acknowledgement transmission.

Due to the master-slave relationship that exists between the two communicating sides, the performance of one model depends a great deal on the other. Using a multiple access scheme that is controlled by the Hub station, the inbound line performance is determined by the operation of the outbound line. And, performing both tasks, the control of the inbound traffic as well as serving the outbound traffic, the outbound line performance is limited by the VSAT operations.

In looking at the interaction of roll-call polling and ATDM as one multiple access procedure for double-hop star network as a whole, we proposed an explicit formula to obtain the average service time of a packet transmitted along both network links (VSAT-to-Hub and Hub-to-VSAT) giving VSAT-to-VSAT link delay.

The method of computation provided in this work rests on analytical study. However, the results obtained could be used as a design guide provided all parameters of the network are specified.

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