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The Feasibility of ATM operations over high frequency radio and the viability of the ATM/HF architecture

Paul Giovanni

THE FEASIBILITY OF ATM OPERATIONS

OVER HIGH FREQUENCY RADIO

AND

THE VIABILITY OF THE ATM/HF ARCHITECTURE

by

Paul S. Giovanni

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CERTIFICATE OF APPROVAL

Master's Thesis

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Dedication

To Melodye, a true friend and loving wife, whose support made it possible for me to finish Graduate School. This is the first step to a better life together.
"Love and always more love ... "

To my children Timothy, Tabitha, Danielle, and Paul who put up with a part time Dad and practically had to go through school with me. "Yes, children, I'm finally done."

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An Abstract

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ABSTRACT

High Frequency (HF) radio is still a vital part of communications networks because its low cost and long distance capabilities, and still plays important roles as primary, supplemental, or redundant backup systems. Asynchronous Transfer Mode (ATM) is increasingly becoming an important part of communications, especially with LAN Emulation (LANE) specifications. Add to this the importance and increasing interest and dependency upon wireless networking, and it becomes inevitable that research into mobile ATM networking over HF radio would be considered.

To test the feasibility of ATM networking over HF radio it was decided that a simulation would be developed to collect some basic information on call blocking and throughput. In order to build the simulation it was necessary to have an architectural framework of a mobile ATM network operating over HF radio. ATM/HF (ATM over HF) is the proposed architecture.

ATM/HF is a proposed architecture that provides for networking mobile ATM nodes such as ships, planes, and trucks, over HF radio. It is based upon a recommended 64 kHz bandwidth which allows for a 128 kbps data rate. The ATM/HF architecture utilizes three different Media Access Control (MAC) protocols for network startup and access from the various network states, and incorporates several recently proposed dynamic capabilities for control of bandwidth and the integration of voice, data, and video. The proposal provides frame and wireless ATM (WATM) packet structures and a

reference model for flow of the cells from the ATM Adaptation Layer (AAL) through the radio. An important feature is the use of channels, called channelization, to increase both network capacity and distance.

The simulation was built to represent an active network state with active nodes connecting and disconnecting calls in a dynamic way with explicit connection messages. The purpose of starting from this network state was to measure the call blocking and throughput of a single channel. Two user types were developed, one to represent telephone voice and the other to represent computer data traffic. By varying the number of users per node and by type, the level of call blocking and throughput could be changed. Graphing the levels it could be determined the maximum capacity a single channel could support and thus determine if ATM over HF radio is feasible. In addition, the same information was used to determine the viability of the ATM/HF architecture. Although the simulation did not incorporate all the dynamic features of the recommended protocols, it does dynamically assign slots, rearrange slots to utilize non-contiguous available slots, and adjust the data rate of computer connections to accommodate voice call requests. This was done to reduce the level of voice call blocking which became the determining factor in deciding feasibility.

It was determined that mobile ATM networking over HF radio is possible since the voice call blocking of a single channel was at the 10% level, overall call blocking was at the 6% level, and throughput was at the 53% level. It was determined that a single channel could support six voice and a minimum of ten data users. Although throughput, which is defined as the number of available slot used, was lower than expected, the possibility exists for utilizing the unused slots by incorporating additional dynamic capabilities that would increase the number of users supportable by a single channel. Throughput can be also be increased by incorporating Available Bit Rate (ABR) and Unspecified Bit Rate (UBR) traffic.

The call blocking and throughput levels prove that ATM/HF is a viable method for supporting ATM operations. Although the call blocking level achieved the voice call blocking level and exceeding the overall call blocking level, the throughput level shows that there is a lot of wasted bandwidth. Further study of the design is required to improve the throughput level. Further development of the simulation is required in order to test the MAC protocols and to test the effects of the Bit Error Rate and fading effects of HF radio. The final conclusion, however, is that ATM over HF radio is feasible, that ATM/HF is a viable architecture, and that further research should be conducted into both.

1 Introduction

1.1 ATM and High Frequency Radio

HF radio plays an important role in today's communications networks, especially within the military community where it is a vital part of the overall communications environment. As noted by J. R. Cleveland, "HF communications networks continue to serve an important role in linking critical command and control elements when other network facilities are unavailable," [1]. Other wireless environments, especially at the VHF and UHF frequencies, provide generous amounts of bandwidth for mobile operations as evidenced by their use in cellular applications. However, these higher frequencies are restricted to line of sight (LOS) operations requiring the use of many relay stations and expensive relaying devices, such as satellite, in order to communicate over long distances.

HF radio provides an alternative to LOS operations making research into its ability to support ATM critical. An inherent advantage of using HF radio is the ability to communicate over-the-horizon (OTH). This ability permits the establishment of wireless, mobile networks that can operate beyond the limit of LOS. In addition, the OTH capability is cost effective since it reduces the required number of relaying stations and devices, and in the case of smaller networks it may eliminate them altogether. HF radio, as mentioned by Cleveland, also provides a reliable, inexpensive backup communications system in the event other long distant services are lost. This is invaluable in time of critical need when loss of communications can have serious effects.

Besides the importance of HF radio in communications, ATM is rapidly becoming the switching network of choice because of its ability to integrate the heretofore separate information systems of video, audio, and data into one all inclusive network, reducing costs by reducing redundancy in equipment and software. As Jeffery Krieger states in his thesis on ATM Switching Architectures [2], " ... the flexibility and simplicity of ATM have made it widely accepted as the protocol for any future broadband network." He points out that,

> "ATM was designed to be a service-independent transport technique used to provide fast and efficient transmission of data from all forms of services over the Broadband Integrated Services Digital Network (B-ISDN), regardless of the type of data or the amount of bandwidth (within limits) required by the service" [3].

By taking advantage of ATM flexibility, not only standard information but, as John Walrod suggests, other non-typical sources of information such as SONAR and RADAR can be integrated into the network [4].

Of special interest and experimentation are mobile ATM networks operating in wireless environments. These types of networks consist of ATM and non-ATM sources combined into a network, or networks, utilizing ATM switches that are mobile and wireless. The network can include ships, aircraft, vehicles, and man-carried communications and information gathering devices. The mobility can be at the individual device level and at the switching backbone level. A good example of this is the RDRN experiment described in [5]. In this experiment, the network was mobile in that the switching nodes would move to a location and then establish connections with nodes. The node could change position by disconnecting from the network, moving to a new site, and then re-establishing the connections.

It is for these reasons, that is the growing importance of ATM and the vital necessity of HF radio, that HF-ATM networks need to be investigated, and this is the focus of this research. Asynchronous Transfer Mode over High Frequency radio (ATM/HF) is a proposed wireless ATM network design which seeks to adapt ATM operations to take advantage of HF radio's long distant, wireless, mobile environment.

1.2 ATM/HF

ATM networks operating in wireless environments, especially in the VHF and UHF band, have been the subject of much research [6, 7, 8]. However, very little work has been done in the HF band because it is considered to be too slow, too noisy, and too limiting in bandwidth. One purpose of this thesis is to provide a framework within which simulations and tests can be conducted on ATM networks operating in the HF radio band. ATM/HF is a broadcast network operating within the ground wave of the transmissions, and consists of mobile and fixed ATM nodes and stations. The ATM/HF network is part of an overall communications suite and as a consequence, can be used in conjunction with other communication systems, or as a backup to these systems.

Ad hoc operations are proposed to allow any node to start up and control a network, take over a network from a controlling node needing to disconnect, or restart a network in the event the controlling node is lost. These proposed ad hoc operations include Media Access Control (MAC) protocols that will reduce startup collisions and allow nodes to connect and disconnect from the network without interrupting ongoing connections. Although not a true distributed network where no node is in control, these capabilities provide an environment where dependence on one master station is avoided and where connection to the network can be accomplished with relative ease.

1.3 Focus of the Research

The idea of using HF radio as a medium for ATM networking is not new, but because of the problems associated with HF radio, such as high transmission error rates and restrictive bandwidth, interest is limited and research spurious. However, the advantages of having a long distance, wireless, mobile network, along with ATM becoming dominant in the communications industry, makes it imperative that research be conducted to determine if HF radio can support ATM operations and if an ATM/HF network can be integrated into existing ATM networks. The focus of this research is on answering this fundamental question of HF radio's ability to support ATM operations. As such, more detailed questions are used to guide the efforts. These questions are:

- Given the limited bandwidth inherent to HF, how many ATM nodes can adequately be supported on a single channel?
- Since each node supports multiple users, and given the same restriction on bandwidth, how many individual users from each node can a single channel adequately support?

- Would implementing the ability to dynamically specify bandwidth [9, 10] increase the number users that can be supported?

- Would allowing dynamic alteration of bandwidth [9] increase the number of users that can be supported?

There are two hypotheses in this study: 1) HF radio can support ATM operations; and 2) the proposed ATM/HF network design is a viable means of accomplishing this. The second hypothesis can be supported only when the first hypothesis is supported. Hence, there are three possible cases:

- The first hypothesis is rejected; i.e., ATM operations can not be supported by HF radio.

- The first hypothesis is supported, but the second one is rejected; i.e., ATM operations can be supported, but ATM/HF is not a good network design.

- Both hypotheses are supported; i.e., ATM operations can be supported, and ATM/HF is a good network design.

To test the above two hypotheses, a simulation of an ATM/HF network was developed from a basic description of ATM/HF in order to measure the amount of call blocking which would occur and determine the number of nodes and users a single channel can support.

1.4 Developing and Testing the Proposed Network

1.4.1 ATM/HF Network Design Issues

The design of ATM/HF needed to resolve several difficulties in order to be considered a workable solution to ATM operations over HF radio. The first was that of initial contention for the medium. Since there are times when none of the mobile ATM nodes will be broadcasting, there will eventually be a period of contention for initial network establishment and control of the medium. If two or more nodes decide to initiate network startup at the same time, contention occurs and startup is delayed. This situation is similar to that found in unslotted ALOHA where two or more nodes attempt to access the same slot at the same time. The proposed ATM/HF MAC protocols needed to reduce this type of contention and, if possible, eliminate it.

Another difficulty requiring resolution was that of access contention on an active network. In ATM/HF, the first node gaining access to the medium becomes the controlling node and transmits frame timing via the frame sync signal. This node controls slot assignment and initially assigns itself a slot assignment for the call generating the startup of the network. All other nodes must now contend for access and reserve bandwidth via the available frame slot that follows the frame sync signal. The proposed MAC protocols needed to support this type of contention and bandwidth reservation requests also.

Handoff of the frame sync signal is another difficulty that needed to be cleared up. ATM/HF has no permanent controlling station for maintaining frame sync or assigning bandwidth. The station performing these functions can terminate transmission if it no longer requires a connection. However, before it can disconnect from the network, it must pass control of the network to another node still utilizing the medium. Although not part of the present simulation, this is mentioned because of reliance on the frame sync signal for timing, media access, cell slot alignment, and contention control.

Another problem faced was minimizing the amount of contention caused by the limited available bandwidth. Once a node gains access to the network, it should be able to request an increase in the bandwidth of an existing connection without having to contend for that bandwidth. This is the dynamic bandwidth assignment mentioned earlier. In addition, nodes should be able to request and reserve additional bandwidth for new connections without having to contend via the available slot. The proposed solution including sending explicit management messages and using metasignaling within the headers of the packets of existing connections. However, metasignaling takes time to accomplish and explicit management messages cause the loss of packets. For the simulation, explicit messages were used.

Finally, since ATM/HF is designed to operate in a quasi-distributed manner, slot assignments and bandwidth control resides with the controlling node. In the proposal, the controlling node is given the ability to assign less bandwidth than requested in order to let more users onto the network, increasing throughput and efficiency. Also recommended is the capability to accommodate priority requests. In the current version of the simulation, the feature of accommodating priority requests is not tested because of the complicated nature of the process. This feature is a good candidate for incorporation into the next version of the simulation. A version of the feature allowing high bandwidth data transmissions to be reduced was incorporated in an attempt to reduce call blocking on voice calls.

1.4.2 Specific Network Features

For this particular research, it is assumed that an increase in bandwidth from the present 4KHz is authorized. The required bandwidth for the simulation is 64 kHz that allows a bit rate of 128 kbps. The specified channel rate is 9.6 kbps that represents the utilization of one slot in each frame for one simplex voice connection.. How this channel rate was chosen is discussed in more detail in chapter 2.

One feature of ATM/HF is a modified form of the available slot as presented in [10]. The available slot is the slot in each frame used for media access by nodes not already connected to the network. This protocol utilizes a form of the Reservation ALOHA (R-ALOHA) access procedure in which the unit gaining access to the slot uses it to request a specified bandwidth. The controlling station replies with slot assignments and the available R-ALOHA slot becomes available for access again. The proposal in [10] is to dynamically increase the number of available slots as the number of stations attempting access increases, thus reducing collisions and access delay. However, because of limited bandwidth in ATM/HF, it is impossible to increase the number of available slots per frame. Instead, the available slot starts out as a percentage such as one slot every fourth frame. This number is then increased or decreased based upon access attempts, however, it will never increase beyond one slot per frame, or drop below one slot per second (every 25 frames). The slots not used for contention can be utilized for low data rate connections. The identification of the available slot will be broadcast within the frame sync signal, or as an explicit management message by the controlling node. This feature is not tested in this version of the simulation. However, it is planned to test this MAC protocol in the next version of the simulation.

The following is a list of requirements taken into consideration in developing the ATM/HF network design. Those features marked with an asterisk (*) have been included in the current version of the simulation either in its entirety or in a simplified version. The ATM/HF network must:

1. Operate in a distributed fashion, as previously discussed
2. Utilize a bandwidth reservation scheme*
3. Allow for the dynamic changing of bandwidth and QoS of an existing connection*
4. Allow for additional channel requests without additional contention*
5. Dynamically adjust the contention period to accommodate more or less nodes
6. Minimize access contention and collisions (follows from 3, 4, & 5 above)
7. Maintain maximum data throughput*
8. Use minimal overhead per frame (different from WATM Packet overhead which is fixed)
9. Minimize cell access delay and cell loss
10. Be easy to implement

11. Provide easy integration into existing ATM networks
12. Minimize blocking of voice calls by reducing rate on or temporarily blocking data connections*

1.4.3 Testing the Proposed Network Design

In order to test the viability of the ATM/HF network design, a simulation was developed. The simulation was used to collect data on call blocking. This data was used to determine whether enough nodes and users could adequately be supported on a single channel in an ATM/HF network, and whether ATM/HF was a workable, realistic design.

In order to facilitate simulation development, several assumptions needed to be made. First, the network will already have gone through the initial startup phase and there is a controlling node and active media. Second, the frame structure was modified from that specified because of the nature of the simulation software. Third, all management messages are explicit in order to facilitate simulation development and test the network. Although the simulation was not capable of testing all the features of the network design, it provided enough data for determining the effectiveness of the ATM/HF network design.

The simulation only tested for call blocking in order to determine the maximum number of nodes and users that the network could support. Although the design calls for dynamic bandwidth reservation and allocation, these features were not included with this version of the simulation. This is because of the need to get the simulation operational and as with priority requests, this is left to the next version for implementation. The purpose of simulating the ATM/HF network is to determine its feasibility. The additional capabilities mentioned can be added to the simulation in order to determine if these dynamic features, such as the MAC protocols and dynamic bandwidth allocation, can improve network performance from that of the present simulation.

The simulation itself consists of a number of nodes with a set numbers of users in each node. As mentioned, only an active network was simulated, and MAC protocols were not tested. The data collected consisted of the number of nodes on the network, the number of users per node, the number of voice and data call requests made, the number of these calls that were blocked, the number of slots available for user data, and the number of slots used. These figures were used to calculate throughput and call blocking.

The only automated features of the simulation were slot assignment and reduction of data call bandwidth to accommodate voice calls. The users consisted of Constant Bit Rate (CBR) Generators representing voice calls, and Available Bit Rate (ABR) Generators representing data connections. The CBR generators were assigned permanent connections of two slots per frame and ABR generators were assigned either 1 or 4 slots per frame depending on slot availability. As mentioned, data calls assigned a bandwidth of 4 slots could be reduced to 1 slot for the remainder of their connection in order to accommodate voice calls, reducing the number of blocked voice calls.

The following chapters present a more detailed discussion on the ATM/HF architecture and network design, and the MAC protocols. Detailed descriptions of the simulation and a presentation of the test results with analysis follow these chapters. The final chapter provides the conclusions and recommendations for further study.

2 ATM Over HF Radio (ATM/HF)

This chapter and the following chapter on Media Access Control (MAC) Protocols are the foundation for the development of the ATM/HF network simulation. Although not all parts of the network architecture discussed could be incorporated into the current version, the detailed description provides enough information for building a simulation of the network, and for building an actual network which is the ultimate goal beyond the scope of this paper.

This chapter starts with a more detailed look at the overall structure of the network and how it fits into current communications suites and networks. In the following sections, the network architecture, network reference layer model, network frame structure, and wireless ATM (WATM) packet are presented.

2.1 ATM/HF Network Architecture

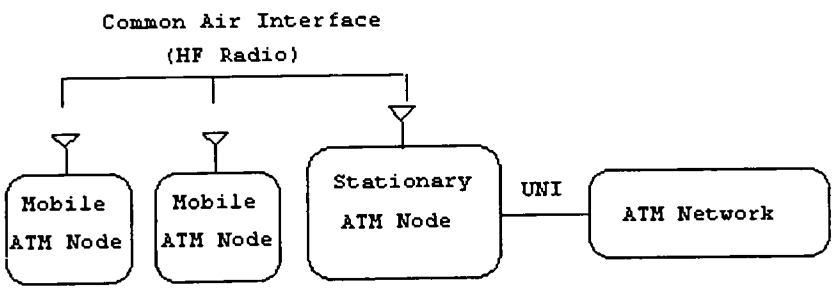


Figure 2.1 - ATM/HF System Architecture

ATM/HF is a wide area network design utilizing HF radio as the wireless transmission media interface over which mobile ATM networks operate. Figure 2.1 illustrates the basic system architecture which consists of mobile ATM nodes connected via a common air interface over HF radio. Also included are stationary, or fixed, nodes as part of the network. These fixed nodes allow ATM/HF to be connected, via a User-to-Network Interface (UNI), to other private or public networks.

The network operations are based upon a type of distributed control of the media, and upon ad hoc network establishment. Ad hoc establishment permits any node in a given area to initiate a network, control the frame sync signal used for network timing, and perform slot assignments. The MAC protocol recommended for this type of network startup is a variation of the IEEE 802.11 CSMA/CA protocol under consideration for wireless LANs [11, 12]. Once the interface is established, the MAC protocol changes to the use of an explicit reservation type.

An important advantage to using distributed control is that any node can assume control of the media and slot assignment upon request of the controlling node, and any node can re-initiate the network if the controlling node suffers a casualty. This type of dynamic, ad hoc networking design allows for impromptu network establishment and disestablishment, and means the network and the air interface need only be in operation when required.

The ATM/HF network also includes provisions for channelization so that the capacity can be increased and the distance extended. With the capability of increasing network capacity the limited bandwidth inherent with HF radio can be overcome, and by extending the reach of the network a wider area of coverage becomes possible.

Channelization takes advantage of the ATM switch for switching cells to the different modems for transmission over the appropriate circuit. Figure 2.2 illustrates the concept of a two channel network.

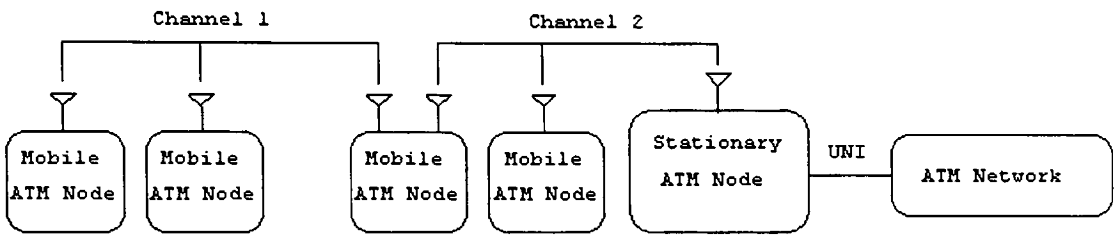


Figure 2.2 - Channelization in ATM/HF

As seen in Figure 2.2, the network is extended to reach a fixed station for connection to landbased networks. A third channel could be added to support the first to reduce call blocking due to its becoming full or to seperate voice from data so that data users would access one channel while voice users would utilze the other.

Conceptually, ATM/HF is integrated into a larger communications suite which includes satellite communications (SATCOM) and UHF/VHF line-of-sight (LOS). The block diagram in Figure 2.3 depicts a typical communications suite with ATM/HF and with the

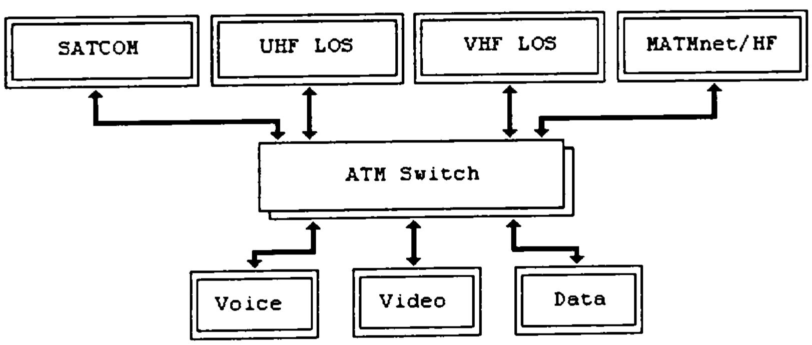


Figure 2.3 - Integrated Network

ATM switch as central to all the communications networks. This design provides a measure of redundancy and casualty recovery which are vital to mission critical, mobile communications, especially in hostile environments.

The mobile ATM node is a self contained local area ATM network consisting of voice, video, and data components. Figure 2.4 shows this integration of the various information sources in a basic ATM/HF LAN with connection to the HF communications equipment. The HF configuration itself consists of a transmitter and receiver pair connected to the ATM switch via an ATM ready modem. The reason for using a transmiter/receiver pair instead of a single transceiver is to allow the transmitter to remain in the transmit mode without having to switch between transmit and receive. This prevents the loss of data due to the time required for HF radios to switch from receive to transmit and also permits the transmitter to keep up with the data rate and slot assignments. The use of a dedicated receiver makes full duplex operations possible. More than one transmitter/receiver pair is required to provide for channelization and for casualty recovery in the event of equipment failure.

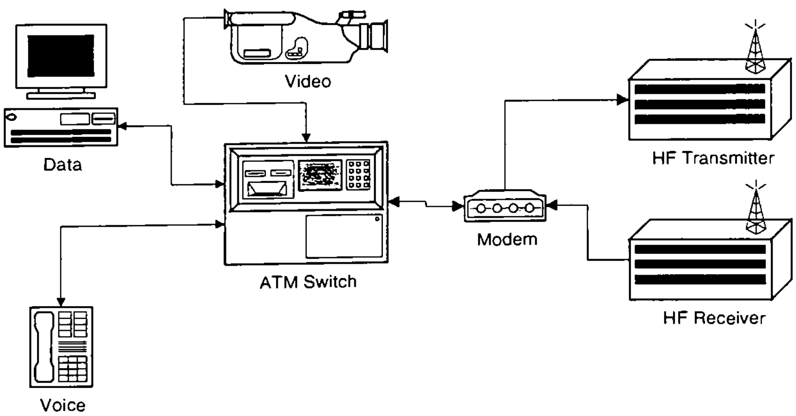


Figure 2.4 - ATM/HF Basic Node Architecture

The operational bandwidth for ATM/HF is specified at 64 kHz in order to provide a 128 kbps data rate. The mode of operation is single sideband (SSB) with low power output (100 watts maximum). This is based upon the premise that ATM/HF is geographically limited to the ground wave and that only the necessary amount of power will be used to establish effective communications. The power limitation is considered an important part of protecting the transmission from interception. It is expected that the transmitter be capable of adjusting its power output based upon feedback from the network.

2.2 The ATM/HF Reference Model

Figure 2.5 depicts the four fundamental layers of the ATM/HF reference model showing the primary functions and their location. Uyless Black provides a good definition of the purpose of a layered reference model when he writes, "The important point to understand is that, at the receiving site, the layer entities use the headers created by the peer entities at the transmitting site to implement actions" [13].

LOCATION	LAYER	FUNCTIONS
Conversion of data to cell format, SAR and Convergence Sublayers	AAL	Software/Hardware exteral to the ATM Switch
ATM Switch	ATM	Switch and Connection Access Control (CAC)
Modem	WATM	Create WATM packet Perform CRC, FEC, Cell sequencing, Media Access Maintain Network Status
Modem and Radio	PHY	Converts data into media specific format, network access, and link establishment

Figure 2.5 - ATM/HF Reference Model

Again he writes, "The idea of the Model is for peer layers to communicate with each other." Using this definition of how a layer model works, an examination of the ATM/HF reference model is now presented.

2.2.1 The ATM Adaptation Layer (AAL)

The AAL is used to interface between the user application and the ATM switch. It converts application specific protocol data units (PDU) into the ATM cell format of 48 octets per cell. There are two sublayers to the AAL, the Convergence Sublayer (CS) and the Segmentation and Reassembly (SAR) sublayer. The job each sublayer performs depends on the class of traffic being supported. The CS is responsible for collecting the different traffic classes and from them forming the ATM cells which can be either 46 or 47 octets in length. This cell is passed on to the SAR sublayer. Conversely, the CS receives cells from the SAR sublayer, puts them into the correct form for the applications, and forwards them.

The SAR sublayer provides the headers and trailers to the ATM cell. These can include some or all of the following: Sequence Numbers, CRC, Length Indicators, and Information Types or Message Identifiers. This additional data brings the cell to a total of 48 octets. The cell is then passed onto the ATM layer for addition of the 5 octet header which brings the total cell size to the specified 53 octets.

The AAL is independent of the ATM switch and can be implemented in a variety of ways. For example, it can be located in a Network Interface Card (NIC) or Network

Interface Unit (NIU) [14], in a Terminal Adaptor (TA), or in the users equipment. It can be either hardware, such as the NIC, software, such as a program on the users computer, or a combination of both.

2.2.2 The ATM Layer

The ATM layer performs Call Access Control (CAC) to determine if it can process a request for a connection. It is responsible for adding the 5 octet header to the ATM cell, and for the high speed routing and switching of the cells. With regard to ATM/HF, the switch queries the modem to determine if there is bandwidth available for the connection. If not, the connection is rejected (blocked) or re-routed.

2.2.3 The Wireless ATM (WATM) Layer

Figure 2.6 shows the flow of the cells through the layers and the formation of the WATM packets. The WATM layer is not part of the ATM layer but follows it so that the ATM

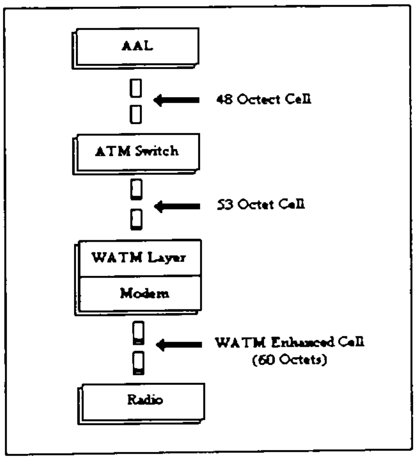


Figure 2.6 - Cell flow through the layers

switch can concentrate on switching functions. The WATM layer is responsible for forming the WATM packets and can be hardware or software. It is preferable, however, that it be a combination of hardware and software within the modem. The WATM layer performs the Cyclic Redundancy Check (CRC) and Forward Error Correction (FEC) functions, adds sequencing to outgoing cells, and compresses the ATM Header. These steps are required to maintain communications over a wireless network.

In ATM/HF, the WATM layer is responsible for queueing the WATM packets and for tracking the Cell Loss Ratio (CLR) of the HF portion of the network. This CLR is reported to the ATM layer which tracks the overall CLR for the entire node. The queueing function performed by the WATM layer relieves the ATM switch of having to store the cells. It is the WATM layer that responds to queries from the ATM layer about available bandwidth. It is also the layer that is responsible for the retransmition of WATM packets in response to a request from other nodes.

On the receive side, the WATM layer performs CRC and FEC and checks the sequence number. Those packets that are good or have only one bit errors are forwarded, but corrupted packets are discarded. For missing packets, it sends the requests for selective retransmission, however, the applications are responsible for requesting the retransmission of discarded packets. Finally, WATM layer strips layer information from the packets and forwards them as ATM cells to the ATM switch.

The number of WATM packets requiring Forward Error Correction (FEC) and the number of WATM packets lost, expressed as the Cell Loss Ratio (CLR), are important peices of information because they represent the quality of the link. These numbers are used to trigger management functions to improve the connection. It is the responsibility of the WATM layer to track these numbers for the radio portion of the network and for sending management messages to the appropriate managment functions in order to maintain link quality.

2.2.4 The Physical Layer

The Physical Layer encompasses the modem and radio equipment. It is at this level that initial media establishment takes place, where slot reservation contention occurs, and where media management transpires. The network has two operational states: STATE_1 where the media is inactive and STATE_2 where the media is active, and nodes have three states. These various states will be discussed further in chapter 3, but for now, it is important to note that for network STATE_1, the Physical Layer is responsible for network startup and for maintaining the frame sync signals which control the timing of the network.

It is apparent that the modem is the heart of the ATM/HF network, being central to the majority of management and operational controls and procedures. It is responsible for creating the WATM packets, for queuing, and for controlling the timing of the data transmissions by monitoring the frame sync signals and sending the queued the packets during the appropriate time slots. It is also where FEC is performed and where the statistics for CLR are collected.

2.3 The ATM/HF Frame

The structure of the frames used in a network are key to the efficient use of the available bandwidth. Frames rates are based upon the rate of the voice circuit which is one slot per frame for as many frames as it takes in a second to provide the full data rate. For example, the data rate for ISDN voice is 64kbps and a cell holds 48 octets. With this information the number of slots can be calculated as following:

(1) 64kbps / 8 = 8kBps

(2) 8kBps / 48 octets per cell $\cong$ 167 cells

This means that it would take 167 frames per second to support a voice channel rate of 64kbps. The voice channel rate is: 48 * 8 * 167 = 64.128kbps.

The number of slots per frame is another factor related to bandwidth – the more bandwidth available, the more slots per frame. For a rate of 1.544Mbps, a cell size of 53 octets, and a frame rate of 167 frames per second (fps), the number of of slots per frame is calcualted as follows:

(3) 1.544Mbps / 8 $\cong$ 193kBps

(4) 193kBps / 53 octets per cell $\cong$ 3641 slots

(5) 3641 slots / 167 fps $\cong$ 21 slots per frame

Therefore, at 167 fps with 21 slots per frame and 53 octets per slot, the total bit rate of the link is: 53 * 8 * 21 * 167 = 1.487Mbps with 57kbps left for other uses such as frame headers and trailers. The 1.544Mbps link can, therefore, provide 21 voice channels. These same concepts can now be applied to the ATM/HF bandwidth of 128kbps to determine frame rate and number of slots per frame.

The first piece of information needed to calculate the number of frames per second is the rate required for a single voice channel. Using the IS-95 data compression rates as a guide, it was decided that the operational rate for a voice channel would be 4kHz with a gross bit rate of 9.6kbps [15]. There are other rates, both faster and slower, but the guiding factor was to get as many voice channels onto a single circuit as possible without losing data to bit errors. A lower compression rate would increase the number of frames per second, consequently reducing the number of slots per frame. A higher compression rate could cause a bad connection if the BER is high, since a discarded cell would contain more data than the 9.6kbps rate. If it becomes possible to reduce the BER in HF radio, then it may be possible to reduce the operational rate of the voice channel to 2kHz with a gross bit rate of 4.8kbps. Also, for the simulation it was decided that full duplex voice connections should be emulated, therefore, each time a voice call connects to the network it is assigned two slots, one for each caller. It may be possible to operate in a half duplex mode where a single channel is shared by both callers, but that is left to further study.

So, by specifying the voice channel rate at 9.6kbps, the number of slots required per second can be calculated as:

(6) 9.6 kbps / 8 bits per byte = 1200 bytes per second

(7) 1200 bytes per second / 48 octets per ATM cell = 25 cells per second

With this number it is clear that it would take 25 cells per second (cps) to support one voice channel. Therefore, it would take 25 frames per second to maintain a voice channel rate of 9.6 kbps.

The next step is to find the maximum number of slots per frame. ATM/HF uses a 60 octet WATM packet which means that each frame slot must be 60 octets. Using this information the number of slots available per frame is calculated at:

(8) 128kbps / 8 octets per byte = 16kBps

(9) 16kBps / 60 octets per slot = 266.667 ($\cong$ 267 slots per second)

(10) 267 / 25 $\cong$ 10 slots per frame

ATM/HF can, therefore, operate at 25 fps with 10 slots per frame. This provides a voice channel rate of: 48 * 8 * 25 = 9.6kbps and a total bandwidth of: 60 * 8 * 10 * 25 = 120kbps which leaves 8kbps for other uses such as guard times and sync signals. One slot will always be available for contention which allows the frame structure of ATM/HF to support a total of 4 full duplex voice channels and one 9.6kbps data channel per frame over a single circuit.

Using the numbers obtained the basic ATM/HF frame structure can be created. Figure 2.7 shows the frame structure in terms of octets with the addition of the Guard Time (GT) and Frame Sync Signal. The GT is required to account for propagation delay allowing the furthest nodes in the network to receive the full frame before the next frame begins.

```
|<—                    Total of 640 Octets                    —>|
|GT|SYNC Signal|     SLOT 1     |SLOT 2|SLOT 3|...|SLOT 10|
|<- 40 Octets ->|<-60 Octets->|
```

Figure 2.7 - Basic ATM/HF Frame Structure

This prevents collisions between the beginning of a frame with the end of the last frame. The GT also determines the maximum size of the network. Radio frequency travels at approximately 6.2 microseconds (usec) per nautical mile (2000 yards). A GT of 1 octet provides a range of 10 nautical miles. By allowing the GT to be 8 octets, the range of the network is extended up to 80 nautical miles, which for most vessels may be well beyond the ground wave of the HF signal. The GT is not specified at this time, so for the simulation the GT and Sync signal are represented by an eleventh slot in the frame.

2.4 The ATM/HF WATM Packet

The WATM packet structure, as shown in Figure 2.8, is 60 octets to accommodate the ATM cell with compressed header and the required overhead needed for operating in a wireless environment.

```
|←              Total of 60 Octets              →|
|GT|SYNC | SEQ| ATM Header| ATM Cell|  Trailer   |
|←6 Octets→|←      51 Octets      →|←3 Octets→|
```

Figure 2.8 - ATM/HF Packet Structure

The WATM packet is divided into the header, user data, and FEC trailer. The header provides guard time, slot sync, and a packet sequence number and is 6 octets in length. The header description is as follows:

- GT – Guard time used to separate the slots in order to prevent collisions.
- SYNC - Defines the start of the slot and keeps timing between the two nodes.
- SEQ Sequence number of the packet. This number is used by the system for selective retransmission requests.
- ATM HDR – This contains the compressed verion of the ATM cell header.
- ATM CELL – This portion of the packet contains the 48 bytes of user data.

- FEC - used by the modem to detect packet corruption and allow for the correction of one bit errors.

Encapsulation of the ATM cell in a WATM packet is required to accomodate the wireless environment. The Bit Error Rate (BER) of the HF environment is normally around 10e-3. This is very high for an ATM network, and would contribute to a large loss of cells. By limiting the geographic area of the network to the ground wave and by allowing dynamic parameters such as power and frequency adjustment, it may be possible to improve the BER. But even with a better BER, there will still be the need to overcome corrupted packets in the attempt to reduce the CLR of the network. One bit errors are corrected by using the CRC and the FEC trailer, and the SEQ number is used by the modem to request lost packets without having to have entire frames retransmitted. Although 5 octets larger than the data cell size proposed by the NEC USA C&C Research Laboratories [16], the number of octets can not be reduced without affecting the quality of the connection.

In this chapter a detailed overview of the ATM/HF network structure has been presented. This design is the overall framework within which the ATM/HF network simulation was developed. In the next chapter, the role and purpose of MAC protocols will be discussed and the recommended ATM/HF MAC protocol presented.

3 ATM/HF Media Access Control Protocol Specifications

Media Access Control (MAC) is the key to every network. Without the ability to gain access to the media no communications occur, and without control of access the network becomes a mass of colliding data, rendering it useless. ATM/HF faces a unique situation in that three MAC protocols are required: one for initial network startup, one for an inactive node attempting to access an active network, and one for a node with an active connection desiring another connection. In this chapter three MAC protocols will be presented based upon the state of the network and nodes. These states and the ATM/HF MAC specifications will be defined to provide the framework for the MAC protocol designs.

The chapter begins with a brief discussion of the problems ATM/HF MAC protocols need to address and remedy. The sections that follow are on the ATM/HF network and node states and the MAC protocol specifications. The final section outlines the relationship of the MAC protocols to the simulation.

3.1 ATM/HF MAC Problems

Using HF radio as the medium for the backbone of a mobile ATM network presents several unique design problems, especially with respect to media access. The first concerns media access during the initial network startup that consists of contention for and establishment of the media. When a network is inactive and a node desires to connect with another, the node begins a process called STARTUP. However, if two or more nodes attempt STARTUP simultaneously, there will be a collision with the subsequent delay of starting the network. This situation is similar to that found in unslotted ALOHA and the IEEE 802.3 CSMA/CD networks where packets collide and delay occurs. If this situation is not controlled it could prevent the network from ever being started. The first ATM/HF MAC protocol is specifically related to this process and must control it in order to reduce or eliminate collisions.

The second problem the MAC protocols must resolve is the accessing of an active network by an idle node. An idle node is one that has no active connection on an active network, and an active network exists once the STARTUP procedure has completed and a sync signal is present. Once the network becomes active and until it is disestablished, all nodes will need to use a second MAC protocol for gaining access to the media.

A third problem the ATM/HF MAC protocols face is requests for new connections made by nodes with active connections. Active nodes will be allowed to request slot assignments for new connections via meta-signalling or explicit management messages. One purpose of this is to keep the number of slots needed for contention to a minimum. However, the potential for contention still exists because several nodes may request an aggregate bandwidth that is more than what is available. In this situation, it may not be possible to accommodate the bandwidth requested by each connection. The third MAC protocol is designed to resolve this situation in the most favorable way possible, however, it may mean that a connection gets blocked if it can not be otherwise accommodated.

Beside the problems mentioned in the previous paragraphs, there are other capabilities that the MAC protocols must support. One concerns explicit bandwidth reservation requests made by a node during the network access attempt. This capability allows accessing nodes to specify the minimum and maximum amounts of bandwidth, in terms of cell slots, required for the specified connection Quality of Service (QoS). The explicit reservation scheme allows for more bandwidth to be allocated to a connection than could be assigned by simply using contention alone, and it allows for variable bandwidth assignment by the controlling node including providing a connection at a rate lower than specified in the request. This scheme allows increased access by a majority of data applications since they can operate at lower rates than the voice channel rate of 9.6kbps. It also allows for the sharing of specific slots, which is the subject for further study.

Another capability the MAC protocols must support is Dynamic Bandwidth Allocation (DBA). This capability allows the nodes to dynamically increase or decrease the bandwidth of a connection during its lifetime. This scheme is new and is related to the new Variable Bit Rate (VBR) service class under consideration for ATM networks. By incorporating this into the network design at this point, it will be easier to implement the final design once it becomes published as a standard. DBA is designed to eliminate VBR queuing problems caused by the variableness of the data rate from the source.

Finally, ATM/HF is designed to operate in a distributed manner which means any node can startup a network and assume the duties of the controlling node upon request. Although it is possible to use a more decentralized algorithm where slot assignment is made by all nodes on the network running the same algorithm at the same time, for the purpose of this paper only the controlling node will control the sync signal and be responsible for assigning slots. For the simulation, slot assignment will be controlled via explicit management control packets. Although not a specifically MAC related capability, it is important to understand that all nodes in the network must have the capability of running the MAC protocols that control network access and slot assignment.

3.2 The ATM/HF States

There are two types of ATM/HF states, the state of the network and the state of the node. Together, these states are used to determine which MAC protocol should be used. The network itself has two states that it can be in, an idle state (NET_IDLE) and an active state (NET_ACT). The NET_IDLE State is characterized by the fact that there is no active media present. And, conversely, an active media characterizes the NET_ACT State. The nodes are always in one of three states beginning with the idle state (NODE_IDLE). There is one transitional state through which the node moves through referred to as NODE_STARTUP. This state is used if no network exists. From the NODE_IDLE state the node can either go through the NODE_STARTUP state if the network is NET_IDLE, or directly to the inactive state (NODE_INACT) if the network is active. From either of these states, the node moves to an active state (NODE_ACT) by establishing an active connection on the network. From any state, the node can go back to the NODE_IDLE State. Figure 3.1 illustrates the concept of moving from state to state.

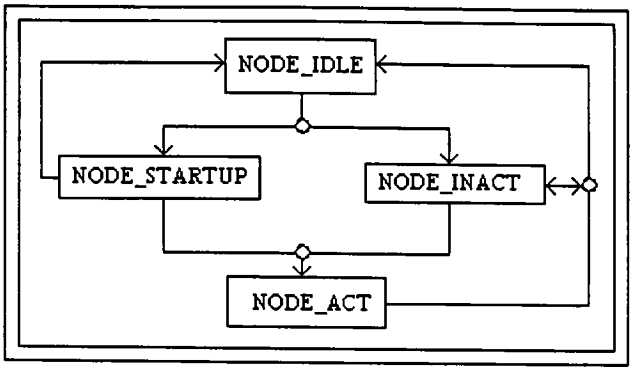


Figure 3.1 - Node State Diagram

The reason for defining these different states is to use them to identify the method for media access. For example, if the network is NET_IDLE, then the node will also be NODE_IDLE. If a node needs to make a connection, it moves to the NODE_STARTUP State for starting the network. If successful, the node moves to NODE_ACT and the network moves to the NET_ACT State. All other nodes on the network would also change states to the NODE_INACT state from which they can access the active network.

If a node is NODE_IDLE and another node starts up the network (NET_ACT), then the node moves to the NODE_INACT State, even if it currently has no intention of connecting to the network. From this state it can connect to the active network or move back to NODE_IDLE when the network goes NET_IDLE. If the network is NET_ACT and an inactive node (NODE_INACT) wants to connect, it does so by using the MAC procedure for the NET_ACT State. If the attempt to access is successful, the node moves into the NODE_ACT State. Once a node is in the NODE_ACT State it uses the third type of MAC protocol for dynamically establishing new connections.

As described above, each combination of network and node states involves a different type of MAC procedure for accessing the network. NODE_STARTUP uses a carrier sensing protocol collision avoidance type of procedure and the NODE_INACT State uses an explicit reservation contention method similar to Slotted or Reservation ALOHA. The NODE_ACT State uses a dynamic connection establishment procedure based upon the Priority Oriented Demand Assignment (PODA) procedure [17, 18]. In the next section a more detailed description of the AMT/HF MAC specifications is presented.

3.3 MAC Protocol Specifications

With a basic understand of the network parameters from the previous chapter and the states from the description above, the following eleven specifications are presented as the core requirements for the ATM/HF MAC protocol. These specifications will be used to the measure the ability of existing protocols to be used in ATM/HF as they now operate, and to guide their modification if that is necessary. The basic MAC specifications are as follows:

a. The protocol must allow any node to startup a network.

b. The protocol must accommodate bandwidth reservation to support QoS requirements and allow efficient utilization of bandwidth.

c. The protocol must accommodate dynamic bandwidth allocation to support variable bit rate connections.

d. The protocol must support dynamic connection requests, working in conjunction with call access, in order to keep the contention period to a minimum.

e. The protocol must be capable of dynamically adjusting the number of available slots for access contention in order to keep collisions to a minimum and reduce access delay.

f. The protocol must minimize startup delay.

g. The protocol must maintain maximum data throughput.

h. The protocol must assign slots to minimize frame overhead by keeping all connections from one node together so that all data can be broadcast in one burst.

i. The protocol must minimize cell loss from full queues by allowing for the dynamic adjustment of bandwidth.

j. The protocol must be easy to implement. That is, it must be capable of implementation with minimum software complexity and cost, and allow existing communications suites to be easily adapted.

k. The protocol must provide easy integration of ATM/HF into existing wireless and wired ATM networks.

3.3.1 Initial Media Establishment

The state of the network and node will initially be an idle state where no active network exists. In this state, the primary problem facing the ATM/HF MAC protocol is reducing startup delay. In addition, the protocol must accommodate the algorithm, which allows any node to startup the network. With this in mind, it is determined that the MAC protocol must be a carrier sensing type of protocol since the current state of the network and node will require sensing prior to broadcasting. In the NET_IDLE state there is no active media, so the initial startup procedure must make it clear, to all nodes in the area, that a node is attempting to startup a network. The protocol must ensure that another node does not interfere with the attempt to startup a network. For this reason, a long sync signal is specified to tie up the media and stop other nodes from attempting startup. The sync signal will be long enough to overcome any propagation delay.

Another factor to reducing startup delay is detecting collisions, which is difficult for the transmitting node to do. To allow the transmitting node to detect a collision, the initial startup frame will allow for the broadcast of a collision message from an idle node that detects a collision. This helps in preventing hidden nodes, which are nodes that have a long propagation delay in relation to other nodes attempting startup, for creating multiple collisions on a network. This problem, however, is beyond the scope of this thesis.

Based upon these two ideas of long sync signals and the need to notify nodes of a collision, the initial startup frame is presented in Figure 3.2.

```
|<--          Total of 640 Bytes (5120 bits)          -->|
| SYNC_SIG      | WAIT_TIME     | SYNC_SIG     |NET_MAN_INFO|
|<-168 Octets->|<-168 Octets->|<-56 Octets->|<-56 Octets->|
    3 Slots         3 Slots        1 Slot         1 Slot
```

Figure 3.2 - NODE_STARTUP Frame

The startup frame is different from the normal frame. It contains a long sync signal to reach the maximum limit of the geographic network, overcoming propagation delay and alerting all nodes of the start up attempt. Next, the frame provides a waiting period for a collision message. If no collision message is heard during the WAIT_TIME, the node continues with the rest of the frame and assumes control of the network. If a collision message is heard, then the MAC protocol will generate a random backoff time, wait that time, and then retry startup. If another SYNC_SIG is heard before transmission begins, then the node will abort the startup process. If in the backoff period after a collision, and a SYNC_SIG is heard, again the startup process is aborted. Once the network is NET_ACT, the node goes into the NODE_INACT State.

Idle nodes, upon hearing the startup sync signal, will not attempt the startup procedure even if they get a request to connect. Instead, they wait until the media is established, change to the NODE_INACT State, and then attempt access using the MAC protocol for that state. If an idle node hears a collision occur, it must transmit a collision message so that the startup nodes can go into a backoff period. Of course, this leads to the possibility of even more confusion if more than one idle node attempts to broadcast a collision message. To simplify this thesis, if any type of broadcast is heard during the WAIT_TIME, even if garbled, then the nodes go into a backoff period. If the media is clear at the end of the backoff period, then the node begins the startup procedure again by sending another NODE_STARTUP frame and following the procedures as outlined above.

3.3.2 Accessing an Active Media

The access procedure just discussed is for the NET_IDLE/NODE_IDLE states. The access method for the NET_ACT/NODE_INACT states is different since the media is already active. There are several methods which could be employed, the first being straightforward S-ALOHA where nodes simply pick a time slot and broadcast their data into the slot. This is not very efficient and therefore not considered appropriate for

ATM/HF. Another MAC protocol is Reservation ALOHA (R-ALOHA) where contention is the same as S-ALOHA except that the slot remains the possession of the node that gains control until given up by that node. This protocol, however, does not meet the requirements for ATM/HF since it only allows a connection to occupy the slots it obtains through contention.

The best way of providing access is by using a contention period and allowing access only during that period. The contention period can be either a full slot, or a smaller control slot. With either the full slot or the control slot, a control message is sent by a contending node requesting slot assignment and providing bandwidth and QoS information.

In comparing the two contention types, it was decided that a form of the Packet Reservation Multiple Access with Dynamic Allocation (PRMA/DA), which will be discussed in the next chapter, would be used. This protocol uses full slots, referred to as available slots, for contention instead of smaller control slots. The distinctive characteristic of this protocol is the dynamic allocation of available slots. That is, the number of available slots varies according to the number of access attempts occurring. The reason for choosing PRMA/DA is that it is a clean and simple protocol that is easy to incorporate into an ATM/HF network.

It is during initial access that the dynamic reservation request is made. The initial access message includes several required parameters so that an appropriate number of slots can be assigned to meet the QoS specifications of the connection. The three most important parameters needed, following the PRMA/DA protocol, are the average data rate, the peak data rate, and the requested number of slots for the initial connection. Using this information, the controlling node calculates and returns the initial slot assignments.

Once assigned, the slots are controlled by the node to which assigned until no longer needed. In ATM/HF, this means the node must explicitly notify the controlling node that is disconnecting so that the controlling node can reassign the slots. This is accomplished by sending an explicit disconnect message to the controlling node. A potential problem exists in that a node can refrain from giving up its slots or it can suffer a casualty which prevents it from sending the disconnect message. For this thesis and the simulation, however, this problem will be assumed to not exist and it is recommended that further study be conducted on this subject.

3.3.3 Dynamic Access for New Connections

The third type of MAC protocol is related to the NODE_ACT State. In this state, the node already has at least one active connection and is seeking to add another, but instead of contending during the contention period it uses an explicit management message or meta-signalling in the packet header to request slot assignments for the new connection. The information provided is the same as that in the other type of access message used in the previous section. Meta-signalling is a way of sending information in band and should not cause access delay. In ATM/HF, explicit messages can cause the loss of cells. This type of dynamic call access is discussed in more detail in the next chapter.

Contention for this state comes if other nodes also require connections. In this case, the controlling node must make a decision as to how many slots to assign or if it must block a request. The algorithm includes the ability to adjust existing connections, such as reducing a two-slot connection to one slot, in order to free bandwidth for higher priority requests. The blocking of a call request, although not a desirable choice, may be necessary depending on the priority of the new requests. For the simulation, all requests are accommodated at either the maximum or minimum rates, or they are denied access.

3.4 Media Access and ATM/HF Simulation

The eventual purpose of the simulation is to test all the MAC protocols, as discussed in this chapter, to determine if they will work as described. However, the first version of the ATM/HF simulation concentrates on a more fundamental question of whether the basic ATM/HF design is fundamentally sound. To determine the viability of ATM/HF the simulation concentrates on the number of blocked calls and the amount of throughput, in terms of frame slots, on a single circuit. This data will provide information on the number of nodes and users single circuits can reasonable support that in turn will allow a determination as to whether further research into ATM/HF is warranted.

The simulation does include several features discussed in this chapter. A limited version of dynamic slot assignment is used along with the ability to change the slot assignment to accommodate new connection requests. The minimum and maximum bandwidth parameters are supported, and are used to reduce the bandwidth of data traffic from the maximum to the minimum rate in order to accommodate voice traffic which has a higher priority. Explicit management messages are used to request connections and to inform the controlling node of a disconnection. The controlling node uses explicit management messages to provide slot assignment and data rate information, and to inform the other nodes of slot reassignment or rate changes.

This chapter presented the specifications for the ATM/HF MAC protocol. It discussed the different states of the network and nodes, and how each combination utilizes a different type of protocol. The three protocol specifications recommended were a carrier sensing protocol for network startup, the PRMA/DA protocol with some modifications for dynamic bandwidth reservation for accessing an active network from an idle state, and a dynamic reservation request method for adding connections to an already active node. In the next chapter, several protocols will be discussed with the view of determining how they work and if they can be used as the MAC protocol for ATM/HF.

4 Evaluation of MAC Protocols

Having presented the ATM/HF architecture and network states, and having discussed the MAC protocol specifications, it is time to examine several MAC protocols considered capable of meeting the requirements of the ATM/HF design. The reason for emphasizing MAC protocols in these next chapters is because of the vital role they play in the operation of any network. In addition, the goal of developing the simulation is to test the various protocols to determine if they will operationally work and if they will operationally meet the required ATM/HF specifications. Chapter 3 provided the specifications, and in these next two chapters several MAC protocols will be detailed and evaluated, including a description of how the protocol will work in an ATM/HF network.

This chapter focuses on both the random access protocols for the NET_IDLE State and the explicit reservation protocols for the NET_ACT State. Two random access protocols are presented with recommendations as to which would be best for AMT/HF, and what modifications, if any, would be required for their use. Several explicit reservation protocols are presented and their chief characteristics and capabilities analyzed. Recommendations are made as to which is best suitable for AMT/HF and what modifications and enhancements may require.

4.1 Basic Types of MAC Protocols

There are three basic types of MAC protocols random access, implicit reservation, and explicit reservation. ALOHA is representative of the random access protocols, Reservation ALOHA (R-ALOHA) is an example of the implicit reservation protocols, and Packet Reservation Multiple Access (PRMA) is an example of explicit reservation.

Random access is a protocol where transmission is conducted randomly with the potential for more than one broadcast to occur simultaneously. These simultaneous broadcasts cause collision, corrupting the data and making them unreceivable. The most famous of this type is Carrier Sense Multiple Access (CSMA) ALOHA. A version of this is Slotted ALOHA (S-ALOHA) which divides the media into time slots and contention is for the individual slots. Transmission occurs at the beginning of the slots only and collisions are limited to that time frame. Contention occurs in every frame for each slot.

Implicit reservation access methods build upon S-ALOHA. This type of access attempt operates in a similar fashion to the CSMA procedure of S-ALOHA; however, the transmitter gaining control of the slot maintains control of it for as long as it is transmitting, hence the term implicit reservation. Other units will not contend for that slot in succeeding frames until it is freed. The slot is considered free when no transmission is heard during that slot time, or in other words, when it is an empty slot. Reservation ALOHA (R-ALOHA) is a good example of this type of protocol.

Explicit reservation methods use the S-ALOHA procedure for and a contention period for access contention. The contention period is controlled and used strictly for access and an access message is used for an explicit request for bandwidth reservation. Once access is successful, the transmitter is assigned to a slot, or slots, which it uses until no longer required. Empty slot(s) indicates that the connection is completed and that they are free for use.

These three access methods operate in a similar fashion in that they all use a variation of the CSMA procedures. The major difference between them lies in what happens after access is successful. For the NET_IDLE State, it is necessary to use a random access type protocol since the goal is network startup and media establishment that can occur in random fashion.

For the states NET_ACT/NODE_INACT and NET_ACT/NODE_ACT, explicit reservation protocols are recommended. These protocols incorporate several flexible and dynamic capabilities, such as allowing for specified bandwidth requirements and for dynamic bandwidth assignment. In the NODE_INACT state, contention for a connection occurs using a CSMA type protocol, whereas, in a NODE_ACT state explicit management messages are used to request new connections.

4.2 Random Access Protocols

4.2.1 CSMA/CD

CSMA/CD is the IEEE 802.3 enhancement of the ALOHA MAC protocol. Stations access the network by sensing for carrier signals and then, if no carrier is present, broadcasting the data. If the data arrives at the receiving station uncorrupted, it transmits back a short acknowledgement message to notify the transmitting station that the data arrived safely. The sending station holds the message until it receives the acknowledgement and then discards it. If the message is corrupt, for whatever reason, the receiving station does not send the acknowledgement message and the transmitting station retransmits the original message after a timeout period. The reasons for corruption pertain to propagation, collision, and interference problems associated with radio transmission.

If the sending station senses a carrier signal, it refrains from transmitting and generates a random backoff period. At the end of this period, it again senses the media. It will continue this process until the media is clear and it can transmit. If two or more stations broadcast at the same time, the messages collide and become corrupt. This corruption is detected at the receiving stations that refrain from sending the acknowledgements. Since no acknowledgement was received by any of the sending stations, they all assume that their message was corrupted by a collision and generate different random backoff periods. At the end of a backoff period, one of the stations will again attempt to transmit a message. By generating random backoff times, one of the contending stations should transmit before the other, thus avoiding more collisions. Notice that the sending station does not know why the message was corrupted, but must assume a collision occurred.

The efficiency of this protocol is low because of the overhead incurred by message corruption, rebroadcasts, and access delays. It is important to remember that the rebroadcast of a message is subject to the same constraints as the original attempt.

4.2.2 CSMA/CA

Efforts to increase the efficiency of CSMA/CD have led to various recommendations including CSMA with Collision Avoidance (CSMA/CA). This protocol works by sensing the carrier for a specified period of time before transmitting. There is some overhead associated with this protocol due to the delay caused by waiting, but the number of collisions is reduced making it more efficient than CSMA/CD. CSMA/CA is an algorithm which " ... specifies that a gap of minimum specified duration exist between contiguous frames, which is called Distributed InterFrame Space (DIFS)" [19, 20]. This CSMA/CA InterFrame space (one of three InterFrame spaces used) works by having a station sense for the carrier for the DIFS duration before transmitting. If during the DIFS period the media becomes busy, the station stops the access procedure, generates a random backoff time, and then waits for a period of time equal to the backoff time. Once the backoff period ends, the station begins the access process over again. If at the end of the DIFS the media is still not busy then the station transmits its message.

The ATM/HF version of this CSMA/CA protocol uses a variation of the backoff period where a station senses the media for a randomly generated period equal to a multiple of the propagation delay of the network. Relating the sense period to the propagation time is important to avoid collisions from distant stations. If two stations start sensing within a time frame equal to the propagation delay it may be possible that the second station will start transmitting just after the first since it has not had the time to sense the first transmission because of the delay in propagation. By using a randomly sized delay period equal to a multiple of the propagation delay, a further reduction of collisions may be possible.

However, collisions can still occur and ATM/HF must overcome the inability of a broadcasting station to detect the collisions of its own signal. The IEEE 802.11 CSMA/CA protocol relies on acknowledgement messages to determine successful transmission without a collision. In similar fashion, ATM/HF relies on collision messages from other network nodes to know if a collision has occurred. This means that after a preamble, a short time period similar to the 802.11 Short InterFrame Space (SIFS) is used to allow any listening station to send a collision message if it detects a collision. If this collision message is heard, then the station attempting network startup will backoff for a random period and then retry network startup.

When the procedure initially starts, all other nodes stay in the NODE_IDLE State and wait for the network to startup. By keeping them in the NODE_IDLE State, only the stations attempting startup will continue the attempt keeping collisions to a minimum. If at the end of the backoff period the network has not been started, then all the nodes reset to NET_INACT/NODE_IDLE and the CSMA/CA process starts over again. Since the network is a geographically small area, the backoff times should be small enough not to create a long access delay time.

Another problem ATM/HF must overcome is that of hidden stations. These are stations that can not be heard by all stations within the network. This is a complicated problem and is not dealt with in this paper except to mention that the backoff algorithm recommended here is an attempt to reduce collisions from this type of problem since stations between the two startup stations will transmit the collision message. Further enhancements to the collision message can indicate the type of collision including those from hidden stations.

The version of CSMA/CA described here need not be complicated since it is only required for network startup. Once accomplished, the access method for all other stations changes. For this reason, CSMA/CD may be just as practical and efficient as CSMA/CA, and may be easier to implement. For this reason, this version of the CSMA/CA protocol needs to be tested and compared to the simpler CSMA/CD protocol. Although not the focus of the current simulation, the eventual goal is to continue to expand the simulation in order to test these protocols for network startup.

4.3 Explicit Reservation Access Protocols

Explicit packet reservation protocols, sometimes referred to as demand access protocols, are characterized by the use of control information within the initial access packet for requesting bandwidth to match the QoS requirements of the connection. These requests allow the network to allocate additional slots without additional contention and subsequent overhead. Another characteristic is the provision for dynamic bandwidth allocation by using control information either embedded into the slot headers or sent via explicit management messages. These capabilities support the use of variable bit rate applications and increase overall efficiency of the network.

Some explicit access protocols allow for S-ALOHA type contention throughout the entire frame, while others utilize an S-ALOHA contention period within the frame. Some allow contention for a full slot, while others allow contention only during a contention period for smaller control slots. For this second type of access method, smaller control packets are sent. Acknowledgment of successful access attempts are characteristic of these protocols, and is usually denoted by a slot assignment message. For ATM/HF two explicit access protocols, PRMA/DA and PODA, are examined. Before going into detail on these two protocols, however, a brief discussion on the importance of dynamic bandwidth allocation and dynamic call reservation is presented to provide a reference point on their use in the protocols.

4.3.1 Dynamic Bandwidth Allocation

Dynamic bandwidth allocation is important because it supports the new VBR class of applications. These applications vary in the amount of bandwidth they require which can cause inefficient use of bandwidth and loss of data. When less bandwidth is required, the empty slots are wasted bandwidth. These slots can be returned to the network for use by ABR and UBR applications increasing overall network efficiency. In some protocols, an empty slot indicates the end of the connection and the slot is reallocated. This causes collisions and additional contention since the VBR application will need to renegotiate the connection causing additional access delay, loss of data, and frustration on the part of the users. To avoid this, it is important not to let the buffer of the VBR application become empty. This is done by dynamically reducing the bandwidth.

Besides wasted bandwidth and loss of connection, another problem with VBR applications is when a heavy burst of data occurs causing the queue to become full. If this occurs, data will be lost degrading the connection. By allowing the application to dynamically request more bandwidth, the burst overflow can be avoided and data loss prevented. Of course, a limiting constraint is that the requested bandwidth must be available or data loss will still occur. However, the goal is to prevent as much loss as possible by allowing dynamic bandwidth adjustment.

Besides the VBR applications, another advantage of dynamic allocation is partial fulfillment of requested bandwidth in order to allow more applications to access the network. For example, a 32kbps data connection request can be approved at a 9.6kbps rate. If more bandwidth becomes available, the rate can be increased. The use of priority indicators can be used to determine resource allocation. Another way dynamic allocation helps is in reducing the rate of high bandwidth data connections to allow higher priority voice connections to gain access. Simplified versions of these two capabilities have been incorporated into the current version of the simulation.

4.3.2 Dynamic Call Reservation

Dynamic call, or slot, reservation reduces the number of stations having to contend for additional slot assignments for new calls by allowing them to dynamically request the additional connection. This capability is associated with multiplexing several connections over a link from a single source node. An example of this is a satellite ground station where multiple sources are multiplexed onto a single uplink to the satellite. When an already connected remote station requires additional bandwidth for an additional call connection, instead of using a contention method, it uses management messages to negotiate the connection with the ground station. This type of dynamic capability reduces contention and access delay, and improves throughput efficiency.

4.3.3 The Need for Developing a New MAC Protocol

At this time there are no explicit MAC protocols that provide both dynamic bandwidth and dynamic slot requests. Most new wireless MAC protocols are interested in either the random access type networks such as the IEEE 802.11 Wireless LAN, or in cellular telephone applications which consist of single mobile stations accessing a base station. The base station in a cellular system does not require a MAC protocol since it is the only broadcast source on the channel.

In ATM/HF all source nodes are multiplexing connections, attempting access from idle states, and seeking to dynamically gain additional bandwidth for additional calls once connected. Also, each type of application has a different QoS requiring the network to provide a variety of bandwidth connections. This situation calls for combining dynamic bandwidth allocation and dynamic call reservation capabilities into the ATM/HF MAC protocol. To accomplish this, two MAC protocols were chosen for incorporation into ATM/HF. The first is Priority Oriented Demand Assignment (PODA) which was developed in 1977 for satellite systems. The second is Packet Reservation Multiple Access with Dynamic Allocation (PRMA/DA) which was presented in 1996 as a means of supporting VBR traffic over cellular telephone networks. PODA has the capability of dynamic call reservation and PRMA/DA was specifically designed for ATM and has the capability of dynamic bandwidth allocation.

4.3.4 Priority Oriented Demand Assignment (PODA)

PODA was introduced around 1977 as a MAC protocol for satellite networks and is a combination of various reservation schemes. The various schemes " ... were synthesized into a reservation system called PODA (Priority Oriented Demand Assignment), which also extended the reservation concept to include integrated packet, voice, and data traffic, and priority allocation of capacity" [21].

```
|<-    Information Subframe          ->|<-  Control Subframe      ->|
+--------------------+-------------------+----------------------------+
|  Centralized       |  Distributed      |          S-ALOHA           |
|  Assignment        |  Assignment       |          Slots             |
+--------------------+-------------------+----------------------------+
```

Figure 4.1 - PODA Frame Structure

Figure 4.1 shows the structure of a PODA frame that is divided into two subframes, the Information Subframe and the Control Subframe. User data resides in the Information Subframe, and the source sites use the Control Subframe for accessing the network from an idle state. For access control, an S-ALOHA procedure is used. Using the Control Subframe limits the contention period and successful access is acknowledged by slot assignment.

```
| Guard time | Preamble | Message_1 | Message_2 | ... | Message_k |
                        /           \
                       /             \
                      /               +-----------+
                     /                             \
                    | Message Header | Message Data |
```

Figure 4.2 - PODA Burst in Information Subframe

Figure 4.2 shows the structure of a PODA slot. An important feature of PODA is keeping together all the traffic from a single source in order to minimize overhead from guard times and preambles. This feature is important to ATM/HF because of the low bandwidth. Although not specifically designed for ATM networks, with slight modification PODA could be adapted to accommodate ATM cells, and by keeping the slots of a source together it may be possible to increase the number of slots available to the network.

PODA allows dynamic call reservation requests to be made through the headers of existing traffic. This process reduces the need for contention that in turn allows the network to use a smaller control subframe or fewer slots for contention. Because additional connections can be made in this dynamic way, more stations can access the network from the idle state.

Another important feature of PODA is the capability to dynamically vary the number of slots in the Control Subframe. The number can be increased or decreased to accommodate the number of access tries. This reduces access delay if a large number of sources are attempting access and can increase the number of slots available for user data when there is little access activity taking place. Because of the limited bandwidth of ATM/HF, this type of dynamic capability will be hard to incorporate, but it is possible to use a modified version to provide extra available slots.

PODA does not allow for dynamically changing bandwidth once the connection is established. The next MAC protocol to be examined was designed to accommodate variable bandwidth connections and the new VBR traffic class. The uniqueness of combining the dynamic capabilities of the two protocols is the major contribution. Table 4.1 is a summary of PODA capabilities.

Access using S-ALOHA for control slot during control subframe period	The reservation request can be for more than one connection
The control subframe can expand to decrease the number of collisions.	Reservations can be made in the header of scheduled traffic.
Successful access is indicated by slot assignment.	

Table 4.1 - Summary of PODA Capabilities

4.3.5 Packet Reservation Multiple Access with Dynamic Allocation (PRMA/DA)

PRMA/DA was introduced in 1996 to provide a MAC protocol for VBR and CBR traffic classes using a common shared medium. It was designed for voice traffic operating over conventional cellular systems and relies upon uplink and downlink capabilities [22].

PRMA/DA was developed for use with ATM and is the first of the PRMA protocols specifically designed for a wireless ATM system. The contention procedure is S-ALOHA and uses a contention period within each frame consisting of available slots. These available slots are full slots, not smaller control slots. Figure 4.3 shows the structure of a PRMA/DA frame and the location of the Available Slots.

```
| GT | Frame k | GT | Frame k+1 | GT | Frame k+2 | GT | Frame k+n |
                   /            \
                  /              +-----------------------------------+
                 /                                                    \
                 | Available | CBR Reserved | ABR Reserved | VBR Reserved |
                 | Slots     | Slots        | Slots        | Slots        |
```

Figure 4.3 - PRMA/DA Frame Format

The number of available slots vary dynamically, increasing to meet increased demand and decreasing for reduced demand. This dynamic ability keeps collisions to a minimum by providing more slots for access as demand goes up. It also improves throughput by freeing slots for use. However, the important dynamic capability concerns the allocation of user slots. PRMA/DA dynamically adjusts the number of slots for use by the different traffic types.

Figure 4.4 shows the encapsulated transport cell used in PRMA/DA. Within the transport cell header is a field for requesting the required number of slots needed in the upcoming

```
| PRMA/DA Header | ATM Cell Header | 48 Byte Payload | PRMA/DA Trailer |
     /                \
    /                  +---------+
   /                              \
| SYNC | NS | Other Control Fields |
```

Figure 4.4 - PRMA/DA Cell Format (Frame Slot)

frames to maintain the traffic QoS. This field is used, along with the average data rate (Rm) and the peak data rate (Rp) specified in the access request, to determine the amount of slots to assign to the connection in each frame. This allows PRMA/DA to dynamically change the amount of bandwidth used for VBR traffic. Because of this, PRMA/DA provides the additional dynamic capabilities needed for ATM/HF.

PRMA/DA does not provide a capability for dynamic connections, but this capability is a function of PODA. Another problem is the number of available slots. For ATM/HF, the specification is for a 60-octet WATM packet size and 10 slots per frame, which does not leave room for many available slots. This problem will be explored further in the next chapter.

PRMA/DA MAC Protocol		
Access using S-ALOHA. Access occurs only during the Available Slots portion of the frame. Contention is for an actual slot.	During initial access, the contending station signifies the average and peak data rates and uses these along with the NS bits to indicate the number of requested slots for each frame.	Successful access is indicated by the BS broadcasting, at the end of each downlink frame the of assigned slots and their addresses to the reserving station. station.
The NS field is used to indicate the current bandwidth demand for every frame. It does not guaranty bandwidth	Slots available for access are dynamically adjusted to keep collisions to a minimum.	The connection is explicitly given up using control slots.

Table 4.2 - Summary of PRMA/DA Capabilities

4.4 Conclusion

In this chapter four MAC protocols that can be used to develop the ATM/HF MAC protocol were examined. The PODA and PRMA/DA protocols provide a good foundation upon which to build, and the important dynamic capabilities can be incorporated with minimal changes. Table 4.3 provides a side-by-side comparison of PODA and PRMA/DA. In the next chapter, the modifications needed to incorporate these two protocols will be provided. It is hoped that by modifying these protocols, a completely new protocol will not need to be developed.

Comparison of MAC Protocols	
PODA	PRMA/DA
Access using R-ALOHA Access occurs at any time during the frame. Contention is for an actual slot.	Access using S-ALOHA Access occurs only during the Available Slot portion of the frame. Contention is for an actual slot.
During initial access, Reservation Request (RR) bits are used to indicate the required data rate for the connection. The BS uses the RR to allocate frame slots.	During initial access, the contending station signifies the average and peak data rates, and uses the NS bits to indicate the number of requested slots.
Successful access is indicated by the BS in the Reservation Acknowledge (RA) bits in the header of downlink messages, and Slot Reservation (SR) bits in the downlink header indicate which slots the station is to use.	Successful access is indicated by the BS broadcasting, at the end of each downlink frame, the number of assigned slots and their address to the reserving stations
RR bits are used to request a change to the required data rate in the following frames	The NS field is used to indicate the current bandwidth demand at every frame.

Table 4.3 - Comparison of PODA and PRMA/DA MAC Protocols

Table 4.3 cont.

The number of available slots for access decreases as the available slots are used for connections	Slots available for access is dynamically adjusted to keep collisions to a minimum.
Empty slot indicates end of transmission.	End of transmission is explicit by using contol information in packet
New connections are required to use the R-ALOHA access method described above.	New requests use the S-ALOHA method described above.

5 ATM/HF MAC Protocols And Network Operation

In the previous chapter it was shown that PRMA/DA has three important capabilities. The first is dynamic bandwidth allocation where bandwidth can be dynamically varied by using meta-signaling in the headers of the WATM packets. The second capability allows specifying the amount of required bandwidth during the call setup procedure. This allows the controlling node to allocate more bandwidth to the connection than could be obtained through simple slot contention. The third capability allows the number of slots available for contention to be dynamically changed in an attempt to balance between use for data and contention. The purpose is to keep access delay for new nodes to a minimum while trying to provide additional bandwidth if the slots are not being used. The importance of the PRMA/DA design is that it was specifically designed to work in an ATM network.

Another protocol presented was the PODA MAC protocol with the capability of permitting dynamic requests for new connections. This allows active nodes to request new bandwidth for additional calls without having to go through the contention process which also helps keep the number of available slots required for contention to a minimum. The important feature of PODA is capability to multiplex the different nodes, with their multiple calls, into a single stream.

By incorporating the key design features of these two protocols, it is possible to develop a new MAC protocol for the NET_ACT state that will make ATM/HF possible. This chapter describes this MAC protocol and how it operates, and leaves the idle network MAC protocol for further study. The goal of this chapter is to provide the guidelines for simulating an active ATM/HF network in order to test and determine if ATM/HF, as described in this thesis, is viable and warrants further research. The next section of this chapter details the frame and packet structures and is followed by a section detailing the protocol. The third section discusses the protocol operations, and the last section contains some concluding remarks.

5.1 ATM/HF Frame and Packet Structure Details

This section provides the final details of the frame and packet structures used in the ATM/HF design. Guard times are added to account for propagation problems. The guard times allow frames and packets to be processed prior to the next transmission beginning. Guard times are required because of multipath transmission problems which cause signals to be delayed enroute to the receiver. The arrival of the multipath signals causes fading and degradation of the original signal because they tend to arrive out of phase.

Since ATM/HF is a new concept, the basic assumption is that guard times must fit within the specified frame time of 40 milliseconds. This keeps the frame guard times from being too long which would reduce the bandwidth available for user data. Keeping the frame, and for that matter the packet, guard time short is possible because ATM/HF is considered geographically limited in size. With this in mind, the frame structure is modified as follows:

```
|<---                        40 msec                         --->|
|<-- 40 octets -->|<---            600 octets                --->|
|-----------------|----------------------------------------------|
|  GT   |  SYNC   | SLOT_1 | SLOT_2 |  ...  |     SLOT_10        |
|-----------------|----------------------------------------------|
| 1.25  |  1.25   |                                              |
|<-   2.5 msec  ->|<---            37.5 msec                 --->|
```

Figure 5.1 - ATM/HF Frame Structure

The 1.25 millisecond guard time allows the network to extend out to approximately 200 nautical miles (a nautical mile is equivilant to 2000 yards). This is based on the propagation rate of a radio signal at 6.12 mirocseconds per nautical mile. Using this number the distance is calculated as follows: (1) 1.5 msec / 6.12 usec = 204 naut. miles

The size of this guard time allows all receivers to finish processing the previous frame before the next one begins. Even though the sync signal is used to coordinate the timing between the various nodes, propagation delay and the drifting of timing circuits cause variation in the transmission and reception of signals. The guard times allow for these timing discrepancies.

Embedded within the sync signal is a frame number used by the receivers to track the frames and slot assignments. For example, a low data rate of 4.8 kbps can be assigned to a connection by using one slot in every other frame. By tracking the frame numbers, the node can transmit in the appropriate slot. Using this type of assigment increases the number of data users that can access the network by allowing rates to drop as low as 1200 bps. The packet guard times fall under the same restriction as the frame guard times. It must fit within the specified packet size which is 3.75 msecs in length. With this restriction, the packet structure is modified to look as follows:

<--- 3.75 msec --->					
<- 6 octets ->			<-- 51 octets -->		<- 3 octets ->
GT	SYNC	SEQ	ATM HEADER	ATM CELL DATA	TRAILER
125 usec each			<--- 3.1875 usec --->		187.5 usec

Figure 5.2 - ATM/HF WATM Packet Structure

The 125 usec packet guard time allows only a network size of 20 nautical miles which is much smaller than the 80 miles recommended. The only way of increasing the guard and packet sync signal times is to increase the size of the packet, but increasing packet size reduces the number of slots per frame. For this thesis, the packet size does not affect the test and will remain at the specified size with a recommendation that further study be done to determine the optimum packet size and structure.

As with the frame guard time, the packet guard time allows enough time for all signals from the previous packet to reach all receivers before the next packet is transmitted. And allow enough time between slots to compensate for signal drift between the nodes. The packet sync signal will help keep timing within the frame, and the sequence number will be used, along with the frame number, by the WATM layer for selective re-transmission of lost packets.

The guard times discussed here need to be analyzed and tested to determine if they provide adequate protection from interference and collision. However, this is beyond the scope of this particular study which is only conerned with information on blocked calls and throughput.

5.2 ATM/HF MAC Protocol Description

The ATM/HF MAC protocol will incorporate the following four characteristics from PRMA/DA and PODA:

- DYNAMIC RESERVATION (DRES): The connection request (CONN_REQ) message, from by nodes attempting access from the NODE_IDLE state, provides the controlling node with call priority, mean and peak data rates, and the number of initial slots required for the initial connection. The controlling node uses this information to evaluate the request, along with other requests, to determine if the QoS requirements can be met, and then to either deny or approve the request for connection.

- DYNAMIC BANDWIDTH ALLOCATION (DBA): During the duration of a call, the amount of bandwidth can be varied. Using meta-signalling in the header of the WATM packets or explicit management messages, an existing connection can request an increase or decrease in the number of slots for the connection. The purpose of controlling the data rate is to control the data in the buffer. Too high a data rate will cause the queue to empty out creating breaks in the data stream, and to low rate will cause the loss of data if the buffer fills.

  Another purpose for DBA is to accommodate more users. If a data application is using a 32 kbps connection (four slots per frame), and a voice connection at 9600 bps is requested, the controlling node based upon the state of the channel and the QoS requirements of both connections, can reduce the data connection to free slots for the voice call. When the voice call finishes, the freed slots could be reassigned back to the data application.

- DYNAMIC AVAILABLE SLOTS (DAS): With DAS, the number of slots available for contention varies dynamically in order to balance between user bandwidth and node access delay. As the number of nodes attempting access from the NODE_IDLE state increases, access delay also increases. To alleviate this condition, the number of available slots must increase. Even though ATM/HF does not have enough slots for more than one slot per frame to be the contention slot, it is still a capability that can reduce access delay and provide unused slots for low data rate connections.

- DYNAMIC CONNECTION REQUESTS (DCR): With DCR, additional connection requests can be made dynamically instead of through the S-ALOHA contention process. This can be accomplished through the use of explicit management messages and/or through the use of meta-signalling in the header of existing connections. This helps to reduce the number of available slotes required for access contention.

A node desiring access to an active network from the NODE_IDLE state will contend for the available slot. The message it sends is the CONN_REQ message which contains at least the following information:

- Node ID and Equipment ID
- Type of connection (CBR, ABR, VBR)

- Priority
- Minimum and Maximum data rates
- Initial number of slots required

The controlling node assesses the requirements and the current state of the network, and assigns as many slots as possible up to the requested rate. All the slots from a single node must be grouped together for efficiency, therefore there may be the need to rearrange existing connections. This capability is partially operational in the simulation and will be discussed in more detail in the following sections.

Although for the simulation all messages and requests are explicit, the controlling node can use meta-signalling. Two places where messages can be embedded are the second octet of the packet SYNC and the last 8 octets of the frame SYNC. Successful access is acknowledged with an access message, either approving or denying access. The access message, at a minimum, will include the following information:

- Message Type - ACCESS_APPROVED or DENIED
- Requesting Node and Equipment ID
- Slot assignment
- Data Rate

If a CONN_REQ message collides, then no access message will be sent and the nodes attempting access will assume a collision occurred. They will initiate a backoff algorithm and then again attempt access.

If any slot reassignments or data rate changes are made by the controlling node, a short management message is sent to the affected node prior to sending the ACCESS_APPROVED message. These messages allow affected nodes to implement the changes before the new connection starts. Once the changes have occurred, the ACCESS_APPROVED message can be sent. This management message will be repeated for as many connections as necessary to complete reassignment. The management message contains, at a minimum, the following information:

- Message Type (slot or rate change)
- Node ID
- Equipment ID
- New frame and slot numbers assigned
- New Data Rate

Slot reassignment and a form of data rate changes have been implemented into the simulation in order to provide priority to voice connections. Simulation operations are discussed in more detail in the next chapter.

5.3 Modem Requirements

It is apparent that the heart of the network is the modem since it is here that the WATM and Physical layers meet, and many of the network management operations occur. Because of this, the modem must keep track of such information as:

- Frame numbers
- Slot numbers

- Connection ID
- ID of the Available Slot

The modem converts the cells coming from the ATM switch into WATM packets by adding the header and trailer. The modem performs the CRC calculations on the cell and adds the FEC information in the trailer. The modem then queues the packets for transmission, and tracks the sync signal and maintains timing in order to output packets into the appropriate slots. It stores copies of transmitted packets for a pre-determined time and if no request for re-transmission is heard, it discards the packet.

The modem uses the frame number and packet SEQ_NUM to request re-transmission of missing packets, performs the CRC, and corrects single bit errors. Messages with multibit errors are discarded and it is up to the application to request the information from the peer application. The modem strips the header and trailer off the WATM packets and forwards the cells to the ATM switch.

When there is no network (NET_INACT state), when a request for a connection arises, the node moves to the NET_ IDLE state and the modem initiates the network startup procedure. The modem also handles the access procedure for additional connections from the NODE_ACT state. In the following section, these procedures are discussed in more detail.

5.4 MAC Protocols Operation

The four states for a node are NET_INACT, NET_ACT/NODE_INACT, NET_ACT/NODE_IDLE, and NODE_ACT. The current version of the simulation was developed to test for call blocking and throughput with the eventual goal of testing the MAC protocols in the next version. Because of this, this thesis concentrates on an active network and forgoes any further discussion on network startup. This section describes how the MAC protocol operates in the NET_ACT/NODE_IDLE and NODE_ACT states.

5.4.1 Establishing a Connection

5.4.1.1 NET_ACT/NODE_IDLE State

In order for a network to be considered active, there must be at least one node broadcasting a sync signal and having an established connection. The node transmitting the sync signal is referred to as the controlling node. The established connection does not need to be bi-directional, but can be a simplex broadcast. In an active network (NET_ACT), all nodes not connected are in the NODE_IDLE state. They would only be in a NODE_INACT state if they are just coming into range of the network and have not yet switched to the NODE_IDLE state. It is from the NODE_IDLE state that the S-ALOHA access procedure is used. Figure 5.3 shows the basic flow of messages for a connection request from the NODE_IDLE states.

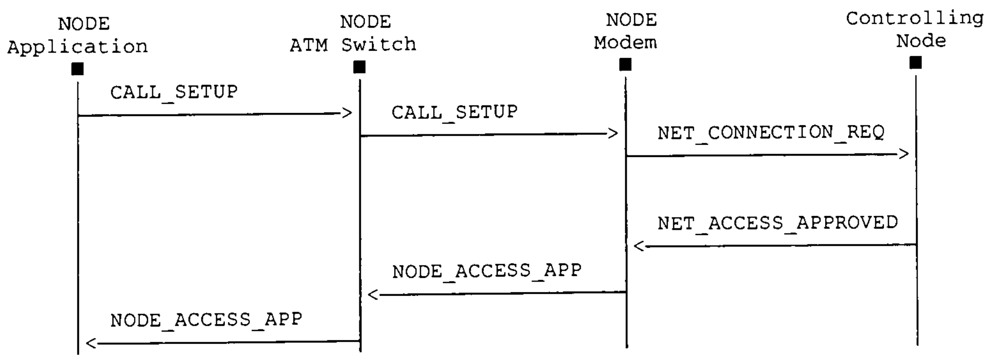


Figure 5.3 - Message Flow For A New Connection (Access Approved)

In the NODE_IDLE state, the ATM switch forwards a call setup request from a user application to the modem. The modem sends a request to connect to the network (NET_CONN_REQ) during the available slot time. The available slot is the contention slot and the procedure is similar to S-ALOHA. If access is successful, the controlling node sends an approval message (NET_ACCESS_APPROVED) containing slot assignments and other necessary information. Upon receiving approval to access the network, the modem sends an approval message (NODE_ACCESS_APP) to the ATM switch, which establishes the connection to the application and forwards the message to it. Upon receiving the NODE_ACCESS_APP message, the application begins transmission.

The call setup request from the applications provides the modem with the information it needs for making the appropriate QoS request from the network, and the node access approval message from the modem provides the application with the QoS response from the network. If no assignment can be made due to a busy network, the controlling node sends a NET_ACCESS_DENIED message and the call is blocked.

If a collision occurs, no access messages are sent by the controlling node, and without the access message the nodes assume that the attempt has failed and random back off times are generated. If collisions occur at a high rate, then the controlling node increases the number of available slots and broadcasts information on their frame numbers. This facilitates access and reducees further delay. The number of available slots can be reduces as access attempts and collisions are reduced. Other nodes that hear collisions refrain from attempting access until they hear a successful access message.

Termination of a call is explicit and follows a similar procedure as the call setup. Basically, the application sends a disconnect message to the ATM switch which notifies the modem that the call is complete. The modem sends a message to the controlling node which frees the slots for reassignment.

This MAC protocol is used by a node only to establish its first connection. Once the first call is established, subsequent requests for new connections are made using meta-signalling or explicit management messages. The next section describes this access procedure.

5.4.1.2 From the NODE_ACT State

Active nodes can request connections for additional calls by using explicit management messages and or meta-signalling. Meta-signalling uses the WATM packet header bits to send the request and provides the same information as in an initial NET_CONN_REQ message used for the S-ALOHA contention procedure. The controlling node considers all requests and makes the appropriate assignments based upon QoS, priority, and available bandwidth.

It is important to remember that more than one node can make requests at the same time using this access procedure. If a request can not be accommodated, the call is blocked, otherwise the connection procedure described in the previous section is followed. The requesting node can change the QoS on the connection a try again. Although voice applications are not prevented from attempting access, data applications can be. This keeps data applications from continuously attempting to make access causing heavy access delays. An additional capability that will help relieve congestion is that of channelization as described in chapters 1 and 2. Although channelization is not part of this simulation version, it can easily be implemented for the next.

5.5 Summary

In this chapter, basic ATM/HF MAC protocols and network operations were described in order to provide a basic understanding of how access is attained and how the network operates. The ATM/HF MAC protocols discussed are new in the sense that they combine the important features of PRMA/DA and PODA. Also presented were the frame and packet structures with guard times, a necessity for helping prevent collisions and interference due to multipath propagation delays and clock drift at the nodes. In addition, the use of meta-signalling and explicit messages for connection management and control was presented.

The description of the network in this chapter was used to establish the criteria for determining the viability of ATM/HF as a workable solution to using HF radio as a medium for ATM networking. It was used as a framework for building the network simulation which will be discussed in detail in the next chapter.

6 Simulation Purpose and Methodologies

6.1 Purpose of the Simulation

The purpose of this simulation is to determine whether ATM over HF radio is feasible and if the ATM/HF network design is a viable architecture for accomplishing this. Put another way, the questions are:

- Can HF radio support an ATM network?
- Will ATM/HF work as an ATM network?

The goal of the simulation is to gather call blocking and throughput information in order to make a preliminary determination concerning both these questions. Since HF radio is restricted to a very narrow bandwidth of 4 kHz per channel, a very basic assumption needed to be made concerning the possibility of getting more bandwidth. It is assumed that the international community would approve channels with more bandwidth if it is proven that HF could support ATM networking. Once approved then it would be a matter of developing the filtering capability of the radios to handle it. This thesis is based upon the assumption that more bandwidth would be approved and that future HF radios will be able to utilize the bandwidth.

In order to answer these fundamental questions more detailed questions need to be asked:

- Given the specified bandwidth, how many nodes can ATM/HF support on a single channel?
- Since each node supports multiple users, how many users can a single channel support?
- What percentage of calls would be blocked?
- What percent of throughput would be attained?

To answer these questions a simulation of ATM/HF was developed using descreet event simulation software. This chapter provides a basic description of the simulation design and features, how the simulation tests were conducted, and the type of data collected. In the following chapter, the analysis of the data is presented along with conclusions concerning ATM/HF, its ability to successfully support an ATM network over HF radio, and if research into its viability as a network should continue.

6.2 Simulation Design and Features

6.2.1 The Simulated Users

There are two types of users available in the simulation voice and data. The modules that simulate the voice users are referred to as constant bit rate generators (CBR_GEN), or voice generators, because they output a continuous stream of data at a fixed rate and are assigned a fixed bandwidth for the duration of the connection. The modules that simulate data users are referred to as variable bit rate generators (VBR_GEN), or data generators, and they represent applications such as email, message broadcasts, and file transfer. They output messages of different sizes and are connected for only as long as they are sending data. Data generators are assigned either 1 or 4 slots per connection representing data rates of either 9.6 kbps or 38.4 kbps respectively. Their rates can be reduced from 4 slots to 1 slot if necessary and are connected for varying amounts of time representing bursts of data.

Although a message broadcast channel exists in actual networks it was not used in this simulation. This type of channel is dedicated to broadcast message traffic and is a continuous stream of data. This type of channel remains active as long as there are messages to send and it can operate for many hours at a time. A message broadcast channel can be implemented in the next version of the simulation as a constant bit rate generator.

6.2.1.1 CBR Generator Details

The CBR_GEN, or voice generator, represents applications that have a constant bit rate output with a fixed bandwidth, and in the case of this simulation it represents voice traffic. The voice generator has a cell rate of 25 cells per second which produces 25 x 48 = 1200 bytes of data per second. For the simulation, this output includes silent periods which are times when no one is speaking and the slots are empty. This data rate is a good representation of the number of cells the AAL and ATM Switch would produce for the specified voice rate discussed in a previous chapter. To represent a full duplex connection, two frame slots are assigned to this connection and the generator ouputs 50 cells per second for a total data rate of 19.2 kbps. A CBR_GEN can have other rates assigned in order to represent other applications that utilize a constant number of slots per frame, such as a message traffic broadcast channel at 9600 bps.

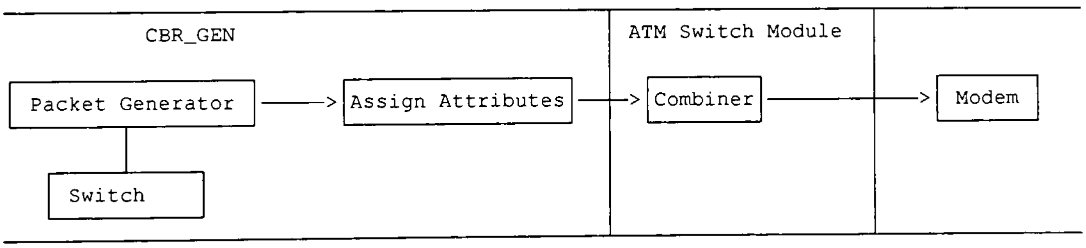


Figure 6.1 - Block Diagram of CBR Generator

Figure 6.1 is a block diagram of the CBR Generator. A TRUE/FALSE signal from the Switch turns the packet generator on and off. This TRUE/FALSE signal represents the duration of the call (CALL_DUR) which is a random value less than the specified OFF_HOOK time. The OFF_HOOK time is a Poisson distributed number that represents how often a call is made.

While the Switch signal is TRUE, the packet generator outputs packets at the specified cell rate, which for the voice generators would be 50 cells/second. The cells are forwarded to the next module where they are assigned attributes such as EQUIP_ID, CELL_ID, and MSG_ID which are used to identify the equipment and cell use. This information represents the cell header and more attributes can be assigned as needed. The cells move from the CBR_GEN to the ATM Switch Module where they are combined with data streams from other CBR generators. These are passed out of the switch to the Modem Module.

6.2.1.2 VBR Generator Details

The VBR_GEN, or data generator, represents applications such as TELNET, FTP, message traffic, http, or any other applications which requires only asynchronous connections. These type of connections last for only as long as data is being sent, and the transmission rate can be reduced or temporarily halted to make slots available for higher priority traffic. The design of the generator is similar to that of the CBR_GEN with the exception of the way the duration of the call is controlled. In the VBR_GEN, the CALL_DUR is based upon the message size.

The VBR_GEN can also represent an application that requires a minimum fixed constant data rate but also occassionally needs additional bandwidth to accomodate high bursts of data. In this situation the request for connection provides the minimum and maximum bandwidth requirements. This type of VBR_GEN represents the newer Variable Bit Rate (VBR) traffic classes such as video and multimedia. Although it is important that ATM/HF support this new class, it was not possible to test in the current simulation version. It is planned to incorporate this into future versions of the simulation.

6.2.1.3 Sync Signal Generator Details

The Sync Signal Generator is different from the other generators because it supplies a constant sync signal packet with frame number and controlling NODE_ID. Based upon the ATM/HF Frame Structure there are 25 frames per simulation second and they are numbered from 1 to 25. The sync signal is used by the nodes to control network timing and start transmission of the packets stored in the buffer. Although not implemented in the current simulation, the frame numbers will be used to control data rates of less than one slot per frame and to test the MAC protocol for access through the available slot. Using the frame number, the number of available slots for access can be varied from between 1 every 25 frames (1 per second) to 1 per frame (25 per second) making more slots available for user data. Only the controlling node supplies the sync signal and for the purposes of this simulation there will be only one controlling node.

The sync signal packet goes directly to the Modem Module which "broadcasts" it to all other nodes. The receiving nodes parse out the sync signal attribute and frame number from the incoming packets and use them to activate a pulse switch to pulse the buffer in the Data Module. The Data Module is where all packets ready for transmission are queued. The packets are transmitted out of the Modem Module at a rate of one packet per pulse where the number of pulses is based upon the total data rate of all the connections. This process is discussed in more detail in the section on the Modem Module.

The simulation frame structure is slightly different from that proposed for ATM/HF because it uses a full slot to represent the guard time and frame sync signal. This was necessary in order to maintain an accurate representation of the number of slots within a frame based upon the limitations of the software. This slight variation does not affect the test results and allows each frame to represent the specified 40 msec time period.

6.2.2 Simulation Module Details

The basic structure of the simulation is illustrated in Figure 6.2. The two basic modules from which the simulated network is built are the Media Module which represents the RF air interface, and the Node Module which represents the mobile ATM network.

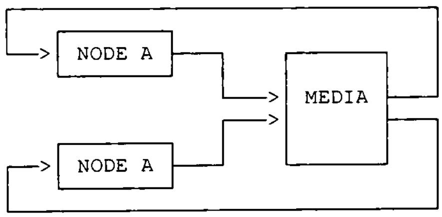


Figure 6.2 - Simulation Block Diagram

Each node connects to the Media Module where all incoming packets are combined into one stream, duplicated, and sent to all nodes at the same time. This operation simulates an RF broadcast environment. This design also allows for expansion of the network by simply adding another node and connecting it to the Media Module. The following sections detail the operations of these two modules.

6.2.2.1 Media Module Details

The Media Module reflects the broadcast media, or Air Interface, of the HF network. It accepts the transmissions from the nodes and rebroadcasts them to all the nodes in the network, including back to the transmitting node. This is an excellent representation of the how the media operates and of the practice of larger nodes monitoring their own broadcasts. The Media Module is capable of supporting as many nodes as desired, however, for the simulation only two nodes where used.

Figure 6.3 is a basic block diagram of the Media Module showing support for three nodes. The inputs and outputs are labled and dedicated to one node. A packet comes into the module from a node and is combined with other packets from other nodes into a single stream using the combiners. For this simulation it is assumed that no collisions will occur. A collision module is

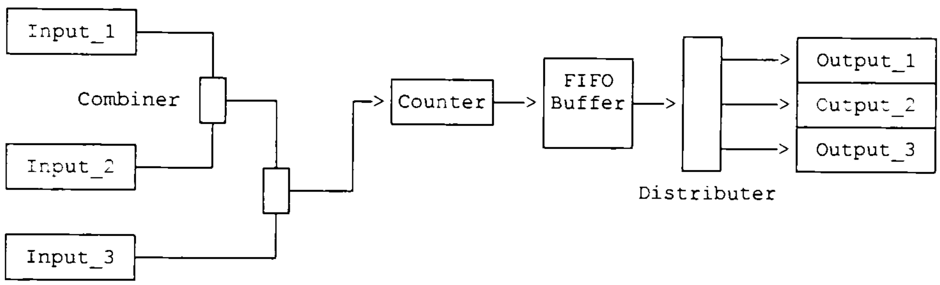


Figure 6.3 - Media Module Block Diagram

planned to simulate the collision of NET_CONN_REQ messages during the availabe slot. This will allow testing of the MAC protocol from a NODE_INACT state. From the combiners, the packet moves through various counters (only one is show here for simplicity) which keep track of the various packets that are transmitted. For example, one counter keeps track of the number of NET_CONN_REQ messages sent, and another tracks the number of requests denied. Using these two numbers, the percentage of blocked calls can be calculated.

The zero delay FIFO QUEUE is required for the simulation to work properly and does not add any delay time. The packets are routed through the various counters and then finally move to a distributer which duplicates and sends them out simultaneously to all network nodes. As the number of nodes increases, the number of combiners and distributers also increase to accommodate them. To simulate channelization, a second Media Module can be added. This would represent additional communications channels operating at different frequencies and allows a node to start or switch to another channel as the available channel becomes busy, increasing capacity and throughput. A busy channel can be represented by a level of call blocking.

6.2.2.2 Node Module Details

The Node Module represents a mobile ATM network such as a truck, ship, or plane. It contains three sub-modules; the Modem, the ATM_Switch, and the Syn_Signal_Generator. Each of these modules are described in more detail below. Figure 6.4 is the block diagram of the Node Module.

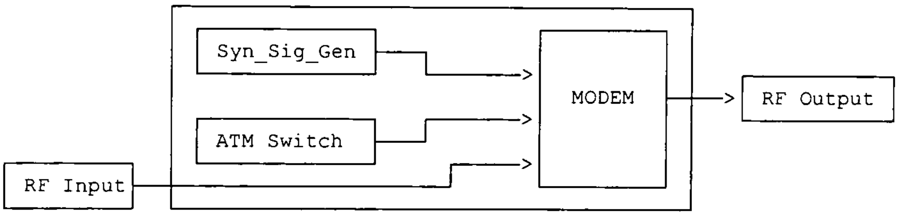


Figure 6.4 - Node Module Block Diagram

The Node Module contains the user generators which create the application cells and packets, and is responsible for combining these data streams into one serial transmission for broadcast out to the other nodes. The Controlling Node has the additional responsibilities of generating and transmitting the sync signal and performing the network management functions such as rate and slot assignments.

The Sync Signal Generator is a single purpose module responsible for generating the frame sync signal with the Controlling Node ID and sync_sig attribute. This module is only used by the controlling node and in the simulation the other nodes do not have this module. The ATM Switch Module and the Sync_Sig_Gen Module input to the Modem

Module which combines the two streams into a single output for broadcast via the Media Module. Packets from the Media Module come into Node through the RF input and are routed directly to the Modem Module.

The Modem Module is responsible for internal routing of the packets and for controlling the ATM/HF network connection. Management functions and messages, packet buffering and transmission, and sync signal timing are all controlled within the Modem. The ATM Switch contains all the users and combines their outputs into a serial stream to the Modem. The ATM Switch routes incoming managment messages to the appropriate users. Because of the design of these modules, a module representing the HF radio was not required.

6.2.2.3 The ATM Switch Module

The ATM Switch contains the modules for generating the cells from two application types, and is responsible for forwarding them to the Modem Module. In a limited way this represents the functions of both the AAL and the ATM switch with the output being a serial stream of cells. The simulation switch does not buffer cells, but this is possible in future simulations to measure their affect on the system. In the simulation, buffering is accomplished in the Modem Module and the buffers were created large enough to prevent them becoming full eliminating lost cells. Figure 6.5 illustrates the basic design of the ATM Switch.

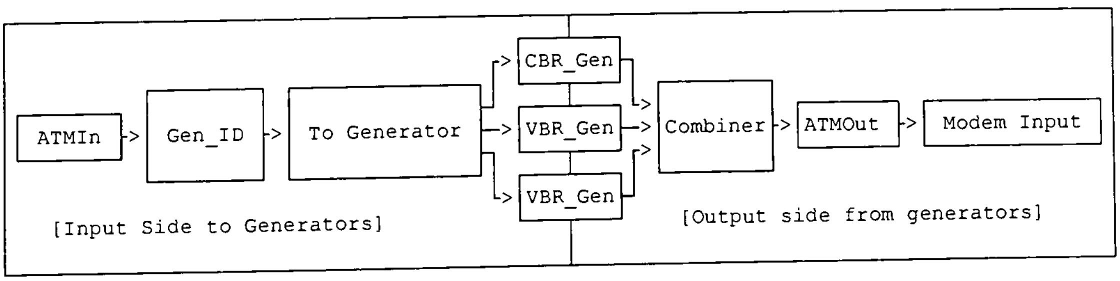


Figure 6.5 - ATM Switch Module Block Diagram

The number of generators will vary to measure the effect on call blocking. Each generator has a unique equipment identification number (EQUIP_ID) and generates it own explicit NODE_CALL_SETUP and NODE_DISCONNECT messages. If the Controlling Node approves the call, the generator starts sending packets that are combined with the outputs of other generators into a serial stream by a combiner. The ATM Switch does not perform any processing in itself and the packets are sent directly to the Modem input. Management messages coming into the ATM Switch Module (ATMIn) are routed to the appropriate generator for processing.

6.2.2.4 The Generator Switch

An important part of generator operations is simulating access attempts and call durations. To control access, a switch was developed for setting the time between attempts. Figure 6.6 is a block diagram of the basic switch design [23]. The switch uses a Random Packet Generator which outputs objects at the specified poisson distribution rate. These objects are assigned

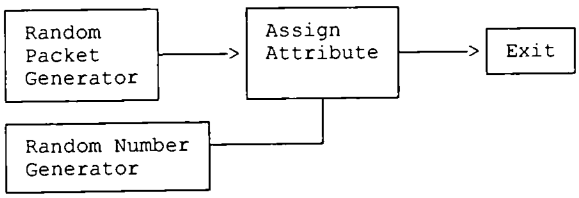


Figure 6.6 - Generator Switch

attributes used by the network to determine access approval, assign slots, and return messages to the correct Node and equipment. For example, a CBR_GEN representing a telephone application can be programmed to generate a call approximately every two minutes of simulation time. This represents the idea of a telephone going OFF_HOOK about every two minutes. The switch generates a packet which is assigned the required attributes in the next module turning it into the NODE_CALL_SETUP message. Some of the attributes assigned include EQUIP_ID, MIN_RATE, and MAX_RATE. This message is sent through the ATM Switch Module to the Modem Module where it is processed and transmitted. If access is approved by the Controlling Node then an approval message is received by the generator and packet generation, which emulates a phone conversion, begins.

Another portion of the switch pertains to call duration (CALL_DUR). This is accomplished by a delay of the approval packet for the time the telephone remains off-hook. If access is approved, then a random time is generated and assigned as an attribute to the packet which moves on to a delay module. If access is not approved, then the packet is discarded and no connection occurs. Figure 6.7 shows the block diagram of this process.

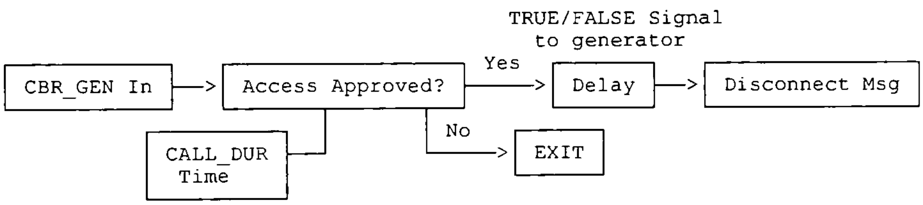


Figure 6.7 - Access Approved Flow Chart

The CALL_DUR time is used by the delay module to hold the packet for the time specified and while the packet stays in the delay module a TRUE signal is output to the call packet generator which transmits cells as long as the TRUE signal remains. When the CALL_DUR time is up, the packet moves out of the delay module and a FALSE signal is output to the call packet generator. This stops the transmission of cells representing completion of the call and the telephone going ON_HOOK. The packet from the delay module is changed into a disconnect message and sent out to the Modem. Two versions of this switch are used, one for the CBR generator and one for the VBR generator allowing for different ways of setting the call duration times.

6.2.2.5 Modem Module Details

The modem is the heart of the system because it takes all the various inputs and combines them into a synchronized stream of packets for transmission. It tracks and associates calls with buffers and buffers with specific slots. It tracks the Available Slot for network access, and it is in the modem that slot assignments occur. It receives packets designated for the node and routes them to the appropriate module for processing. Figure 6..8 shows the overall block diagram of the modem.

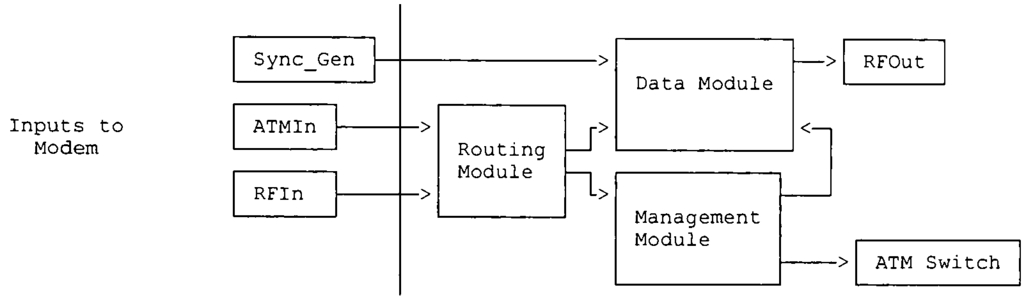


Figure 6.8 - Overall Modem Module Block Diagram

There are three inputs to the modem, the sync signal from the internal sync signal generator, the ATM switch input, and the RF input which is the incoming packets coming from the Media Module. There are two outputs from the Modem, one to the ATM Switch and one to the Media Module (RFOut). The sync signal goes directly to the Data Module for transmission. This occurs in the first slot of all frames, and as mentioned, is only done from the controlling node. The ATM cells are routed based upon whether they are data or management cells. If they are data cells then they go directly to the Data Module where they are buffered for transmission. Management cells are routed to the Management Module where they are processed.

The RF input from the Media Module goes to the Routing Module where the sync signal is parsed out and sent to the Data Module to pulse out the buffered packets. Packets are parsed first by NODE_ID and then by whether they are managment or data packets. Packets not for the Node are discarded. Management and data packets for the Node are routed to the appropriate module for further processing.

6.2.3 Automated Features

There are three automated features in the simulation dealing with slot assignments. The first makes assignment to either the MIN_RATE or the MAX_RATE, the second rearranges slot assignments to accomodate a new request, and the third deals with reducing the rate of a data connection to accomodate voice call requests which have higher priority.

The bandwidth assignment algorithm checks the incoming NET_CONN_REQ message to see if the required minimum rate can be supported. If not, it looks to see if the request is for a voice call which has a higher priority than data connections. If it is a data connection request, then a NET_ACCESS_DENIED message is sent back to the requesting Node and the call is blocked. If the request is for a voice connection, then the algorithm looks to see if there are any data connections with four slot assignments. If not, the call is denied access and blocked. If there is a data connection utilizing four slots, then the rate is reduced to one slot and the voice call is processed for access.

If the minimum rate of an incoming request can be supported, then a second check is made to determine if the maximum rate can be supported. If the maximum rate can be supported, then the assignment is for the maximum rate, if not, then the assignment is for the minimum rate. Once it is determined that the request can be supported the request packet is forwarded to the Bandwidth Support Module within the Management Module for assignment. If the call can be supported, but the slots are not contiguous, then the Bandwidth Support Module rearranges the current assignments putting all the empty slots together and then completes the assignment processed. For both rate changes and assignment changes, explicit messages are sent to notify the affected Nodes so they can take appropriate action. Finally, the ACCESS_APPROVED message is sent back to the requesting Node with slot and rate assignments.

6.3 Simulation Tests

6.3.1 How the Tests Were Conducted

The goal behind development of the ATM/HF simulation is to test the MAC protocols to determine if the system would work as described in this thesis. Of more immediate concern, however, was whether HF radio could support an ATM network and if ATM/HF was a viable way of doing this. To determine the answers to these questions, the present version of the simulation tested for call blocking and data throughput, with throughput being defined as the percentage of available bandwidth used by the generators. The tests were limited to a single channel since this is the basic building block of the network. Knowing the capacity of a single channel will facilitate planning and design.

In order to collect call blocking and throughput data for a single ATM/HF channel, a series of simulations were run. Each test run contained two nodes and with the only variable being the number of active voice and data users for each test. The simulations were run for one hour of simulation time representing the busy hour. The simulation took 91 actual hours for the full simulation time using a 166 MHz PC with an Intel Pentium processor. Six total simulations were run for the first series of tests. One simulation was run for only half a simulated hour to see if that would suffice for data collection, but it was determined that even though close to the full hour, it was better to run the full hour if possible. This first run is still used to represent one of the tests. A second simulation run failed due to a communications error between the simulation software and the spread sheet used for tracking slot assignments. The time for this test represents only 15 minutes of simulation time. This test was also be included in the results.

A second series of runs was conducted using the same two nodes. In this series, the number of voice generators was varied from 0% to 50% of the total number of users. The purpose was to determine how the voice generators affected call blocking and throughput as the network went from a strictly data network to a mixed network. The data generators were held to a constant number for all test runs.

6.3.2 Type of Data Collected

The following list shows the data that was collected for each simulation run:

- Number of nodes
- Number of user generators per node
- Total number of slots available
- Total number of packets transmitted
    - Number of sync signals
    - Number of user data packets
        - Number of CBR user data packets
        - Number of VBR user data packets
    - Number of management packets
        - Number of CBR call requests
        - Number of CBR calls approved
        - Number of CBR calls blocked
        - Number of VBR call requests

- Number of VBR calls approved
- Number of VBR calls blocked
- Number of all other management packets

Some of the data is explicit, such as the number of nodes and users per node. Most other data, such as the number of blocked calls and the number of user data packets, is collected as the simulation runs. The granularity of the test is at the packet level so that all information collected and presented will be at the packet level. When required, the data is converted to bits-per-second and bytes-per-second. Data on blocked calls are related to the reply messages coming back from the controlling node. A call can be blocked if there is not enough bandwidth available (for example, only one slot available when a CBR request comes which requires a minimum of two slots), or when there are no slots available for any type of connection, representing a busy network. The simulation does not differentiate between the two, but counts them both as simply a blocked call.

The total number of slots available for the simulation run works out to 16,500 slots per minute (11 slots * 25 frames * 60 sec). The number of slots available for user data is the total number of slots minus the number of sync signals and the number of slots reserved for media access. This leaves a total of nine slots per frame for user data, or 13,500 slots per minute. In the actual network the slot reserved for access contention is considered to be available for user data since it can be varied to provide a low bandwidth connection. However, for the simulation the first slot in the frame is always reserved as the access slot leaving only 9 slots per frame for user data.

6.3.3 Data Collection Points

The simulation software provides a built-in module for use in counting the number of objects passing through it. By placing these built-in count modules in the Media Module, the number of packets passing through can be counted based upon assigned attributes. For example, to count the number of blocked calls a decision tree was used to parse out those messages with attributes indicating that the packet was an ACCESS_DENIED message. The only packets not counted in the Media Module are the sync signals which are counted in the Sync Signal Generator Module.

6.4 Expectations

In order to determine success of the network architecture and of the ability of HF radio to support ATM operations, the simulation must show that a reasonable number of users can be supported by a single channel. It is not know at what point saturation will occur, but it is expected that a total of ten generators on each node (five voice and five data) should be able to access the network with a reasonable and acceptable call blocking level of 10%. It is also expected that the throughput, as defined in this thesis, be at least at the 80% level. If these numbers hold to these levels then ATM over HF radio can be considered possible, and ATM/HF can be considered a viable architecture. In the next chapter the results of the simulation runs are presented along with an analysis.

7 Simulation Data Analysis

This chapter presents the analysis of the data collected from the simulation test runs. This analysis is used to determine the feasibility of ATM networking over HF radio and the viability of the ATM/HF architecture. Throughout this paper it has been emphasized that the focus is on determining whether efforts should continue in developing ATM networking capabilities over HF radio. It was decided that call blocking and throughput are the two factors to be used in making this decision and that using a simulation of the ATM/HF architecture was the way to collect this data.

Analysis begins by determining the acceptable service levels for call blocking and throughput for an ATM network operating over HF radio. The collected data is then analyzed to determine if ATM/HF meets these service requirements. If it does, then it may be reasonable to conclude that further efforts to develop such this network architecture should continue. A heuristic approach is used in determining if the requirements are met.

The first section of this chapter discusses how the simulations were conducted and the type of data collected. The second section discusses call blocking and the service requirements for networking ATM over HF radio, and presents an analysis of the call blocking data collected from the simulation. The third section discusses throughput including a determination of a throughput standard and presents an analysis of the collected throughput data. The last section summarizes the findings and presents the final conclusions.

7.1 The Simulation

Simulations were conducted on Pentium PC computers with the time for each run varying from 4 to 9 days depending on the speed of the processor. Each test, except where noted, ran for one hour of simulation time (3600 seconds). Two series of tests were conducted and within each series the tests were run with only one variable. In the first series the variable was the total number generators, voice and data, per run. In the second series, the number of data generators was held constant and only the number of voice generators changed for each run. Table 7.1 shows the number of generators per node and the total number of generators for each run of the first series of tests. Only two nodes were used in both series of simulations.

Test Num	Voice Gen	Data Gen	Total Per Node	Total Per Test
1	5	5	10	20
2	5	5	10	20
3	6	6	12	24
4	8	8	16	32
5	8	8	16	32
6	3	3	6	12

Table 7.1 - Number of Generators Per Run for Series One Tests

In the first series, four tests were run for the full hour of simulation time. Test number one was run for only thirty minutes of simulation time, and test number four failed after fifteen minutes of simulation time. Both were included in the analysis. The purpose of the this series of tests was to:

- Determine the number of users a single channel can support at the stated level of service
- Collect data on call blocking and throughput
- Test the dynamic features of the ATM/HF simulation

The first series of tests used equal numbers of voice and data generators per node to determine the maximum number of users a single channel could support. The second series of tests used a constant number of data generators per run and increased the number of voice generators up to a total of five per node. This was done to determine the affect of the voice generators on call blocking and throughput and to help determine the maximum number supportable.

7.2 Call Blocking

7.2.1 ATM/HF Call Blocking Standard

The telecommunications industry strives to provide the best service possible by designing and implementing networks with a call blocking service level of only one-percent whenever possible. However, the actual level of service can range from one to ten-percent depending on such factors as cost and expansion of the user base. Once a level of service is decided upon, the next step is to determine the appropriate number of trunks required for meeting the requirement. One way of determining the number of trunks is to use the Erlang B tables.

Erlang B tables were developed for network engineers to facilitate designing networks where the number of users are considered to be infinite and where all blocked calls are lost, that is, the blocked callers do not attempt the call again during the test period. Erlang B tables were designed for voice telephone networks and do not take into consideration video and data users, and since ATM/HF is designed to consider these and newer applications, the Erlang B tables can only provide an estimate of the number of trunks required. Also considering that ATM/HF is not a system of fixed trunks but a dynamic network capable of expanding as more mobile nodes connect to the network, it is likely that the Erlang B tables will not be usable.

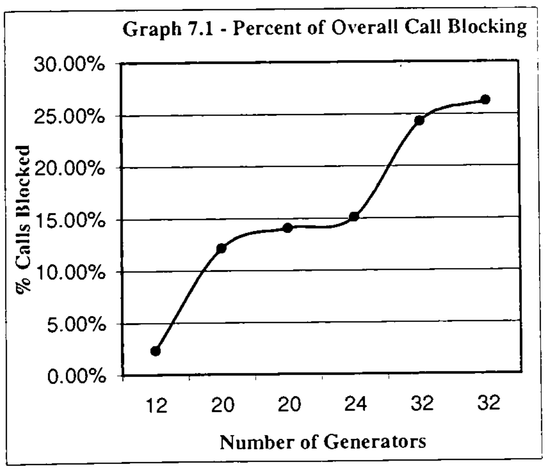
Although the Erlang B tables will not be used, they represent the service level requirements that provide will be the basis for determining the success of the ATM/HF architecture. Erlang B tables provide call blocking service levels of one, three, five, and ten-percent, so if ATM/HF is to be considered a feasible, it must attempt to meet these levels. If the simulation tests prove that a single ATM/HF channel can support a reasonable number of users within a determined service level, then it can be concluded that both ATM networking over HF radio is feasible and that the ATM/HF architecture works. The information could then be used to set the level of service for a single channel and the network can be designed to dynamically add channels as the level is exceeded. What is required to claim success of the ATM/HF design and of ATM operations over HF radio is a stated objective percentage for both call blocking and throughput.

In ATM/HF, call blocking is expressed as a percentage of the calls attempted. As discussed above, the industry standard for call blocking can vary between one and ten-percent. For these initial simulations it was decided that ATM/HF did not need to be strict in its service standards since the tests are designed to determine the maximum number of users capable of being supported by a single channel. For this reason it was decided that the service level should be ten-percent which still allows the network to meet the upper end of industry standards.

To facilitate an understanding of call blocking data collected, three graphs were generated to show Overall Call Blocking, Voice Call Blocking, and Data Call Blocking. Overall call blocking includes both voice and data generators. Voice call blocking and data call blocking are related to their own respective attempted and blocked calls.

7.2.2 Overall Call Blocking

Overall call blocking concerns the total number of user attempts, both data and voice, and the total number of these calls blocked. The number of voice and data generators per node remained equal throughout the first series of tests with both incremented equally for each run (See Table 7.1). The results of

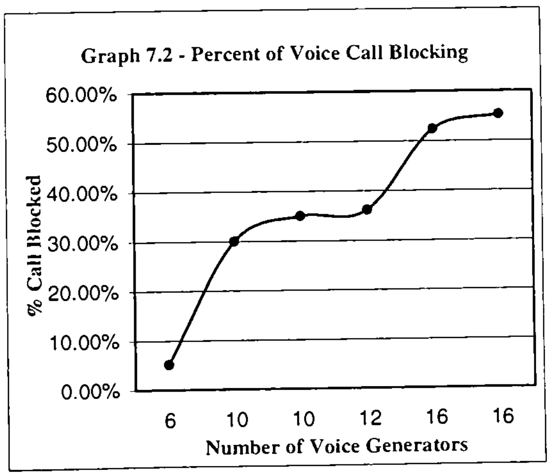


the tests were used to create Graph 7.1. By using this graph, the number of generators a single channel can support can be determined. Following the ten-percent level across the graph it can be seen that the number of generators crosses at the fifteen-user mark. This would then be the overall number of generators that can be reasonably supported on a single channel. To explain why so many user can be supported it must be remembered that the simulation is counting the total number of users that can have access to the network. Behind concept are the following ideas:

1) Not all users are active at the same time (statistical multiplexing)
2) Half the users are data generators that only access the network for short periods of time whereas voice users connect for relatively long periods
3) One slot is always free for use by data generators since the design provides an odd number of slots available for use.

So when it is said that the network can support fifteen users, it is referring to the number of users that can gain access and not the total number of users that can be connected.

7.2.3 Voice Call Blocking



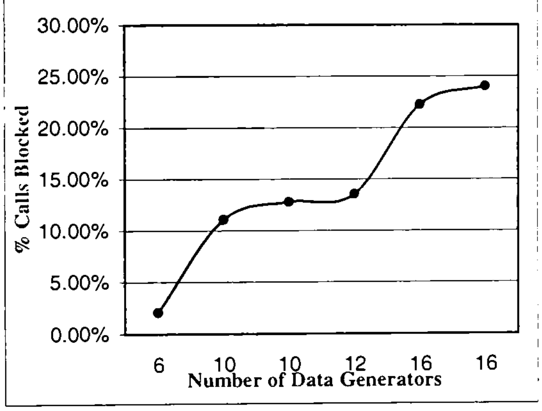
Graph 7.2 is specific to voice call blocking and shows the percentage of calls blocked to calls attempted. Notice that the number of voice users crosses the ten-percent point at the seven-user mark. This would then be the number of voice users a single channel should

support. Considering that no more than four voice users can be active on the network at any one time, this number is significant because it points to the possibility of any HF ATM network design being able to support a minimum of seven voice users for every four voice circuits provided. For ATM/HF this means that at most seven users are supportable per channel. It is possible that refinements to the ATM/HF design could increase support for voice to twice the number of voice channels available. This is considerable since the same number of channels would also continue to support a large number of data generators.

7.2.4 Data Call Blocking

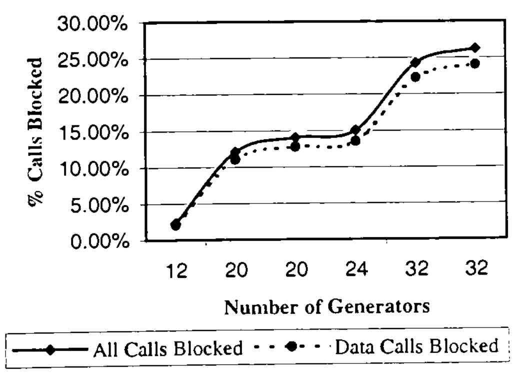
If the preceding two sections are correct, then the number of data generators a single channel should support will fall between eight and ten. Examining Graph 7.3 shows that indeed nine data generators are supportable by a single ATM/HF channel. This confirms the original finding that fifteen users are supportable. It also suggests the possibility that more that fifteen could be supported.

Graph 7.3 - Percent of Data Call Blocking

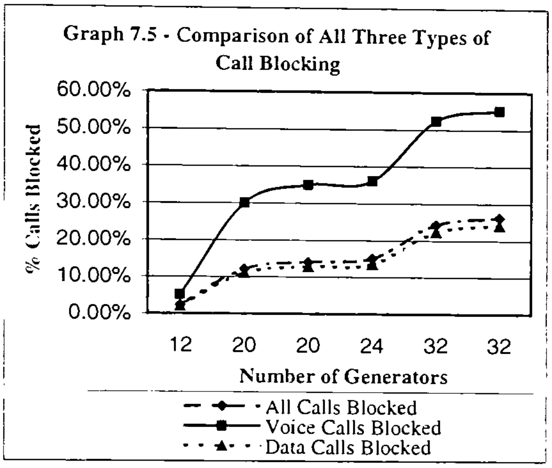


When comparing graphs 7.1 and 7.3 a correlation can be seen between overall call blocking and data call blocking. Both graphs have similar call blocking values that produce very similar curves. Graph 7.4 shows this relationship between the two, and the

Graph 7.4 - Comparison of Overall and Data Call Blocking



implication is that the data generators are not major contributors to overall call blocking. This leads to the conclusion that the voice generators are responsible for most of the congestion and blocking. If this is true, then subsequent testing should prove it.



Graph 7.5 shows that when voice call blocking reaches ten percent, the percentage of overall call blocking reaches only the five percent level. It has already been shown that at this point, the number of voice generators is seven leaving the number of data generators at this point at eight with a call blocking level of only around five-percent. To prove these numbers are correct, a second series of tests was conducted. By starting with a maximum number of ten data generators (five per node) and then incrementing the number of voice generators from zero to a maximum of ten (five per node), the overall call blocking should climb to only the five percent level. Voice call blocking should reach the ten-percent level at between six and eight generators.

Conversely, by holding the number of voice generators constant at six (three per node), and incrementally increasing the number of data generators beyond the maximum of ten (five per node), the overall call blocking should increase to the ten percent without causing voice call blocking to go over the ten-percent mark.

7.2.5 Series Two Results

The second series of tests were conducted to validate the first series. The purpose was to confirm that it is the voice generators that have the greatest effect on overall call blocking. The tests will also confirm the maximum number of voice generators supportable by a single ATM/HF channel. In this series of tests each node started with five data and no voice generators. Table 7.2 shows the number of generators per test. In each subsequent test, the number of voice generators was incremented by one per node. The only exception is test eight where the number of data generators was inadvertently lowered to four per node. This did not adversely affect the results

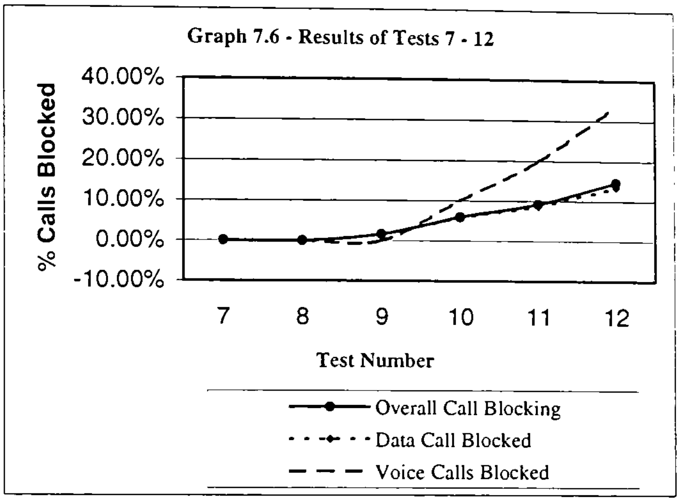
Test Num	Voice Gen	Data Gen	Total Per Node	Total Per Test
7	0	5	5	10
8	1	4	5	10
9	2	5	7	14
10	3	5	8	16
11	4	5	9	18
12	5	5	10	20

Table 7.2 - Number of Generators Per Run for Series Two Tests

If the conclusions from the previous section are correct, then the results from this series will show that overall call blocking will start at four voice generators. At six voice generators the level will be at five-percent, and at eight voice generators the overall call blocking level will reach ten-percent. Overall call blocking should go over the ten-percent limit for the last two tests since the number of users exceeds the fifteen-user limit. The tests in this series were run for only thirty minutes of simulation time in order to complete them during the allotted time.

7.2.6 Series Two Data

Graph 7.6 shows the results of the second series of tests. As predicted, call blocking was minimal through the first three tests since the number of voice generators was well below the maximum limit of seven. In test nine the percentage of overall call blocking increased to just under two-percent. All blocked calls were from the data generators.



Overall call blocking reached six percent when voice call blocking reached the ten-percent level. Data call blocking also reached the six- percent level. This validates the first series of tests proving that a single channel will support six voice generators with an overall call blocking of only six percent. This is well below the specified level of ten percent and it appears possible to substantially increase the number of data generators without forcing overall call blocking over the ten-percent service level.

It must be remembered that with data generators, messages can be stored for retransmission so that the information is not lost as with voice. This ability to store and retransmit data cells must be tested to determine how much of an affect it has on overall and voice call blocking. However, it is probable that the results would not change the conclusions that it is the voice generators that determine overall call blocking.

It must also be remembered that voice generators represent CBR applications that can also include VBR applications such as video. These applications will also have a major effect on overall call blocking because they operate with specified minimum data rates.

7.3 Throughput

This section examines the collected throughput data to determine if ATM/HF is an efficient architecture. Throughput efficiency is the measure of how well the available bandwidth is being utilized and can be defined as " ... the number of information bits correctly transferred per unit of time" [24]. For this thesis, throughput is defined as the percentage of slots available for user data utilized.

Increasing throughput is a major emphasis in telecommunications engineering and is a driving force behind much of the protocol design efforts today. ATM is one such attempt at increasing throughput efficiency and ATM/HF was designed with some of the newer proposed protocols with the intent of increasing efficiency as well as integrating newer technologies.

7.3.1 ATM/HF Throughput Standard

In ATM/HF, throughput is calculated by dividing the total number of data packets transmitted by the number of slots available for use. The number of slots available for use is based upon the total number of slots available during the simulation. The total number of available slots is determined by subtracting the sync_signal slots, the access control slots, of simulation slots. The number of access control slots equals the number of sync_signal slots. As an example, for a full one-hour of simulation time there are 990,000 available slots.

Test	AvailSlots	sync_sig	UserPkts	ManPkts	TotalPktsXmted
6	990,000	90,000	419,653	6,071	425,724

Figure 7.1 - Test Six Throughput Data

Using the data from test 6, Figure 7.1, the number of slots available for data is calculated as:

(1) 990k - (2*90k) = 810,000 slots available for user data

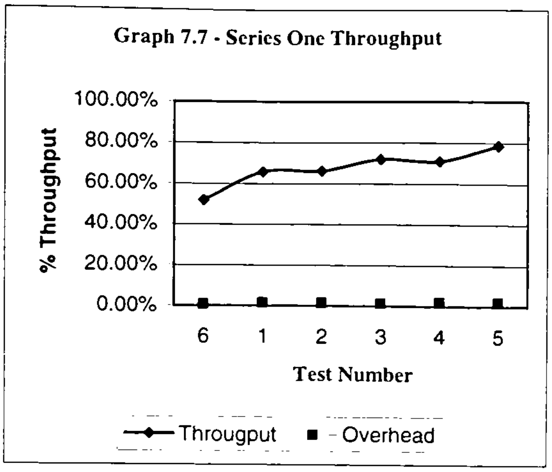
To calculate the percentage of throughput, the number of user data packets and management packets are added together and then divided by the available slots. For the example, this would be:

(2) (419653 / 810K) * 100 = 51.81%

Since the simulation is operating in the NET_ACT/NODE_ACT State all connections are established through explicit managment messages. The number of management messages was collected and is used to calculate the level of overhead. Overhead is calculated by dividing the number of management packets by the total number of available slots. Finishing the example, overhead is calculated as:

(3) 6071 / 810000 * 100 = 0.75%

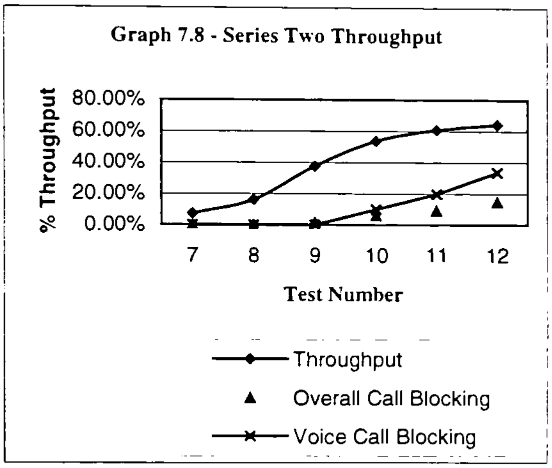
7.3.2 Analysis of Throughput Data



The results of the first series of tests are shown in Graph 7.7. It can be seen that throughput reached 78% during test five when thirty-two generators were in use. However, overall call blocking was 26.42%, which is over the specified 10% limit. Using only fifteen generators, which corresponds to test six, decided upon in the previous section on call blocking, it can be seen that throughput drops to 52%. This shows an important correlation between throughput and call blocking. In order to achieve the required levels of call blocking, it was required to sacrifice higher levels of throughput.

It is important to note, however, that this figure reflects a perfect system without any Bit Error Rate (BER) implying there are no corrupted packets with subsequent retransmission attempts. Second, the number of data generators is been limited to maintain overall call blocking at the ten-percent level. But, based upon the conclusion of the previous sections that data generators do not have much impact on overall call blocking, it may be possible to increase their number to improve throughput efficiency without affecting call blocking.

Graph 7.8 shows the throughput results for the series two tests. Test ten is the best choice to represent all the parameters required to determine throughput. In this test, voice call blocking is at 10% and overall blocking at 6%. In test ten the percentage of throughput for a single channel is 53.57%, very close to the test six results. Although it appears there is a large amount of inefficiency, there are two possible explainations for this; one is that the ATM/HF design may have a problem and, two, the specified level may not be attainable.



With the present frame structure only a maximum of four voice generators can connect to the network at one time. Based upon the discussion of voice being the controlling factor in the network, it may be possible to change the frame structure to allow more voice channels and improve throughput. This is possible by using a different compression ratio that allows voice generators to use half the current amount of bandwidth, doubling the number of users that can access the network. The problem to solve is that of the BER which would greatly affect the quality of such a highly compressed data packet.

The possibility that the 80% throughput level of service may be unrealistic and unattainable may be true, but regardless, the level achieved in the simulation is not considered detrimental in determining the feasibility of ATM over HF radio. Although this level may bring into question the viability of ATM/HF it is not enough of a problem to declare the design unusable.

7.4 Conclusion

In this chapter the results from the simulation test runs have been examined for both call blocking and throughput. The results show that the maximum number of users a single channel can support is at most six voice and at least ten data with an overall throughput efficiency of 53%. It has been shown that overall call blocking is a function of the number of voice generators with data generators having minimal effect.

The results prove that ATM over HF radio is feasible based upon the conclusion that the number of voice generators supported by a single channel is a reasonable number. The results also show that the ATM/HF architecture operates reasonably well and can be considered a viable means of implementing an ATM network over HF radio.

The simulation leaves room for improvement. For example, there are many seconds of simulation time when no data is transmitted even though the backbone remains in service. The fact that the media remains operational is not the problem, but the fact that it is not being taken advantage of is a problem. Building in the capabilities of a general message broadcast channel as well as generators that would simulate email could resolve this problem. Programming blocked messages to be stored and retransmitted, as slots become available, is another way of improving the simulation. Another consideration is developing the MAC protocols so that the controlling node can terminate the connection if no activity is occurring will also improve throughput.

Another important consideration is the tradeoff between call blocking and throughput. In order to obtain the specified levels of call blocking some sacrifice of throughput was required. This situation allows for comparing ATM/HF with other types of network designs. For example, when comparing ATM/HF with Ethernet, there is a great improvement in throughput (18% for Ethernet and 53% for ATM/HF). It can be stated, then, that ATM/HF provides a greater throughput than Ethernet at a voice call blocking level of only 10%. It similar ways, ATM/HF can be compared to other network designs and architectures.

Although there is room for improvement, the simulation has provided a specified starting point for understanding the capacity of a single channel and can be considered successful. It has shown that ATM over HF radio is feasible and worth continued study, and that ATM/HF is a viable architecture. In the next chapter the conclusions of this chapter will be discussed along with recommendations on further developing the simulation for testing at the next level, including testing the MAC protocols.

8 Conclusions and Recommendations

8.1 Review of Study

The focus throughout this research has been on determining the feasibility of ATM networking over HF radio and the viability of the proposed ATM/HF architecture. In support of this, the first chapter presented an argument for changing the current bandwidthn of HF radio from 3 kHz to 64 kHz in support of a 128 kbps data rate. Without broader bandwidth, ATM operations over HF radio become a matter of simply connecting the ATM network to an HF radio and using the 3kHz band as a low data rate, one-way channel. This requires the use of two frequencies to support full duplex operations and defeats the purpose of using ATM. This type of operations can not truly be considered ATM operations over HF radio. It is important that the additional bandwidth be allocated in order to take full advantage of the way ATM works. Without this change, developing advanced high-speed HF communications networks will be nearly impossible.

In chapters two through five the ATM/HF architecture was proposed along with brief descriptions of network operations and the MAC protocols for each of the network states. Although the design is in the preliminary phases of development and require more detailed and rigorous descriptions, they provide enough to guide the development of network simulations and for creating operational test suites. The original intent of these first simulation tests was to study the recommended MAC protocols. However, as the study progressed, it became apparent that more fundamental questions needed to be answered. That question concerned the feasibility of ATM operations over HF radio. Specifically, would this type of network work well enough to warrant further investigation? A second question emerged pertaining to the viability of the ATM/HF architecture. Would this architecture work well enough to support ATM operations? With these changes in emphasis, it was decided to proceed along the lines of developing the simulation to test call blocking and throughput as a way of measuring both feasibility and viability. This also provided an opportunity to see how well the simulation would work and what improvements would be needed to test the MAC protocols.

Chapter six provided an overview of the simulation and its operations and chapter seven provided the analysis of the collected data on call blocking and throughput. The discrete event software allowed the development of modules that simulate actual voice and data generators. Logic trees were used to process messages from the generators and to collect data on call blocking and throughput. The result of the analysis provides the baseline information required for further study.

The first two sections in this chapter provide final concluding remarks on the feasibility of ATM operations over HF radio and the viability of the ATM/HF architecture. These sections are followed by a discussion on the simulation, which is the core of this research, including shortcomings and recommendations for improvement. The next two sections discuss recommendations for further study and suggested applications for ATM/HF. Last are some final concluding comments.

8.2 Feasibility of ATM over HF Radio

The possibility of ATM operations over HF radio is not the question of this study because ATM is very flexible and can operate over any type of media. What is in question is the feasibility and cost effectiveness of developing such a network. Is it worth the time and expense of research in this area? It is important to remember that ATM was not designed simply for voice networks. The flexibility that ATM provides for Variable Bit Rate (VBR) applications such as video, and for Available Bit Rate (ABR) applications such as email, is what makes it so attractive. Also, if HF radio is to remain useful as a primary or backup communications system it must have the ability of connecting to the newer ATM communications suites under consideration.

This study proves that ATM operations over HF radio is feasible by showing that a single 128kbps channel can support up to six voice and at least ten data generators with a reasonable level of call blocking of ten-percent for voice and six-percent for overall call blocking. By tracking access attempts and the level of call blocking, the network can dynamically establish additional channels to accommodate increasing numbers of users. Based upon the number of users suggested, a simple three-channel network could support eighteen voice and thirty data generators. Channelization is a concept related to both network capacity and network extension. The ability to extend the network, with the possibility of spanning very long distances, comes from the switching ability of the ATM nodes. This type of networking allows for more sophisticated communications such as email, Intranet, file transfer (ftp), remote telnet, and multimedia. All of which can be secured with the addition of cryptographic equipment at the WATM layer.

It is envisioned that users would simply place calls or request data connections in the same manner as when using the PSTN or local network. For example, a user in a ship at sea would use the telephone as if in port by simply dialing the area code and number of the other party.

Whether this party was on another ship, plane, field unit such as a truck, or in an office, the network would route the call using the same sophisticated routing algorithms used in the current cellular systems and the mobility would be transparent to the system. HF radio fits into this network as either the primary means of communications, or by providing overflow connections when the primary communications network becomes busy, or as a backup system if the existing systems if they fail.

A basic premise to the study is that if a reasonable number of users could access and use the network then ATM over HF radio could be considered feasible. This study proved that the voice generators control call blocking and that up to six voice generators can be accommodated with an overall call blocking level of six-percent. This is well within the specified ten-percent level of service. This level of service and number of users can be considered reasonable and therefore proves that ATM operations over HF radio are feasible and worth pursuing in further study. Considering that not all of the dynamic features recommended have been incorporated into the current simulation, it is possible that even more users could be supported on a single channel than recommended here. Also, the recommended 9.6kbps voice channel could be reduced to 4.8kbps with improvements in the equipment that would reduce the BER. This would allow for doubling the number of voice circuits per frame to eight. All this makes ATM over HF radio worth further research.

8.3 Viability of the ATM/HF Architecture

The second question considered in this study concerns the viability of the recommended ATM/HF architecture. Although in preliminary form, this architecture provides enough information for the development a simulation of the network and for creating an operational test suite including hardware and software. Included in the proposal are MAC protocols required for the network to operate from any of the previously discussed network and node states. These protocols include several dynamic capabilities including an adaptation of the IEEE 802.11 CSMA/CD MAC protocol for network startup from the NET_IDLE State. They also include the PRMA/DA available slot, the slot reservation scheme for access from the NET_ACTIVE State, and an adaptation of the PODA design using explicit and embedded messages for access from the NODE_ACTIVE State.

The proposal includes frame and WATM packet structures, controlling node operations and responsibilities, and a layered reference model for the flow of data. Several newer dynamic capabilities are included such as dynamic reservation of slot assignments, priority assignment of voice over data, and dynamic reassignment of slots and data rates. ATM/HF also incorporates the ability to dynamically add additional channels based upon access attempts and call blocking.

Although not all these features were included in the current simulation, the results supply enough data to prove that ATM/HF provides access for an adequate and reasonable number of users to be considered viable. This decision is based upon the level of call blocking attained. At this level, ATM/HF was able to support up to six voice and a minimum of ten data generators on a single channel. The ability of a single channel to support this number of users proves that the proposed architecture will work and is worth the effort of further research and refinement. Although the call blocking data supports ATM/HF, the level of throughput did not reach the specified level of 80%. This can be for two reasons; 1) Either ATM/HF is inadequate and wastes bandwidth, or 2) The specification is not realistic and impossible to attain. Using the current simulation, it may be possible to determine which it is by manipulating the number of data users in an attempt to increase throughput without pushing overall call blocking beyond the specified level of ten-percent.

Throughput is related to management overhead and the BER. The BER effects the loss of cells (CLR) and the retransmission of packets. Real time voice and video packets are simply lost, but data packets are stored and can be retransmitted. The retransmission of packets has a major affect on throughput. Future simulations need to emulate and measure the effects of message retransmission on throughput, especially as the complexity of the network increases. Management traffic, consisting of the explicit management messages, was measured and found to have little influence on throughput.

In ATM/HF, throughput is measured in terms of the number of slots used compared to the number available for use. It was found that the throughput for ATM/HF is at the 53% level. Although not reaching the specified service level, this does not preclude the adoption and use of ATM/HF. This low level leaves room for increasing the number of data users and for adding and testing ABR and Unspecified Bit Rate (UBR) generators.

8.4 Simulation Comments

The simulation was designed to test call blocking and throughput, and it worked well for this study. The various modules worked as designed including generation of calls, access denial and access approval with slot assignment, rate assignment, and reassignment of both rate and slot assignment to accommodate voice priority. The ability to assign attributes to messages and packets allowed for keeping track of them as they moved through the simulation. This helped in trouble shooting the simulation and in collecting the data used to analyze the ATM/HF design.

8.4.1 Simulation Shortcomings

Although adequate for the current study, the simulation has several shortcomings that should be addressed. The first is that media quality is not considered. It is a stated assumption that there would be no errors in transmission and that all nodes would receive all packets without problem. This is not a true representation of real world operations and should be corrected by redesigning the Media Module to simulate the effects of both fading and the BER.

The second shortcoming is that the simulation is not an accurate representation of the proposed ATM/HF architecture. Many of the recommended features of the architecture were not incorporated into the simulation limiting the value of the data collected. These features need to be added and the number of users a single channel can support needs to be verified. In addition, VBR, ABR, and UBR users need to be added in order to fully test the frame structure and the MAC protocols. VBR is of particular interest since its slot assignments would vary from minimum to maximum.

A third shortcoming is that the simulation does not test the MAC protocols. One difficulty would be in incorporating the collision algorithm that would simulate collisions in the network startup CSMA/CD MAC protocol and in the NET_ACT State S-ALOHA MAC protocol. This could be accomplished by using a time stamp attribute in the call request packets that could be used to compare incoming requests in the Media Module. If the time stamps are within a specified period, a collision can be considered to have occurred and the packets discarded and the backoff algorithms would take affect.

A fourth shortcoming of the current simulation is that only explicit management messages were used to request call connections. One of the recommended MAC protocols includes using implicit messages embedded in the headers of the packets of current active connections. In addition, packets are not lost when an explicit management message is sent. This is not a true representation of how the network would work since explicit messages take the place of user data packets in voice and VBR connections causing those packets to be lost. The effect is not as apparent in the data packets of data generators since those packets are stored and can wait for transmittal.

8.4.2 Suggested Improvements

In addition to the shortcomings mentioned in the previous section, there are several improvements that should be made to the simulation. The first is the need to add a way of tracking CLR. This is an important statistic in ATM networks and is used to measure the quality of the connection. With this statistic a node can determine if the channel is degrading and take steps to improve its quality. Two possible procedures for improving channel quality would be to request either an increase in power output or to request a change of frequency. Output power can be controlled in increments from one to ten watts. Periodic SIGNAL_STATE messages would allow the nodes to request feedback from other nodes pertaining to the quality of their signals. They could then adjust power up or down as required. Frequency changes are hard to accomplish since they produce an interruption to the transmission of the signals. Data generators would not lose data because they can retransmit the messages, but there would be a disruption to voice and VBR generators. However, with the electronic tuning capabilities frequencies can be stored so that the amount of time required to change frequencies can be reduced to less than one frame (40 msec). Another way would be to use a second transmitter/receiver pair to tune to the new frequency using Automatic Link Establishment (ALE) procedures. Once the channel is established, the ATM switch could begin routing packets through that channel and once all circuits have moved the original channel can be shut down. The key would be implementing both of these procedures in the simulation to overcome BER and fading problems to determine the affect on throughput and call blocking.

Another area requiring improvement concerns the buffers in the Data Module. In the current simulation only one buffer is used with subsequent unrealistic delay in the transmission of all packets. A more realistic approach would be to have several buffers available to handle the different traffic types and generators so that the packets are transmitted more realistically and effectively. In the current simulation, the emphasis is in the number of packets transmitted per frame and the total number transmitted over the test without regard to the slot assignment. All the data rates are added and that number of packets are transmitted during each frame. The improvement would force the buffers to transmit only during their assigned slot times. This would require an improvement in the assignment algorithm so that all slots from a node are transmitted in sequence.

The access rates and call duration both need to be reworked to more accurately reflect real world conditions. The current simulation uses Poisson Distribution for the access rates and Exponential Distribution for the Call Duration times. The times were picked to simulate a busy network where calls are generated every 1, 2, 3, and 5 minutes for voice and every 5, 10, and 15 seconds for data generators. Maximum network load consisted of using two of each generator. By studying an actual network, the operations of the simulation can be improved to more accurately reflect real world operations.

Other improvements required is building a call retry process into the current generators to simulate call holding and retransmission of blocked data messages and implementing the sync signal handoff process. This would more accurately reflect how network would operate. The effect of call holding on call blocking could be measured and compared to the current level of call blocking. The purpose of adding the sync signal handoff algorithm would be to see how well the concept would work. It is, however, a complicated process since it would mean adding all the controlling node functions to all the nodes.

One final recommendation is to build the simulation with a more rigorous software package such. This would make it easier to build-in complicated algorithms as math functions instead of a series of modules. Another possibility would be to build the simulation as a series of objects using the OOP method of software development. This would allow the nodes and other modules to be built to operate exactly as they would in the field.

8.5 Recommendations for Further Study

Because of the focus of this research, there are several pertinent areas of network operations that were either briefly touched upon or not mentioned at all. One is the approach used to design the network since this will affect the architectural design and the approach to software development. Another is the network management system and the compatibility to other networks since it is a requirement that ATM/HF be able to connect to other networks. Finally, the operational possibilities provided by the development of http and the Internet need to be explored. This section discusses these areas as important to the overall design effort of ATM/HF.

8.5.1 CORBA and Objects Oriented Telecommunications Designs

One approach to software and network design is the use of the object oriented design paradigm. One of the newer recommendations for telecommunications is the Common Object Request Broker Architecture (CORBA). CORBA is a design specification that utilizes Object Request Brokers (ORB) for the handling the exchange of information between different types of systems. Object Oriented Programming (OOP) is a software design principle for writing programs that utilize software objects containing the functions and attributes of the objects that they represent. This style of programming is different from the more common procedural programming techniques. Procedural programming consists of a series of procedures whereas object oriented programming consist of interacting objects [25].

It is important these design principles be explored before further development of the ATM/HF architecture is accomplished because the approach of each particular style is completely different. Either style will work with the current state of the telecommunications industry, but since current trends in communications software are moving toward OOP it is important to consider what future systems will look like and how they will operate. The advantages of OOP are flexibility of the design and reusability of the code. These advantages can be seen in the way the CORBA architecture allows networks with different software to connect. Using CORBA specifications in the initial phase of the ATM/HF software design can make it easier to add future hardware and software upgrades.

Once a design process begins it is very difficult to go back and change. Procedural programming has been around a long time and is easy to understand. Because of this it is also easy to implement. OOP, on the other hand, is relatively new and not very well understood making it sometimes difficult to implement. However, the advantages should be considered and future research into ATM networking should look into how OOP and CORBA can be used and if they will improve the design.

8.5.2 Management Systems

The ATM/HF management specification needs to be more fully developed. The basic architectural proposal contained in this paper provides an outline of how network management should work and how it should respond to several situations, but there still many details that need to be specified and tested. An example would be the collection of QoS data and then using it to make decisions about improving network service. Research needs to be done on developing the details of the various layers of the ATM/HF Reference Model.

Another area requiring further study is the network signaling protocols. This is in reference to messages such as CALL_REQUEST and CALL_ACCESS_APPROVED used for controlling access and other management functions. It is important the signaling protocol be based upon the Q.2931 standard defined by the International Telecommunications Union (ITU). This makes ATM/HF compatible with other ATM networks, which is a fundamental requirement.
Two management designs worth mentioning are the Common Management Information Service Element (CMISE) and the Telecommunications Information Networking Architecture (TINA). CMISE " ... is a service element in the Open Systems Interconnection (OSI) model and is used to exchange information and commands for the purpose of network management" [26]. TINA " ... defines a framework for development of service and network management applications which relies on the use of a distributed processing platform such as CORBA" [27].

TINA is synchronous and is a replacement for the older Telecommunication Management Network (TMN) applications whereas CMISE is asynchronous and is designed to work with

TMN. Both are object oriented with TINA being closely related to CORBA [28] while several proposals have been made to use OOP design in developing CMISE applications [29]. The obvious connections to OOP should not be missed. It is important to recognize the significance OOP is having on telecommunication, especially in the area of network management and services. For this reason alone it would pay to conduct research into these management designs.

8.6 Applications

8.6.1 High Frequency Intranet (HiFIN)

One application for ATM/HF is the High Frequency Intranet (HiFIN). HiFIN is a proposed secure intranet which integrates video, voice, imaging, and data (ViVID). HiFIN has the look and feel of the typical enterprise wide Intranet and is accessible through any of the mobile nodes in the network. Users could access any server on the network, both mobile or fixed, and would be capable of browsing through the web sites. Other capabilities include accessing database information, using email, and file transfer (ftp) and remote access (telnet) applications. ATM/HF also allows the use of newer multimedia applications such as interactive workgroups and teleconferencing.

HiFIN can be utilized for scientific, military, emergency response, and law enforcement operations where mobility and remoteness precludes the use of wired networks and where physical terrain or equipment cost precludes LOS or sattelite links. ATM/HF channelization provides access for multiple users making command post operations possible with reduced equipement costs. In addition, the flexibility of the HiFIN design allows units not capable of ATM/HF operations to still have access to the network via conventional HF operations.

8.6.2 Emergency Communications

Where loss of installed telecommunications systems occurs due to diaster, an ATM/HF network can be established to provide emergency telecommunications services with long distance capabilities. This is especially important where microwave towers and Central Offices are destroyed. Using mobile ATM units, long distance emergency voice and data services can be established. Mobile units would be in contact with other mobile ATM units established at distant, operational Telephone Company Centeral Offices (TCCO) providing connection to the national telecommunications infrastructure. Figure 8.1 illustrates this concept.

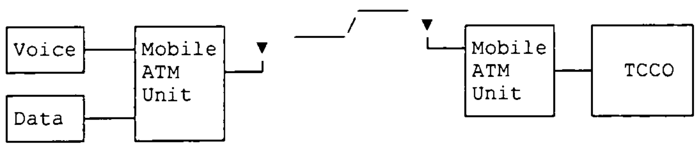


Figure 8.1 - Mobile Connection to Telephone CO

With the advantage of ATM/HF channalization, the ability exists to route calls between mobile units as well as with the TCCO. ATM/HF also has the advantage over networks such as RDRN [25] in that there is no requirement for LOS allowing the network to operate around physical obstacles such as rubble, buildings, and mountains.

8.6.3 Remote Site Communications

Scientific research, military, and law enforcement operations in remote areas can take advantage of the long distance capabilities of ATM/HF. Base camps can be established using an ATM node with one transmitter-receiver pair and provide communications for mulitple voice and data users. Handheld HF ATM units can connect through the node to local or other remote users. An example would be that of a user in a remote area using a hand held unit connecting through the base camp ATM node to a ship off shore. This ship would then switch the call through a SATCOM link to any world wide destination. Meanwhile, a user at the base camp would be using HiFIN to enter collected data into a remote database located in the home office. At the same time another user could be sending and receiving email. All this can be accomplished with only one HF radio transmitter-receiver pair. Inexpensive monitoring of multiple remote sensors is also possible.

8.6.4 Long Distance Business Use

Businesses can take advantage of ATM/HF by using HF radio in place of long distant telephone or Internet services. The effect would be to save costs on equipment and services since the initial equipment costs would be recovered within several years and the system would last for many years beyond that. Connections could be established and maintained for long periods of time without incurring long distance charges. Simultaneous voice, video, and data channels can be established between points providing true multimedia interactivity at minimal cost. Inexpensive point-to-multipoint broadcasts can be established and maintained for long periods of time. Other uses include large file transfers, and multimedia workgroups with mobile capability. As Table 8.1 shows, all applications are available to all services in ATM/HF.

Sevices / *Applications*	Intranet	Voice	ftp	telnet	video	multiple users	interactive workgroups	multimedia
Law Enf.	√	√	√	√	√	√	√	√
Remote	√	√	√	√	√	√	√	√
Business	√	√	√	√	√	√	√	√
Disaster	√	√	√	√	√	√	√	√
Fleet Ops	√	√	√	√	√	√	√	√

Table 8.1 - Applications Available to the Various Services

8.7 Conclusion

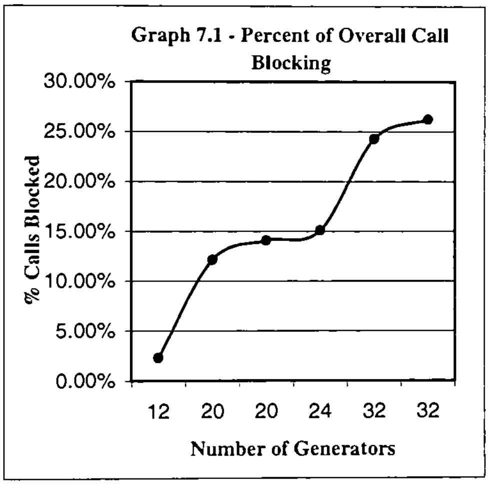
Both ATM and HF radio are important to future mobile communications. This research has shown that ATM operations over HF radio and the ATM/HF architecture will work and that research should continue. The versatility, long distance capability, and cost effectiveness of HF radio should not be lost. The international community should seriously consider allowing an increase to the bandwidth to at least the 64kHz specified for ATM/HF, and possibly up to 1MHz. Building a more sophisticated simulation can quickly and more cost effectively test the recommended MAC protocols and network operations. The immediate goals are to develop such a simulation and to build a prototype network for gathering real time data. Accomplishment of these goals allows for a final determination to be made on the effectiveness of the recommendations and architecture proposed in this study.

Appendices

Appendix A: Simulation Data

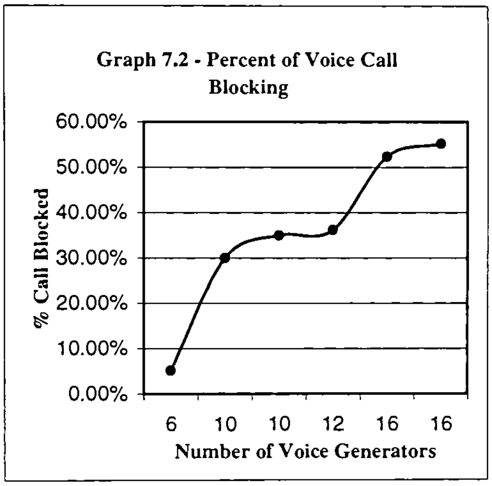
Overall Call Blocking

Test	NumGen	NumReq	NumBlock	%CallsBlocked
6	12	2023	47	2.32%
1	20	2117	257	12.14%
2	20	4194	590	14.07%
3	24	3879	585	15.08%
4	32	1603	389	24.27%
5	32	5991	1572	26.24%



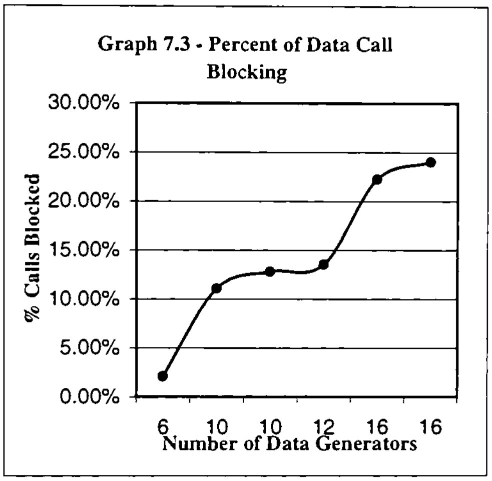
Voice Call Blocking

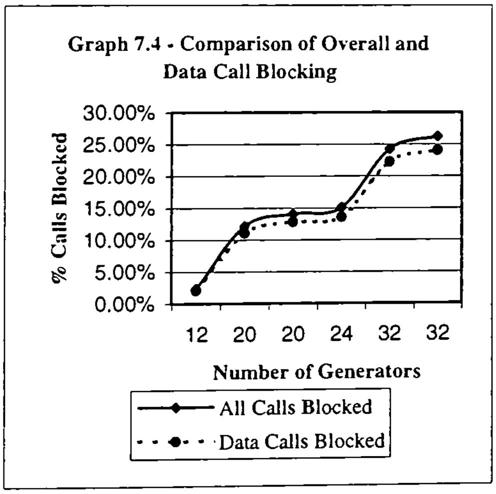
Test	NumGen	NumReq	NumBlock	%VoiceBlocked
6	6	156	8	5.13%
1	10	120	36	30.00%
2	10	240	84	35.00%
3	12	263	95	36.12%
4	16	107	56	52.34%
5	16	422	233	55.21%

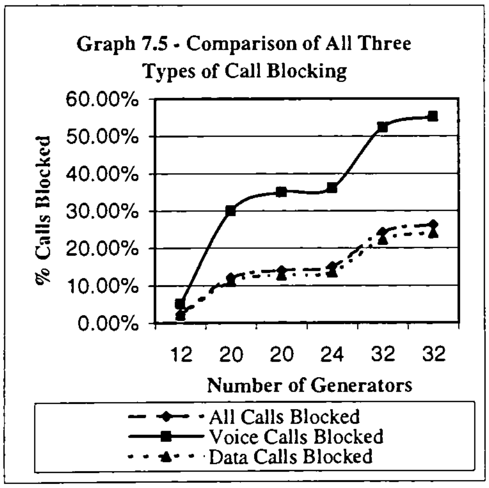


Data Call Blocking

Test	NumGen	NumReq	NumBlock	%DataBlocked
6	6	1867	39	2.09%
1	10	1997	221	11.07%
2	10	3954	506	12.80%
3	12	3616	490	13.55%
4	16	1496	333	22.26%
5	16	5569	1339	24.04%







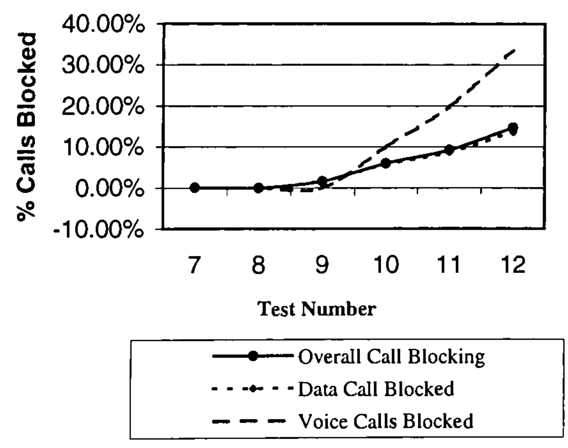
Series Two Call Blocking Results

Test	Number of Generator*	Number of OverallRe	NumOvera Blocked	%Overall Blocked	Number of VoiceReq	NumVoice Blocked	%Voice Blocked
7	10/0	2464	1	0.04%	0	0	0.00%
8	8/2	1883	0	0.00%	32	0	0.00%
9	10/4	2232	36	1.61%	65	0	0.00%
10	10/6	2180	130	5.96%	100	10	10.00%
11	10/8	2202	204	9.26%	116	23	19.83%
12	10/10	2244	329	14.66%	131	44	33.59%

*Data/Voice

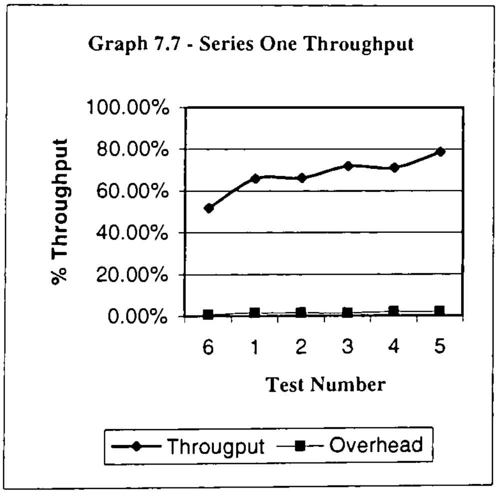
Number of DataReq	NumData Blocked	%Data Blocked
2464	1	0.04%
1851	0	0.00%
2167	36	1.66%
2080	120	5.77%
2086	181	8.68%
2113	285	13.49%

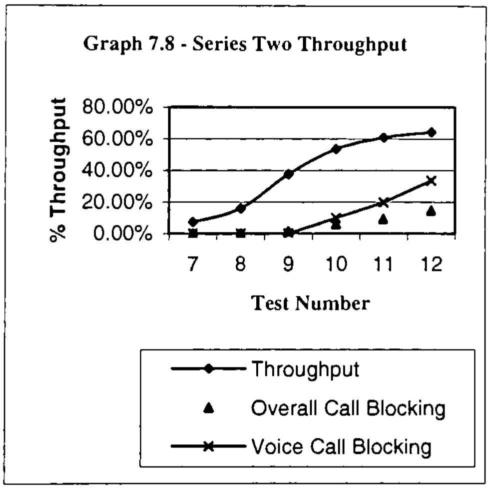
Graph 7.6 - Results of Tests 7 - 12



Throughput

Test	NumGen	AvailSlots	UserPkts	ManPkts	TotalPkts	Throughpu	Overhead
6	12	810000	419653	6071	425724	51.81%	0.75%
1	20	404460	266030	6176	272206	65.77%	1.53%
2	20	810000	536253	12110	548363	66.20%	1.50%
3	24	810000	582399	11160	593559	71.90%	1.38%
4	32	214356	152267	4462	156729	71.03%	2.08%
5	32	810000	637300	16535	653835	78.68%	2.04%
7	0/10	523329	38186	7391	45577	7.30%	1.41%
8	2/10	424645	68091	5649	73740	16.03%	1.33%
9	4/10	438876	164755	6719	171474	37.54%	1.53%
10	6/10	443950	237833	6476	244309	53.57%	1.46%
11	8/10	435149	264603	6485	271088	60.81%	1.49%
12	10/10	443130	284837	6450	291287	64.28%	1.46%





ATM/HF Data Collection Sheet

Test Number: 1

Date: 13JUL98

Number of Nodes: 2

Number of Available Slots: 494340

Number of Voice Gen/Node: 5

Number of Data Gen/Node: 5

Total Number of Packets Transmitted:	317146
Number of Sync Signals:	44940
Total Data Packets (CELL_ID = 1):	266090
Total Number of Voice Packets (CELL_ID = 1, EQUIP_ID = 1):	238934
Total Number of Data Packets (CELL_ID = 1, EQUIP_ID = 2):	27096
Total Number of Management Packets (CELL_ID = 2):	6176
Total Number of Call Requests (CELL_ID = 2, MSG_ID = 3):	2117
Total Number of Voice Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 1):	120
Total Number of Data Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 2):	1997
Other Management Messages:	1943
Total Number of Calls Approved (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 1):	1859
Total Number of Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0):	257
Total Number of Voice Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 1):	36
Total Number of Data Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 2):	221

ATM/HF Data Collection Sheet

Test Number: 2

Date: 17 JUL 98

Number of Nodes: 2

Number of Available Slots: 990k

Number of Voice Gen/Node: 5

Number of Data Gen/Node: 5

Total Number of Packets Transmitted:	638363
Number of Sync Signals:	90000
Total Data Packets (CELL_ID = 1):	536253
Total Number of Voice Packets (CELL_ID = 1, EQUIP_ID = 1):	483170
Total Number of Data Packets (CELL_ID = 1, EQUIP_ID = 2):	53083
Total Number of Management Packets (CELL_ID = 2):	12110
Total Number of Call Requests (CELL_ID = 2, MSG_ID = 3):	4194
Total Number of Voice Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 1):	240
Total Number of Data Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 2):	3954
Other Management Messages:	3722
Total Number of Calls Approved (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 1):	3604
Total Number of Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0):	590
Total Number of Voice Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 1):	84
Total Number of Data Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 2):	506

ATM/HF Data Collection Sheet

Test Number: 3

Date: 21 JUL 98

Number of Nodes: 2

Number of Available Slots: 990k

Number of Voice Gen/Node: 6

Number of Data Gen/Node: 6

Total Number of Packets Transmitted:	683559
Number of Sync Signals:	90000
Total Data Packets (CELL_ID = 1):	582399
Total Number of Voice Packets (CELL_ID = 1, EQUIP_ID = 1):	534021
Total Number of Data Packets (CELL_ID = 1, EQUIP_ID = 2):	48378
Total Number of Management Packets (CELL_ID = 2):	11160
Total Number of Call Requests (CELL_ID = 2, MSG_ID = 3):	3879
Total Number of Voice Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 1):	263
Total Number of Data Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 2):	3616
Other Management Messages:	3402
Total Number of Calls Approved (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 1):	3294
Total Number of Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0):	585
Total Number of Voice Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 1):	95
Total Number of Data Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 2):	490

ATM/HF Data Collection Sheet

Test Number: 4

Date: 29 JUL 98

Number of Nodes: 2

Number of Available Slots: 261990

Number of Voice Gen/Node: 8

Number of Data Gen/Node: 8

Total Number of Packets Transmitted:	180546
Number of Sync Signals:	23817
Total Data Packets (CELL_ID = 1):	152267
Total Number of Voice Packets (CELL_ID = 1, EQUIP_ID = 1):	134294
Total Number of Data Packets (CELL_ID = 1, EQUIP_ID = 2):	17973
Total Number of Management Packets (CELL_ID = 2):	4462
Total Number of Call Requests (CELL_ID = 2, MSG_ID = 3):	1603
otal Number of Voice Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 1):	107
Total Number of Data Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 2):	1496
Other Management Messages:	1257
Total Number of Calls Approved (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 1):	1213
Total Number of Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0):	389
Total Number of Voice Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 1):	56
Total Number of Data Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 2):	333

ATM/HF Data Collection Sheet

Test Number: 5	Date: 31 JUL 98
Number of Nodes: 2	Number of Available Slots: 990k
Number of Voice Gen/Node: 8	Number of Data Gen/Node: 8

Total Number of Packets Transmitted:	743835
Number of Sync Signals:	80000
Total Data Packets (CELL_ID = 1):	637300
Total Number of Voice Packets (CELL_ID = 1, EQUIP_ID = 1):	570685
Total Number of Data Packets (CELL_ID = 1, EQUIP_ID = 2):	66615
Total Number of Management Packets (CELL_ID = 2):	16535
Total Number of Call Requests (CELL_ID = 2, MSG_ID = 3):	5991
Total Number of Voice Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 1):	422
Total Number of Data Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 2):	5569
Other Management Messages:	4553
Total Number of Calls Approved (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 1):	4419
Total Number of Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0):	1572
Total Number of Voice Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 1):	233
Total Number of Data Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 2):	1339

ATM/HF Data Collection Sheet

Test Number: 6	Date: 5 AUG 98
Number of Nodes: 2	Number of Available Slots: 990k
Number of Voice Gen/Node: 3	Number of Data Gen/Node: 3

Total Number of Packets Transmitted:	515724
Number of Sync Signals:	90000
Total Data Packets (CELL_ID = 1):	419653
Total Number of Voice Packets (CELL_ID = 1, EQUIP_ID = 1):	390850
Total Number of Data Packets (CELL_ID = 1, EQUIP_ID = 2):	28803
Total Number of Management Packets (CELL_ID = 2):	6071
Total Number of Call Requests (CELL_ID = 2, MSG_ID = 3):	2023
Total Number of Voice Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 1):	156
Total Number of Data Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 2):	1867
Other Management Messages:	2025
Total Number of Calls Approved (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 1):	1976
Total Number of Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0):	47
Total Number of Voice Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 1):	8
Total Number of Data Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 2):	39

ATM/HF Data Collection Sheet

Test Number: 7	Date: 5 SEP 98
Number of Nodes: 2	Number of Available Slots: 638069
Number of Voice Gen/Node: 0	Number of Data Gen/Node: 5

Total Number of Packets Transmitted:	102947
Number of Sync Signals:	57370
Total Data Packets (CELL_ID = 1):	38186
Total Number of Voice Packets (CELL_ID = 1, EQUIP_ID = 1):	0
Total Number of Data Packets (CELL_ID = 1, EQUIP_ID = 2):	38186
Total Number of Management Packets (CELL_ID = 2):	7391
Total Number of Call Requests (CELL_ID = 2, MSG_ID = 3):	2464
Total Number of Voice Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 1):	0
Total Number of Data Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 2):	2464
Other Management Messages:	2463
Total Number of Calls Approved (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 1):	2463
Total Number of Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0):	1
Total Number of Voice Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 1):	0
Total Number of Data Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 2):	1

ATM/HF Data Collection Sheet

Test Number: 8

Date: 21 SEP 98

Number of Nodes: 2

Number of Available Slots: 519011

Number of Voice Gen/Node: 1

Number of Data Gen/Node: 4

Total Number of Packets Transmitted:	120923
Number of Sync Signals:	47183
Total Data Packets (CELL_ID = 1):	68091
Total Number of Voice Packets (CELL_ID = 1, EQUIP_ID = 1):	39119
Total Number of Data Packets (CELL_ID = 1, EQUIP_ID = 2):	28972
Total Number of Management Packets (CELL_ID = 2):	5649
Total Number of Call Requests (CELL_ID = 2, MSG_ID = 3):	1883
Total Number of Voice Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 1):	32
Total Number of Data Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 2):	1851
Other Management Messages:	1883
Total Number of Calls Approved (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 1):	1883
Total Number of Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0):	0
Total Number of Voice Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 1):	0
Total Number of Data Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 2):	0

ATM/HF Data Collection Sheet

Test Number: 9	Date: 26 SEP 98
Number of Nodes: 2	Number of Available Slots: 536400
Number of Voice Gen/Node: 2	Number of Data Gen/Node: 5

Total Number of Packets Transmitted:	220236
Number of Sync Signals:	48762
Total Data Packets (CELL_ID = 1):	164755
Total Number of Voice Packets (CELL_ID = 1, EQUIP_ID = 1):	132216
Total Number of Data Packets (CELL_ID = 1, EQUIP_ID = 2):	32539
Total Number of Management Packets (CELL_ID = 2):	6719
Total Number of Call Requests (CELL_ID = 2, MSG_ID = 3):	2232
Total Number of Voice Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 1):	65
Total Number of Data Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 2):	2167
Other Management Messages:	2255
Total Number of Calls Approved (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 1):	2196
Total Number of Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0):	36
Total Number of Voice Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 1):	0
Total Number of Data Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 2):	36

ATM/HF Data Collection Sheet

Test Number: 10

Date: 01 OCT 96

Number of Nodes: 2

Number of Available Slots: 542600

Number of Voice Gen/Node: 3

Number of Data Gen/Node: 5

Total Number of Packets Transmitted:	293634
Number of Sync Signals:	49325
Total Data Packets (CELL_ID = 1):	237833
Total Number of Voice Packets (CELL_ID = 1, EQUIP_ID = 1):	208208
Total Number of Data Packets (CELL_ID = 1, EQUIP_ID = 2):	29625
Total Number of Management Packets (CELL_ID = 2):	6476
Total Number of Call Requests (CELL_ID = 2, MSG_ID = 3):	2180
Total Number of Voice Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 1):	100
Total Number of Data Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 2):	2080
Other Management Messages:	2116
Total Number of Calls Approved (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 1):	2050
Total Number of Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0):	130
Total Number of Voice Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 1):	10
Total Number of Data Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 2):	120

ATM/HF Data Collection Sheet

Test Number: 11	Date: 06 OCT 98
Number of Nodes: 2	Number of Available Slots: 532453
Number of Voice Gen/Node: 4	Number of Data Gen/Node: 5

Total Number of Packets Transmitted:	319740
Number of Sync Signals:	48652
Total Data Packets (CELL_ID = 1):	264603
Total Number of Voice Packets (CELL_ID = 1, EQUIP_ID = 1):	235237
Total Number of Data Packets (CELL_ID = 1, EQUIP_ID = 2):	29366
Total Number of Management Packets (CELL_ID = 2):	6485
Total Number of Call Requests (CELL_ID = 2, MSG_ID = 3):	2202
Total Number of Voice Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 1):	116
Total Number of Data Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 2):	2086
Other Management Messages:	2081
Total Number of Calls Approved (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 1):	1998
Total Number of Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0):	204
Total Number of Voice Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 1):	23
Total Number of Data Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 2):	181

ATM/HF Data Collection Sheet

Test Number: 12	Date: 12 OCT 98
Number of Nodes: 2	Number of Available Slots: 541000
Number of Voice Gen/Node: 5	Number of Data Gen/Node: 5

Total Number of Packets Transmitted:	340522
Number of Sync Signals:	49235
Total Data Packets (CELL_ID = 1):	284637
Total Number of Voice Packets (CELL_ID = 1, EQUIP_ID = 1):	256667
Total Number of Data Packets (CELL_ID = 1, EQUIP_ID = 2):	28170
Total Number of Management Packets (CELL_ID = 2):	6450
Total Number of Call Requests (CELL_ID = 2, MSG_ID = 3):	2244
Total Number of Voice Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 1):	131
Total Number of Data Call Requests (CELL_ID = 2, MSG_ID = 3, EQUIP_TYPE = 2):	2113
Other Management Messages:	1964
Total Number of Calls Approved (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 1):	1913
Total Number of Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0):	329
Total Number of Voice Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 1):	44
Total Number of Data Calls Blocked (CELL_ID = 2, MSG_ID = 5, ACCESS_CODE = 0, EQUIP_ID = 2):	285

Appendix B

B.1 List of Attributes and Codes:

1. CELL_ID	0 - INVALID_CELL 1 DATA_CELL 2 - MANAGEMENT_CELL
2. MSG_ID	0 - INVALID_MSG 1 NODE_CALL_SETUP 2 NODE_ACCESS_MSG 3 - NET_CONNECT_REQ 4 NET_DISCONNECT_REQ 5 - NET_ACCESS_MSG 6 - NODE_CALL_TAKEDOWN 7 CHANGE_SLOT_ASSIGN 8 - CHANGE_DATA_RATE
3. ACCESS_CODE	0 - ACCESS_DENIED 1 _ ACCESS_APPROVED
4. BROADCAST_MSG	111 in the TO_NODE_ID attribute of message
5. TANK_TYPE	1 - CONNECT 2 - DISCONNECT

6. FM_NODE_ID
7. TO_NODE_ID
8. EQUIP_ID
9. MIN_RATE
10. MAX_RATE
11. REQ_SLOTS
12. RATE
13. START_SLOT
14. SYNC_SIG
15. FRAME_NUM
16. CALL_DUR

B.2 Messages with attributes:

NODE_CALL_SETUP	
EQUIP_ID	
CELL_ID	2
MSG_ID	1
MIN_RATE	
MAX_RATE	
REQ_SLOTS	

NET_CONNECT_REQ	
FM_NODE_ID	
TO_NODE_ID	
EQUIP_ID	
CELL_ID	2
MSG_ID	3
MIN_RATE	
MAX_RATE	
REQ_SLOTS	

NET_ACCESS_MSG	
FM_NODE_ID	
TO_NODE_ID	
CELL_ID	2
MSG_ID	5
EQUIP_ID	
ACCESS_CODE	
RATE	
START_SLOT	

NET_DISCONNECT_REQ	
FM_NODE_ID	
TO_NODE_ID	
EQUIP_ID	
CELL_ID	2
MSG_ID	4
RATE	
START_SLOT	
ACCESS_CODE	
CALL_DUR	

NODE_ACCESS_MSG (Appr'd)	
EQUIP_ID	
CELL_ID	2
MSG_ID	2
ACCESS_CODE	1
RATE	
START_SLOT	
REQ_SLOTS	

NODE_ACCESS_MSG (Denied)	
EQUIP_ID	
CELL_ID	2
MSG_ID	2
ACCESS_CODE	0

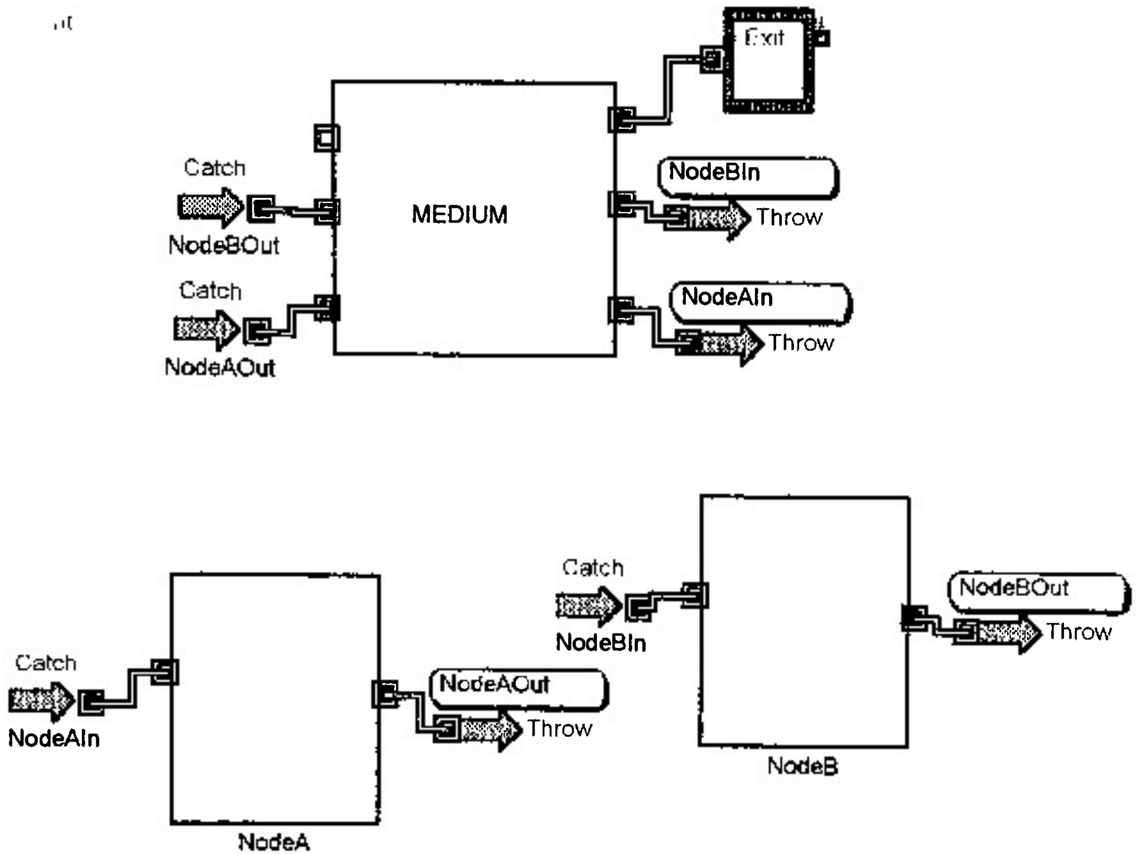
NODE_CALL_TAKEDOWN	
EQUIP_ID	
CELL_ID	2
MSG_ID	6
RATE	
ACCESS_CODE	1
CALL_DUR	
START_SLOT	

SYNC_SIG	
SYNC_SIG	1
FRAME_NUM	1-25
NODE_ID	
OPEN_SLOTS (not used)	

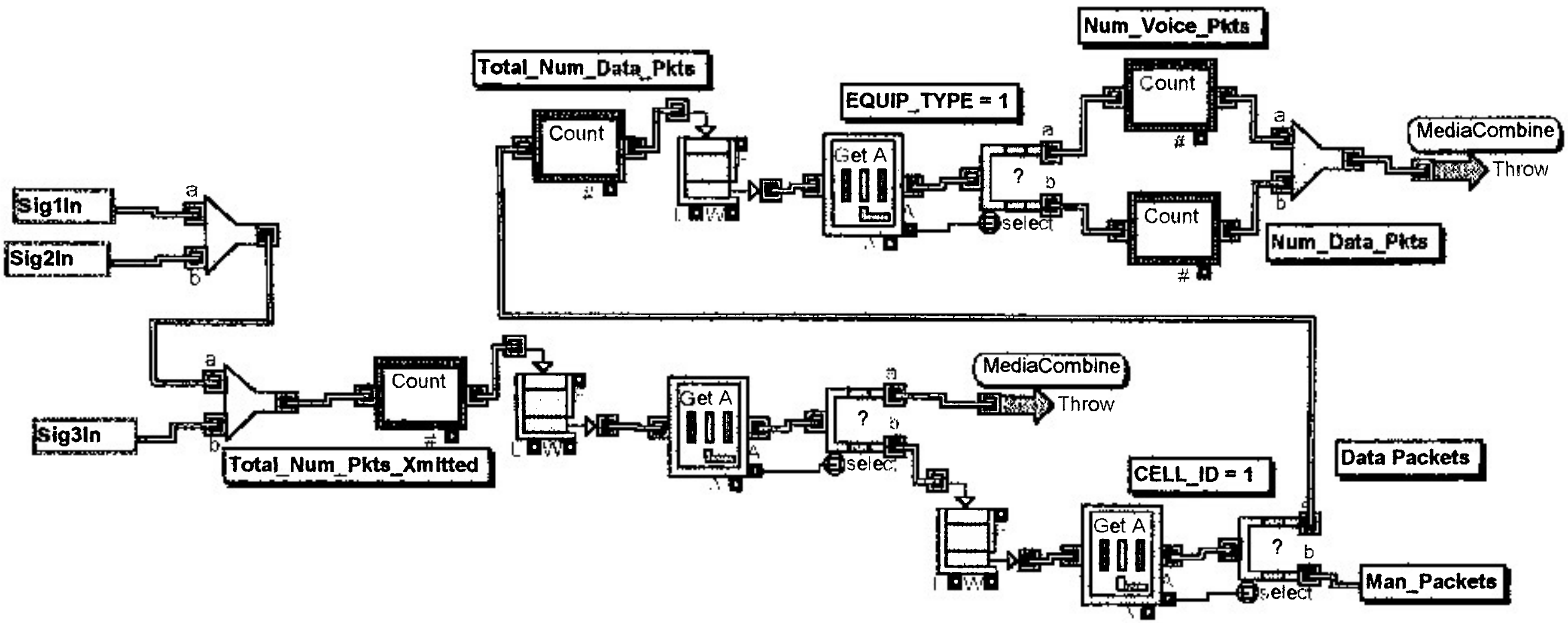
TANK_MSG	
TYPE	
RATE	

Appendix C: Simulation Block Diagrams

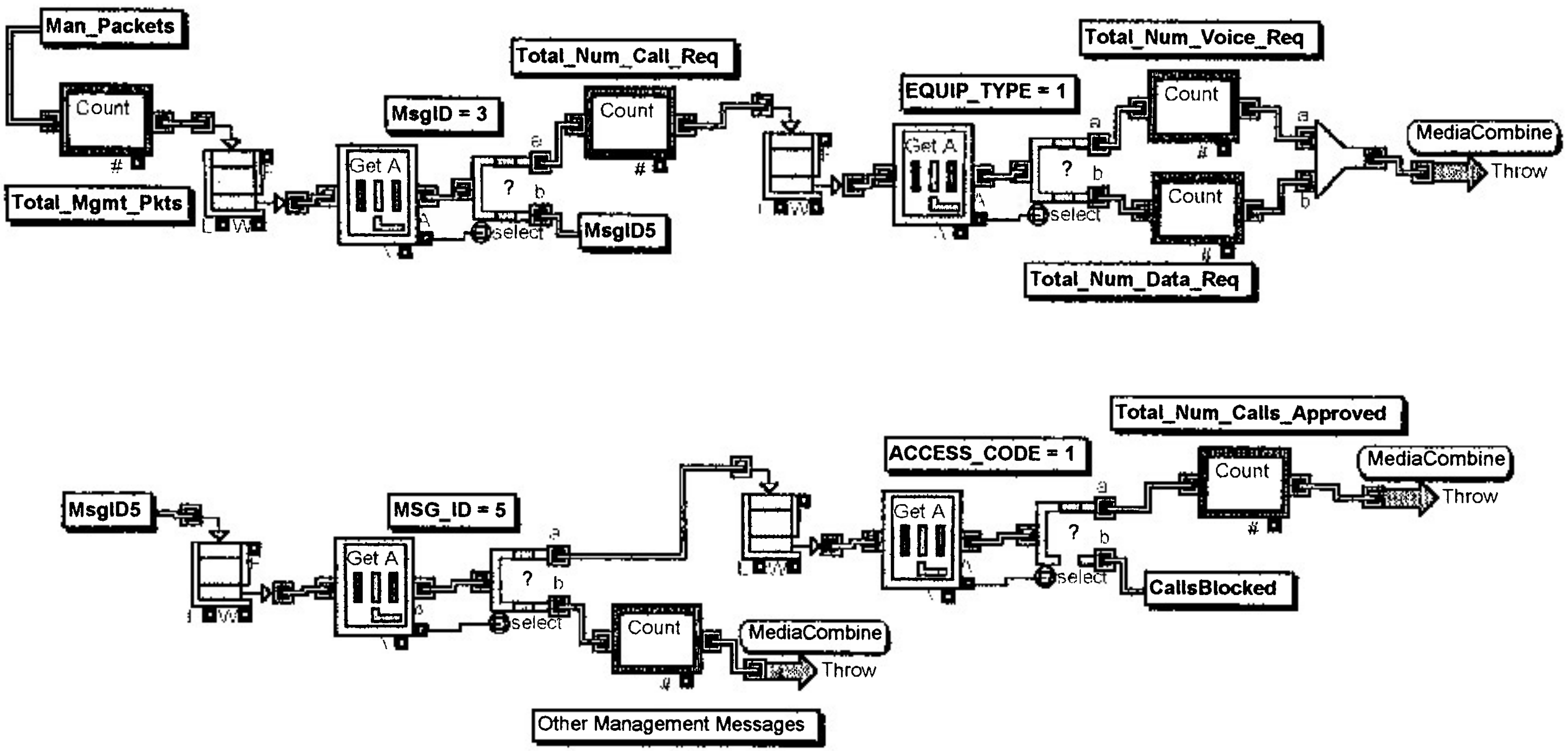
ATM/HF Overall Block Diagram



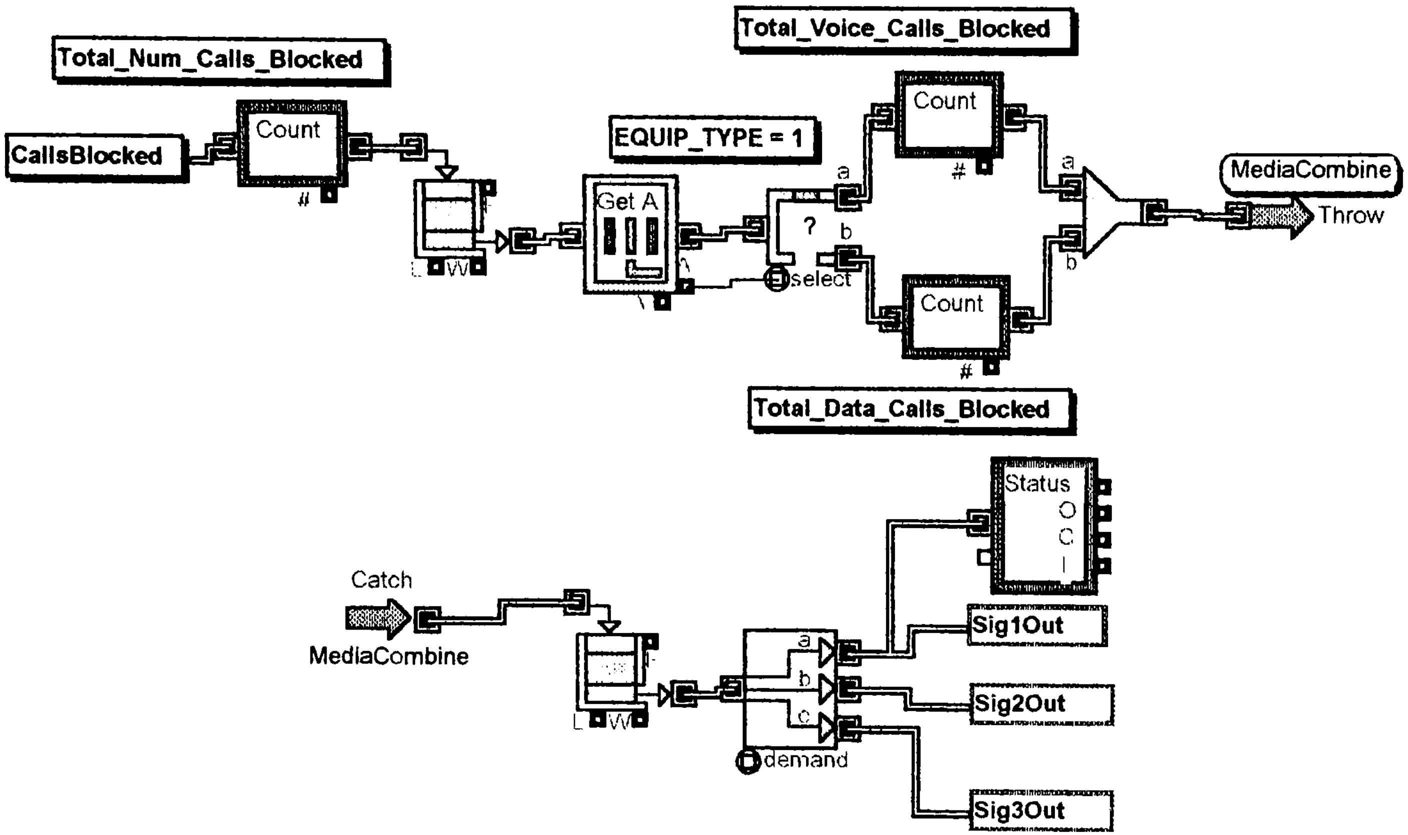
Block Diagram of Medium Module

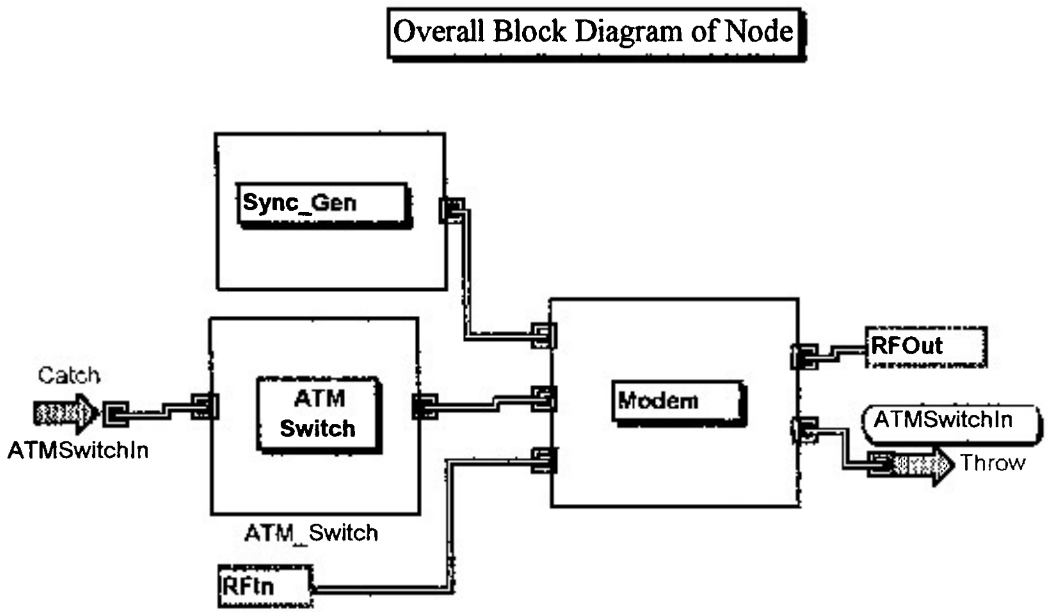


(Medium Module cont.)

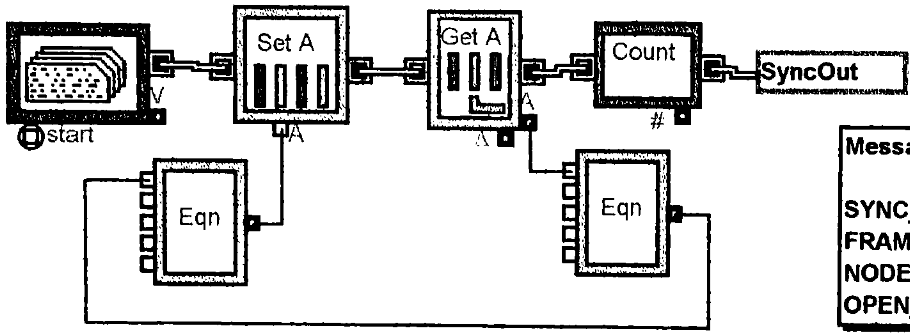


(Medium Module cont.)




Overall Block Diagram of Node
Sync_Gen
Catch
ATMSwitchIn
ATM
Switch
ATM_Switch
RFIn
Modem
RFOut
ATMSwitchIn
Throw

Block Diagram of Sync_Gen

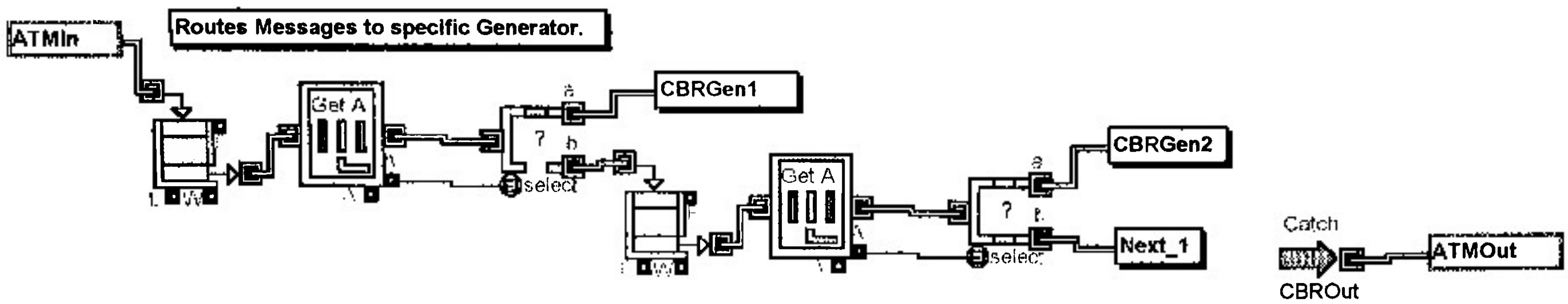


Message Format:

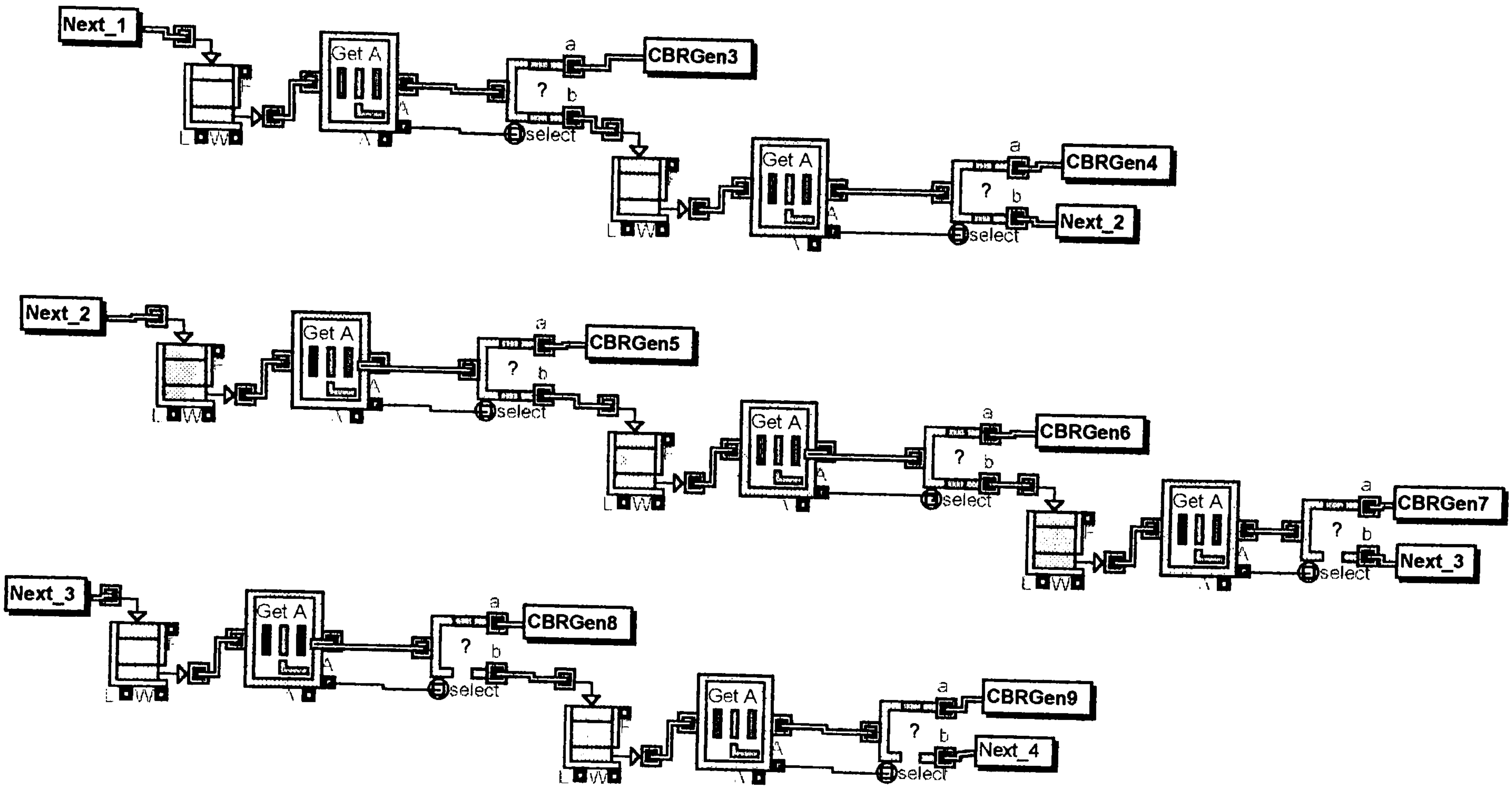
SYNC_SIG = 1
FRAME_NUM
NODE_ID = 1
OPEN_SLOTS = 0

Need to add number of open slots where zero slots represents NET_BUSY.

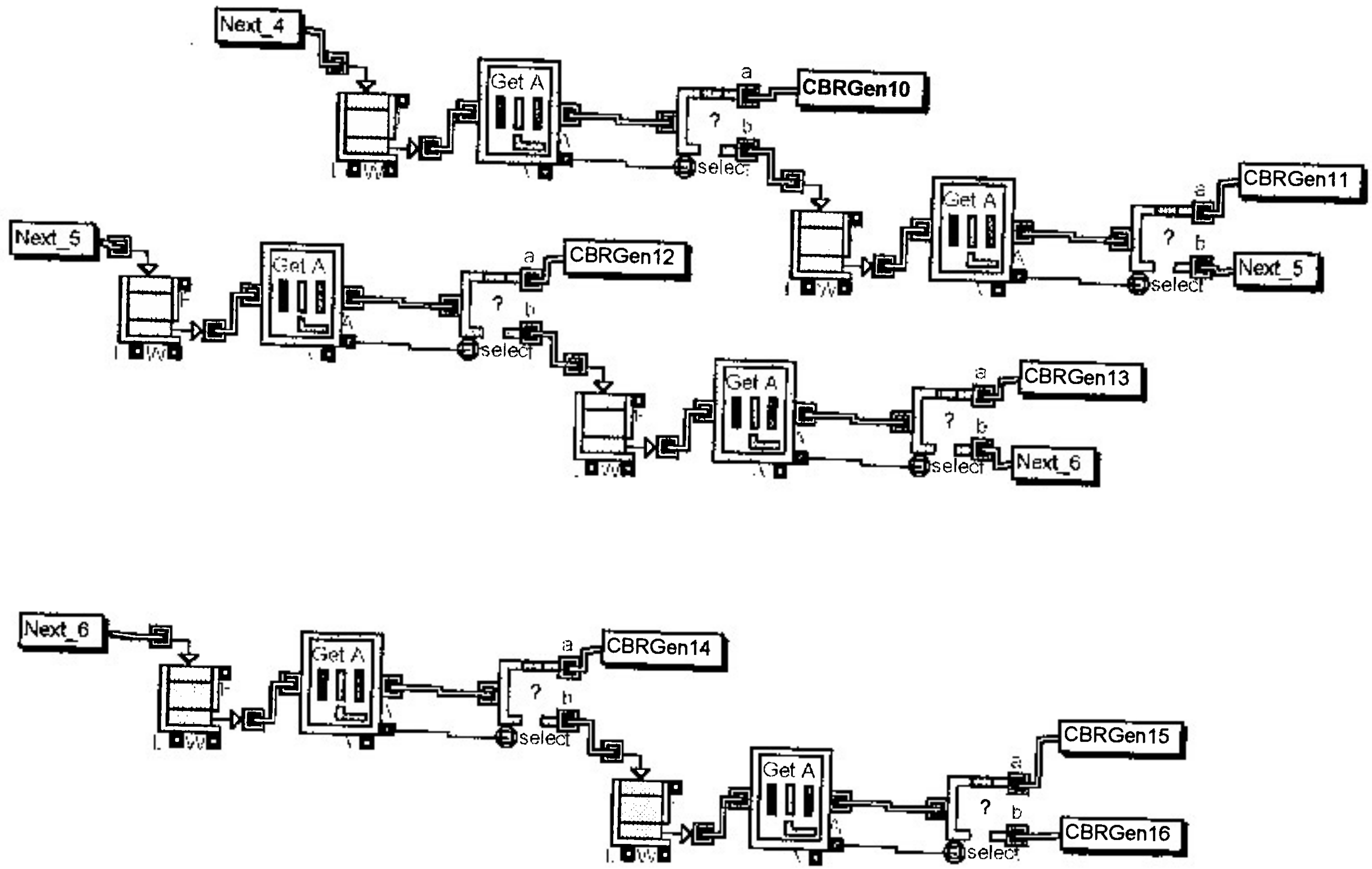
Block Diagram of ATM_Switch



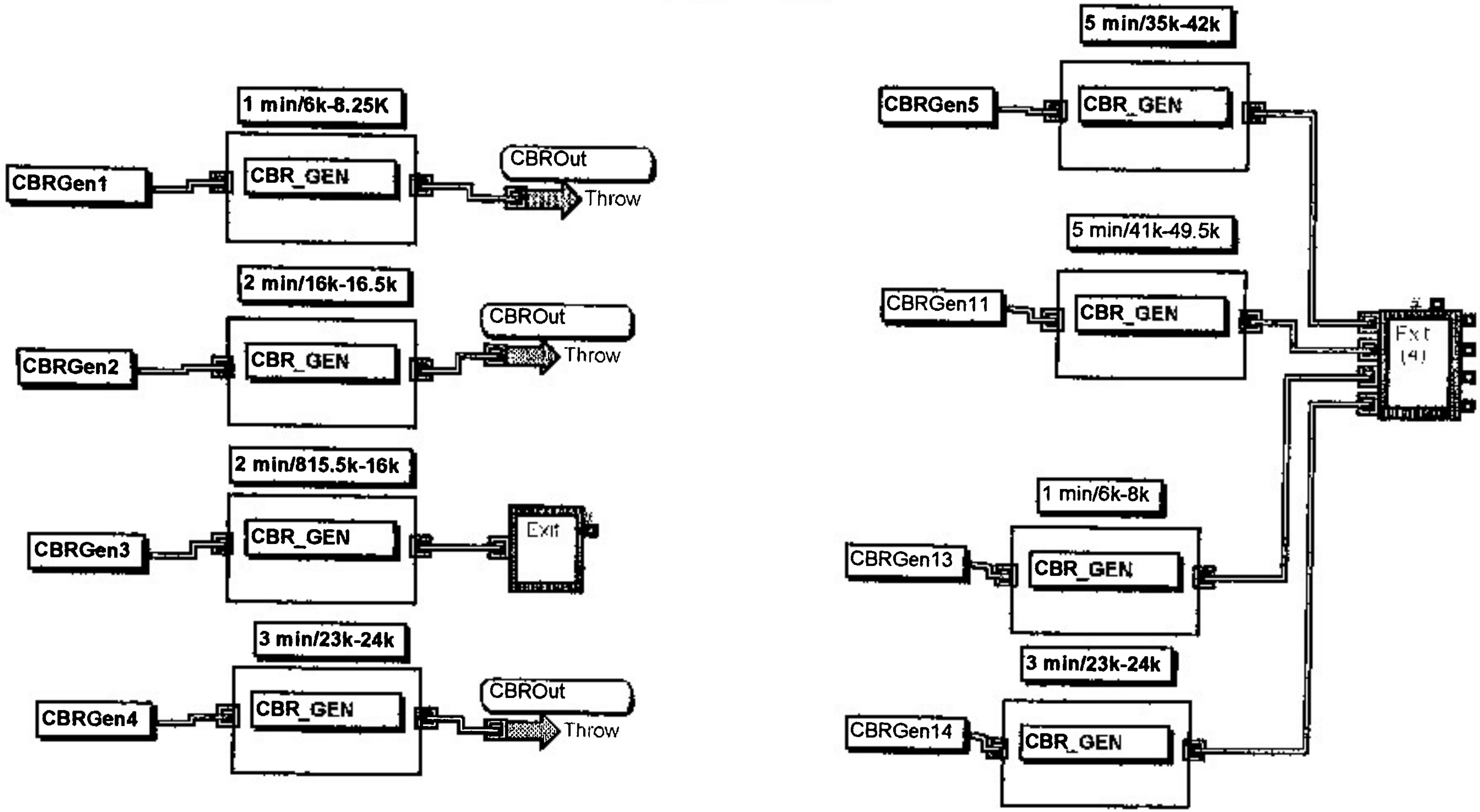
(ATM_Switch cont.)



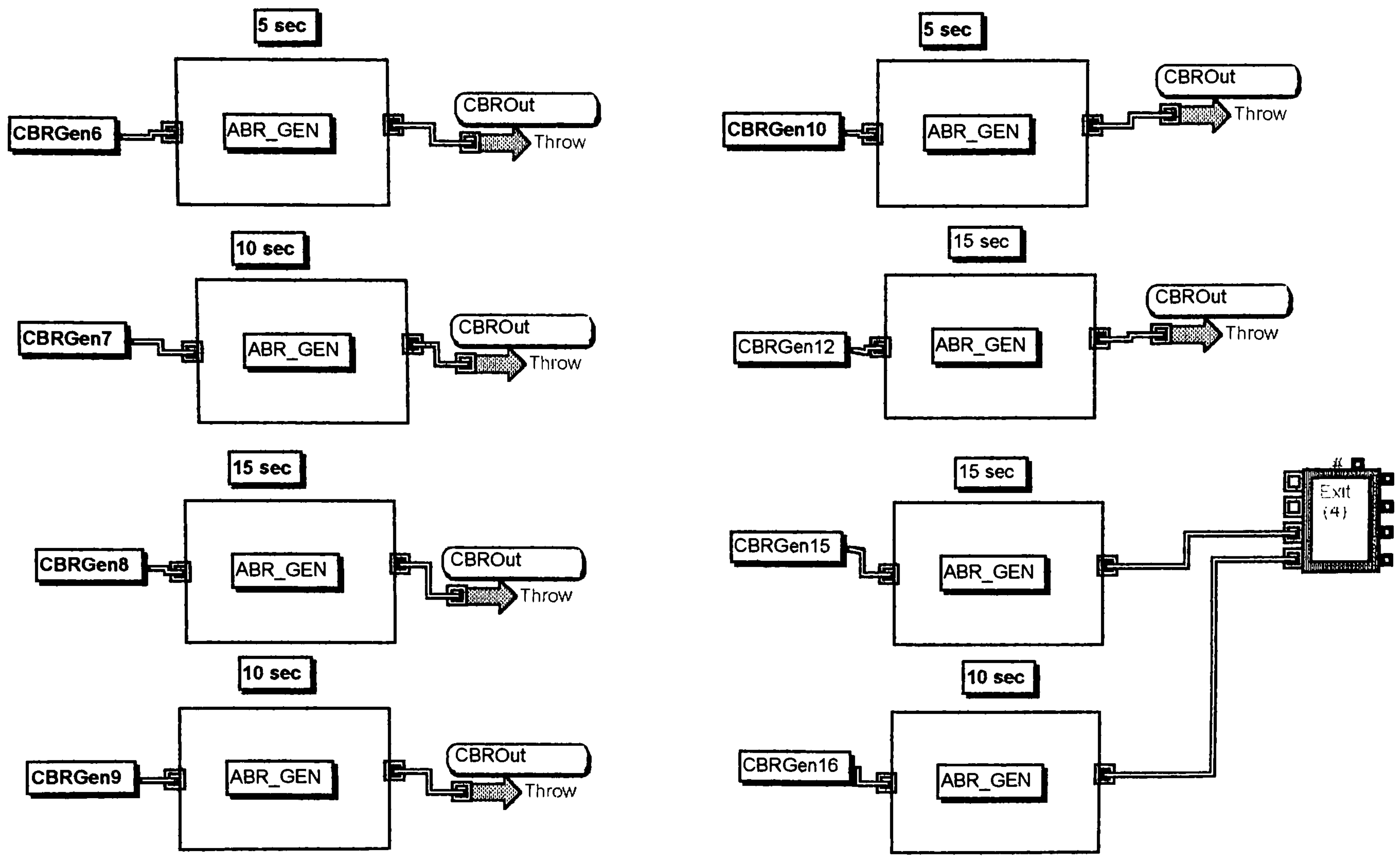
(ATM_Switch cont.)



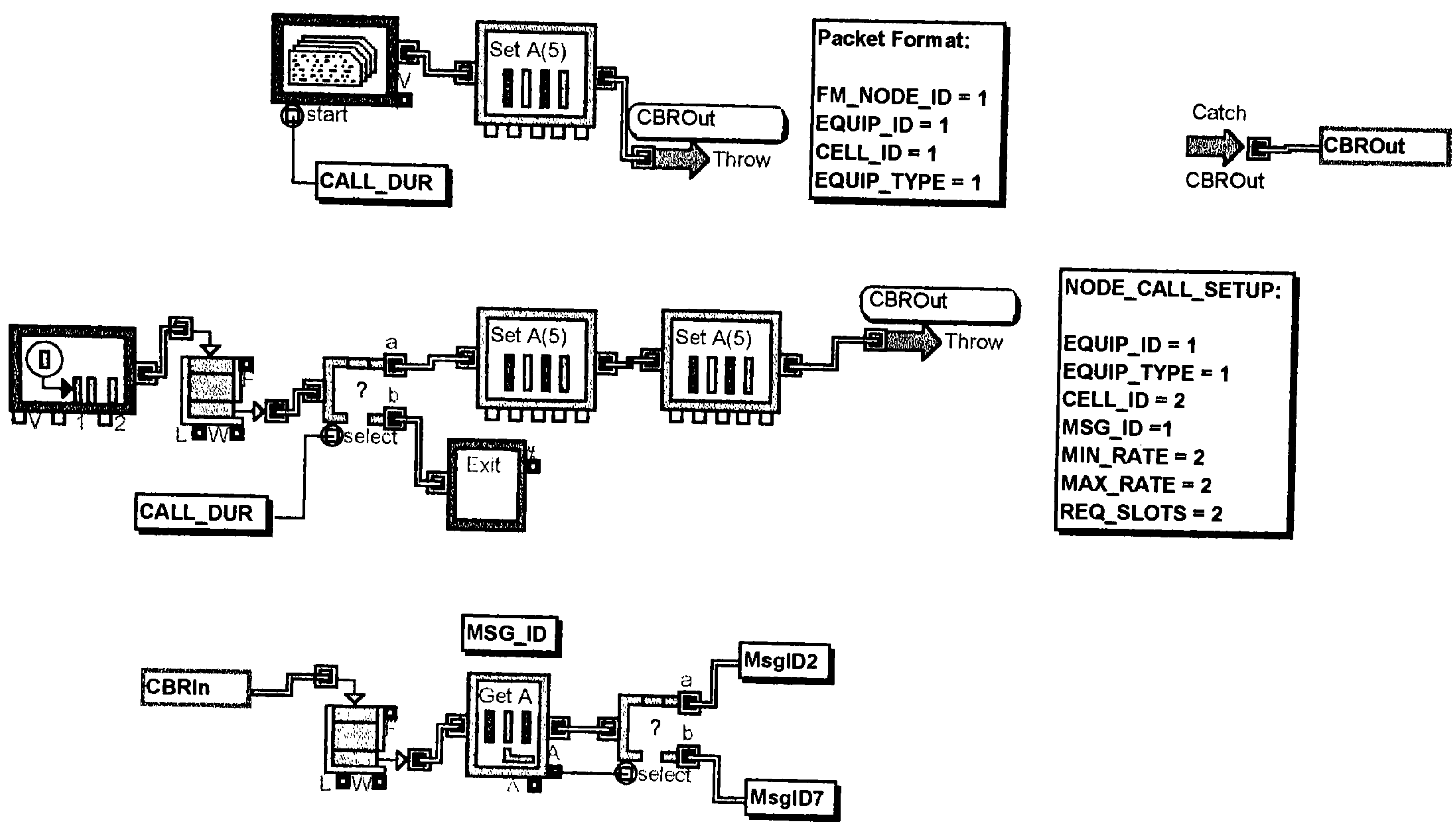
(ATM_Switch cont.)



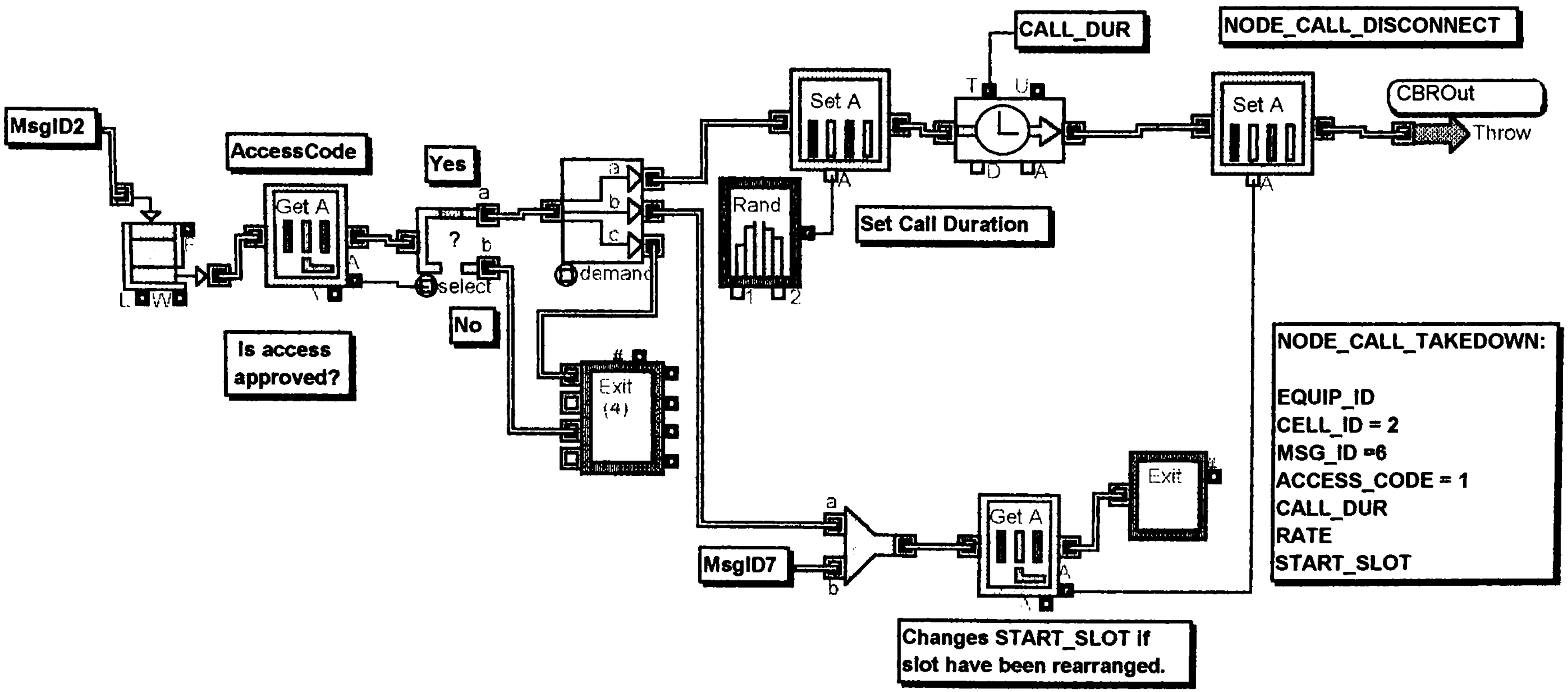
(ATM_Switch cont.)



Block Diagram of CBR_GEN

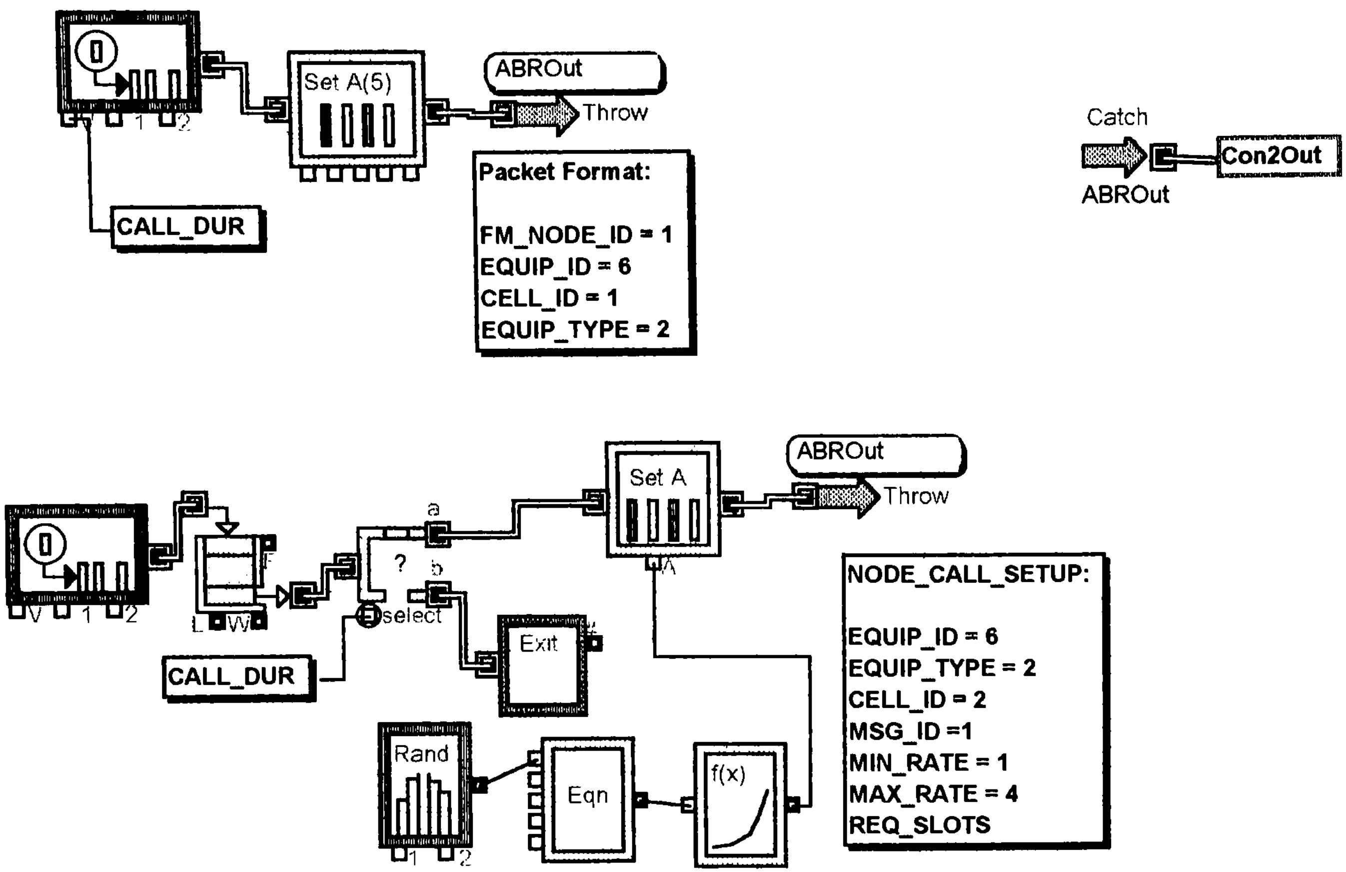


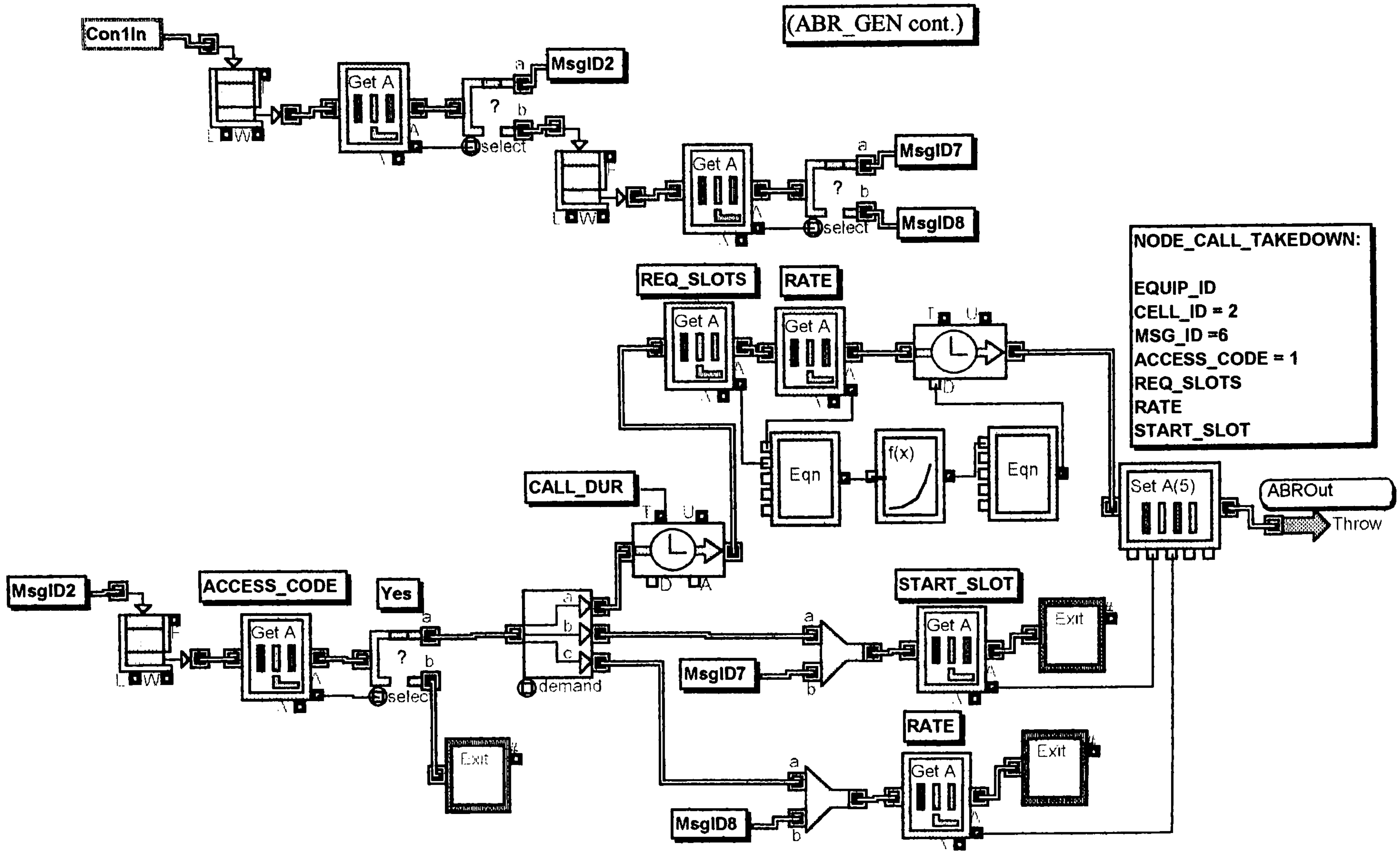
(CBR_GEN cont.)



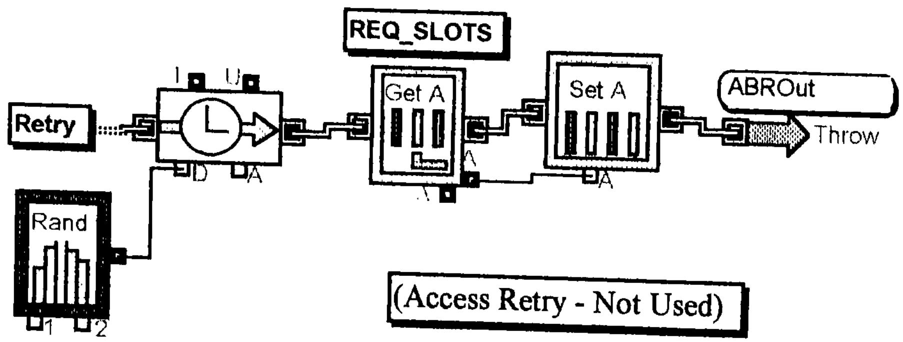
MsgID2
AccessCode
Get A
Is access approved?
Yes
No
select
demand
Exit (4)
Set A
Rand
Set Call Duration
CALL_DUR
NODE_CALL_DISCONNECT
Set A
CBROut
Throw
MsgID7
Get A
Exit
Changes START_SLOT if slot have been rearranged.
NODE_CALL_TAKEDOWN:
EQUIP_ID
CELL_ID = 2
MSG_ID =6
ACCESS_CODE = 1
CALL_DUR
RATE
START_SLOT

Block Diagram of ABR__GEN



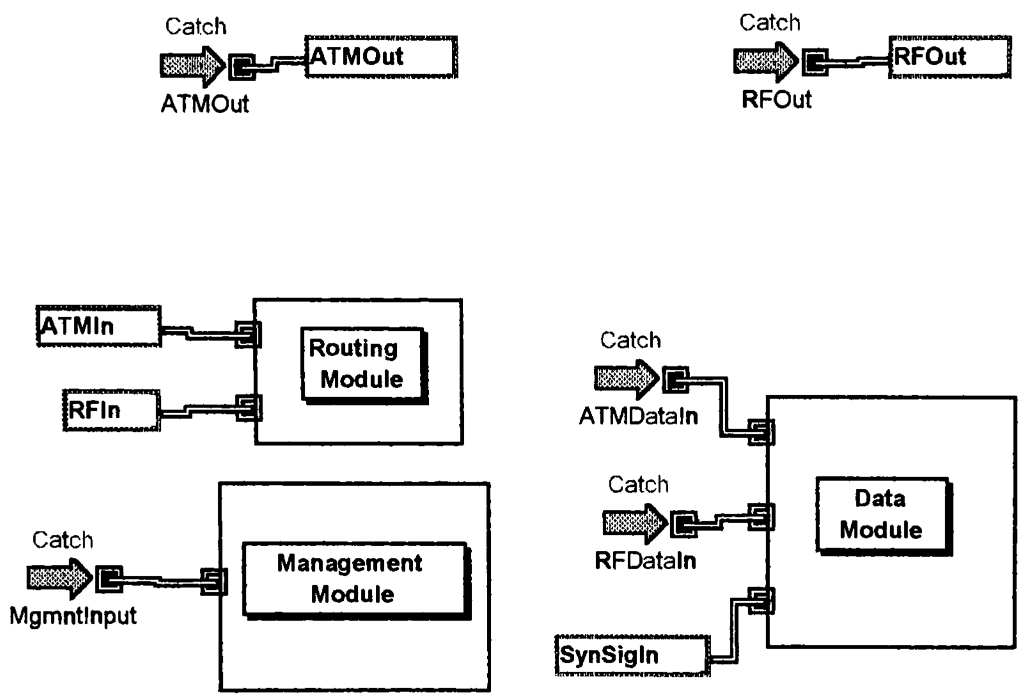

(ABR_GEN cont.)
Con1In
Get A
MsgID2
select
MsgID7
MsgID8
NODE_CALL_TAKEDOWN:
EQUIP_ID
CELL_ID = 2
MSG_ID =6
ACCESS_CODE = 1
REQ_SLOTS
RATE
START_SLOT
REQ_SLOTS
RATE
Eqn
f(x)
Eqn
CALL_DUR
Set A(5)
ABROut
Throw
ACCESS_CODE
Yes
demand
Exit
START_SLOT
RATE

(ABR_GEN cont.)

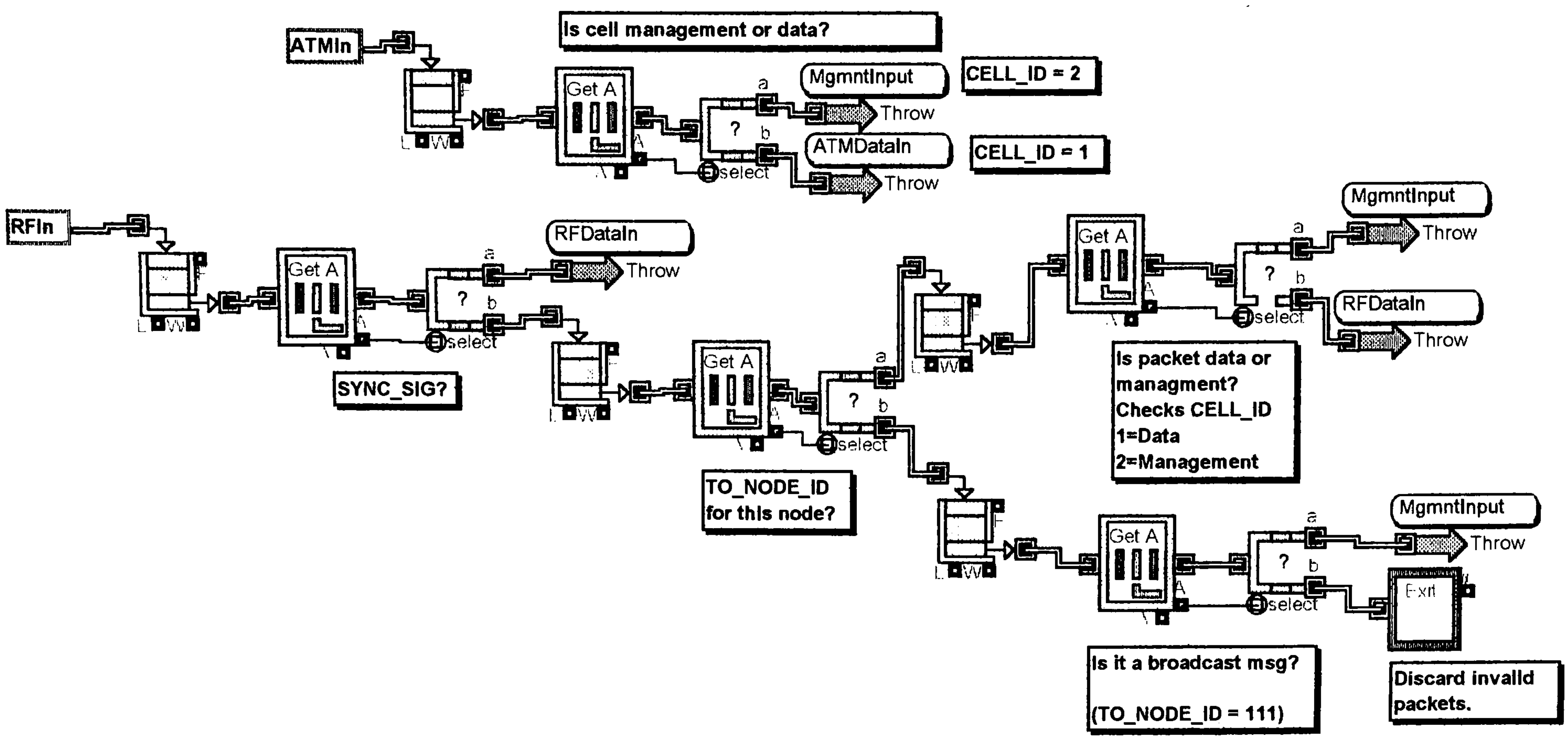


(Access Retry - Not Used)

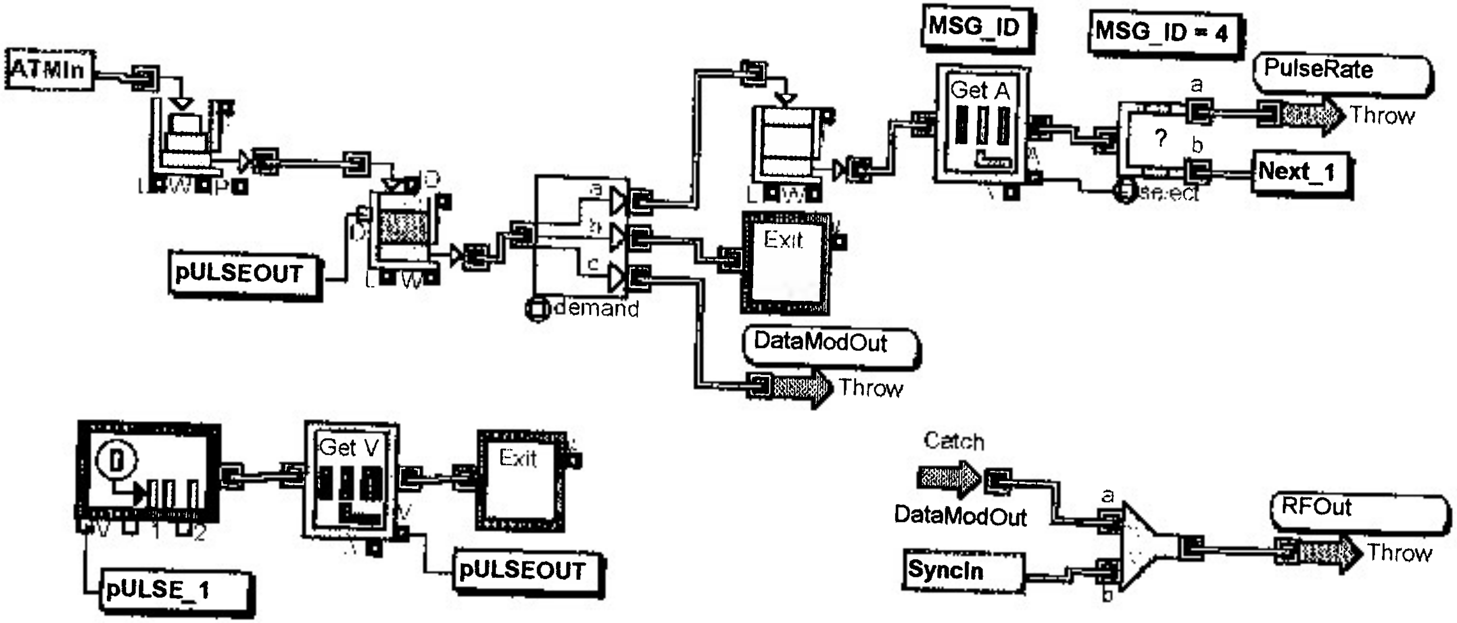
Overall Block Diagram of Modem Module



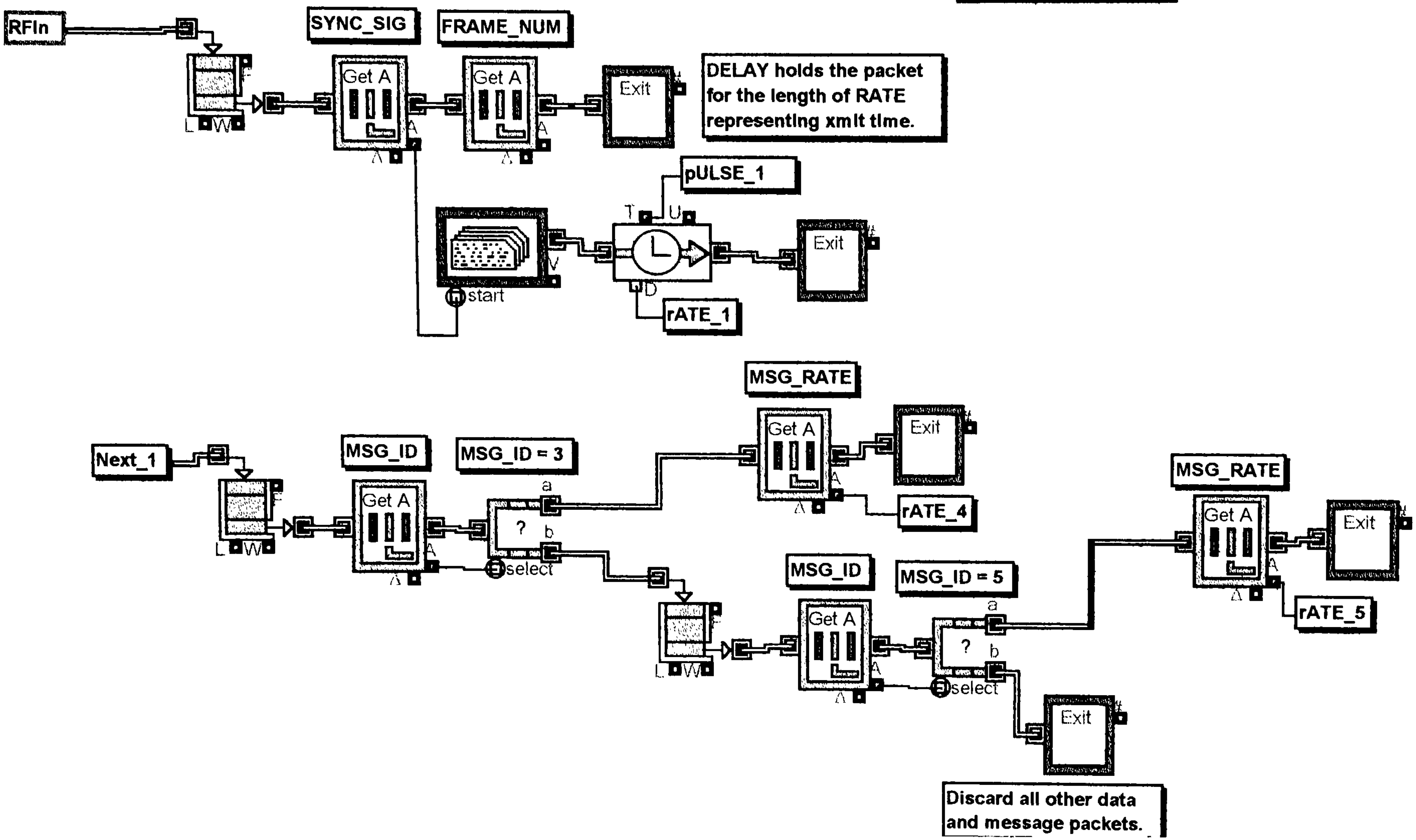
Block Diagram of Routing Module



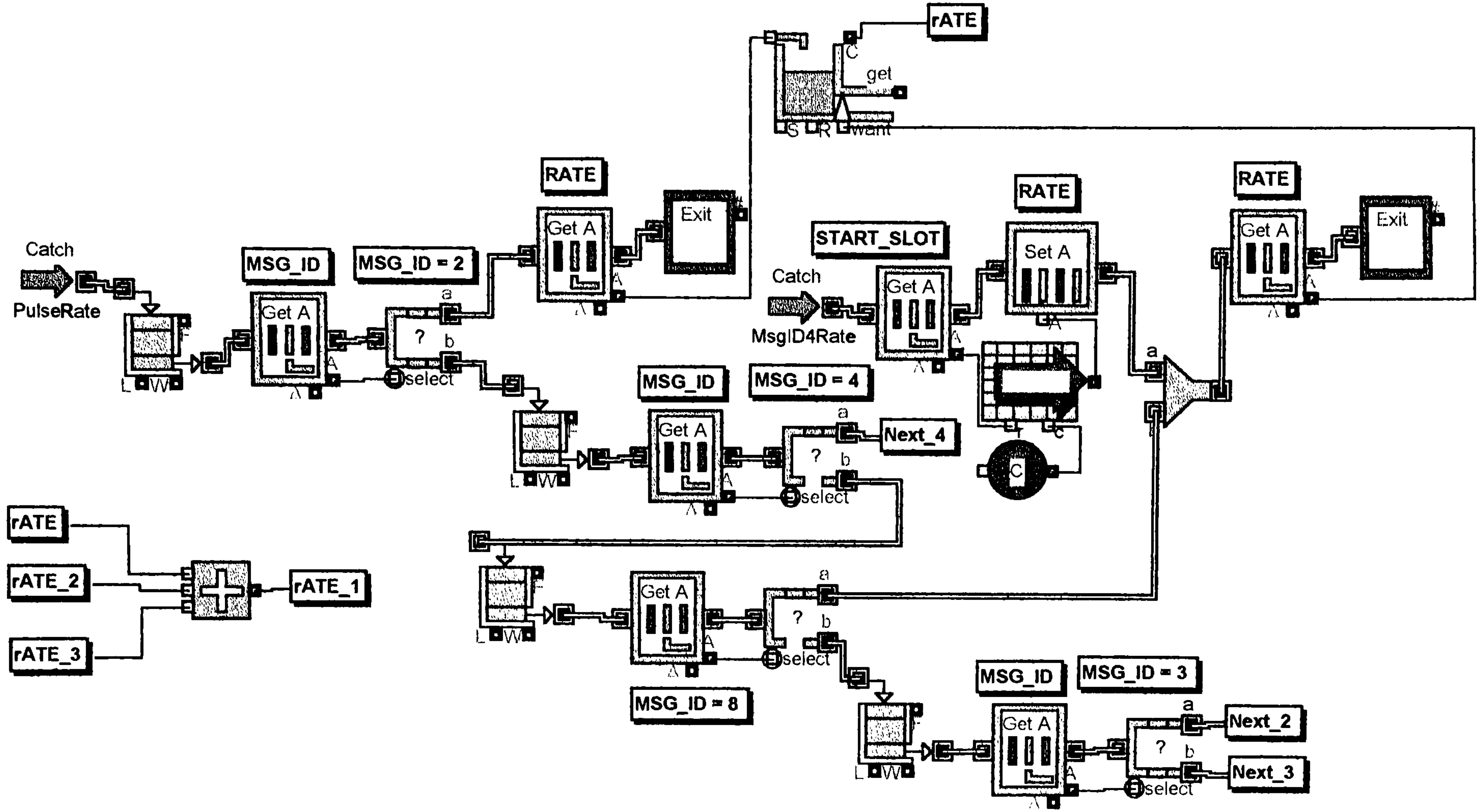
Block Diagram of Data Module



(Data Module cont.)

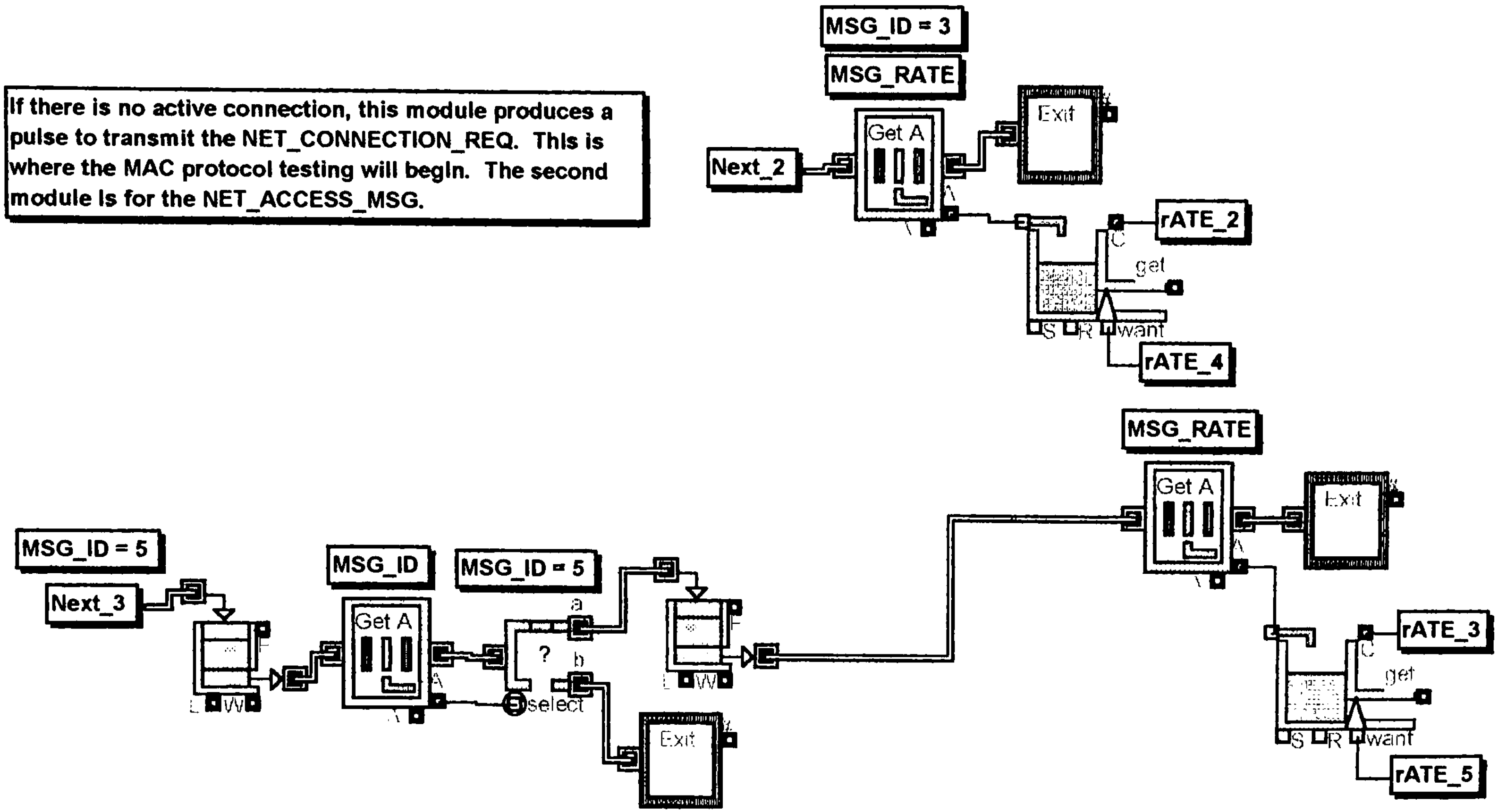


(Data Module cont.)

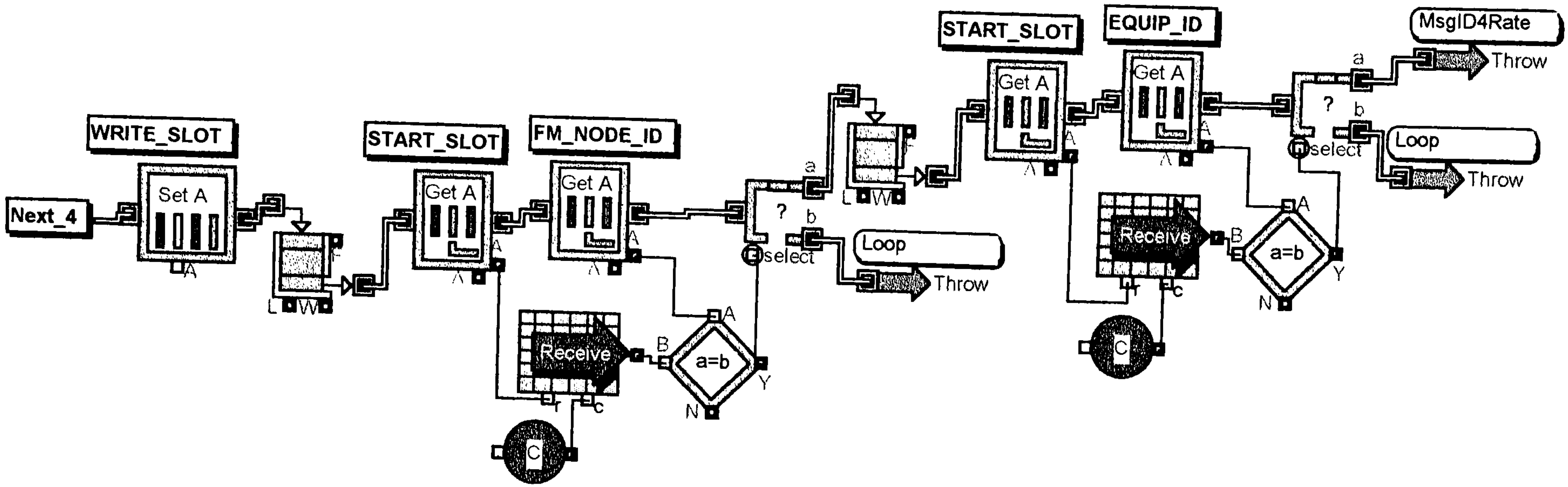


(Data Module cont.)

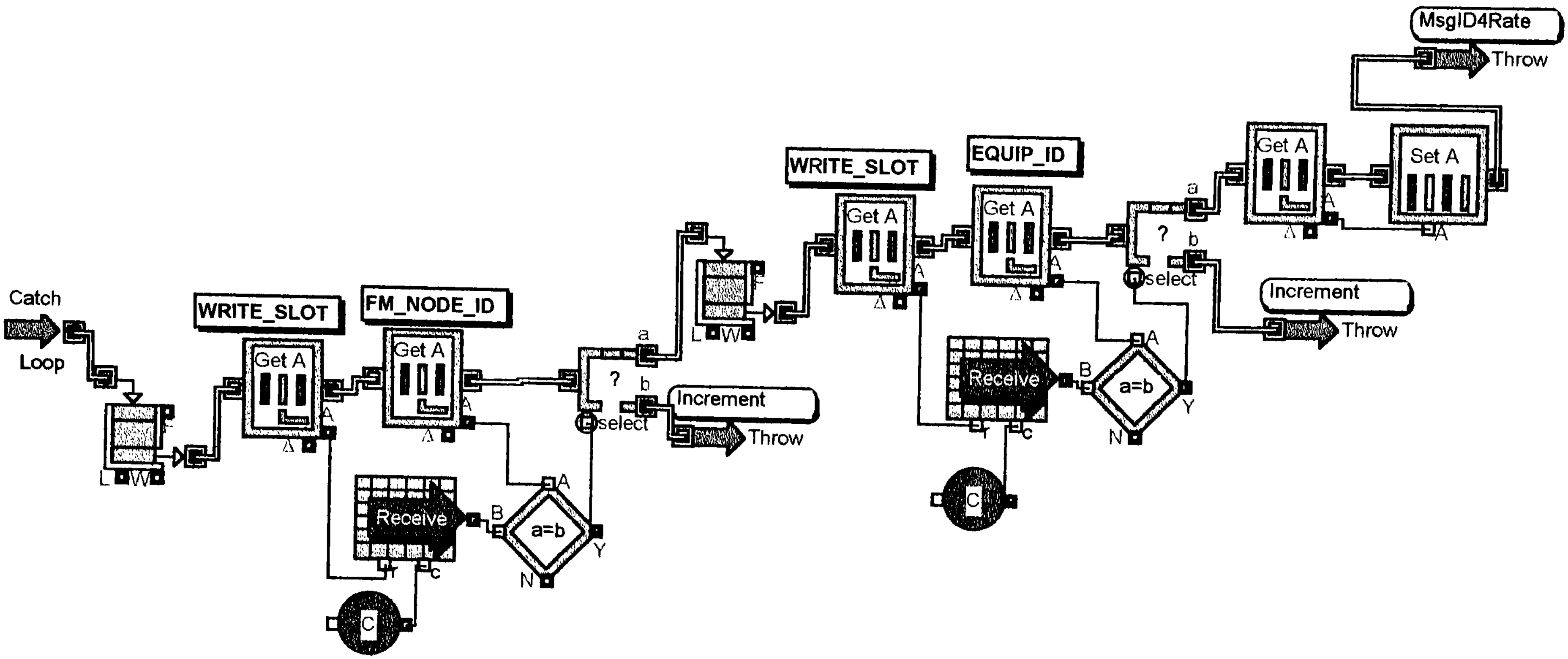
If there is no active connection, this module produces a pulse to transmit the NET_CONNECTION_REQ. This is where the MAC protocol testing will begin. The second module is for the NET_ACCESS_MSG.



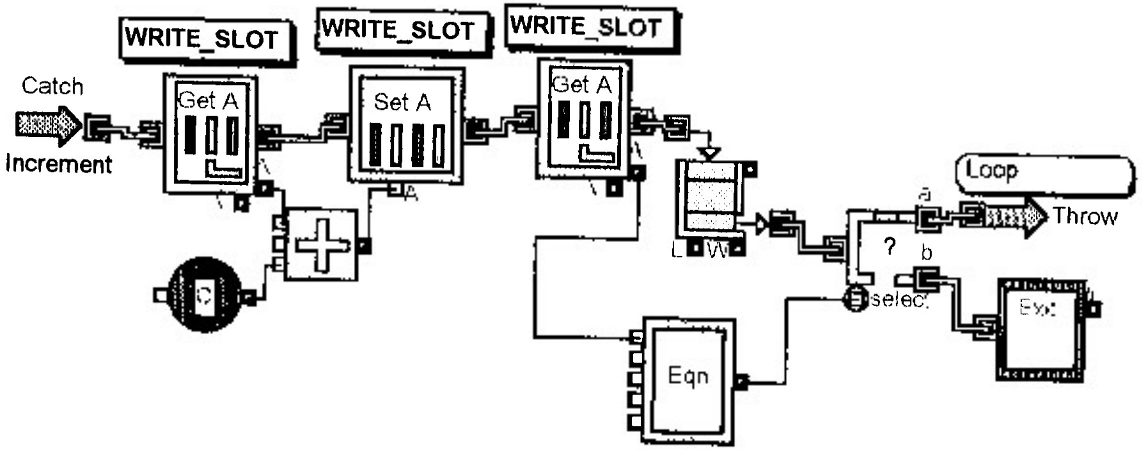
(Data Module cont.)



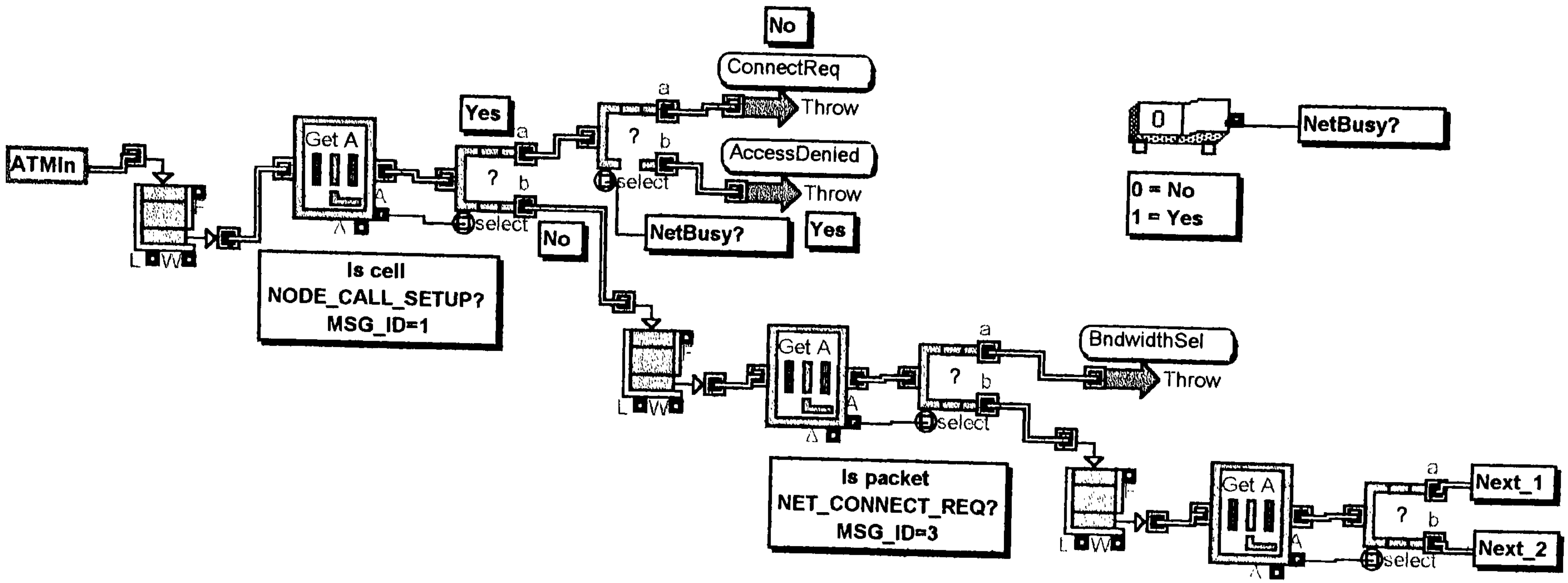
(Data Module cont.)



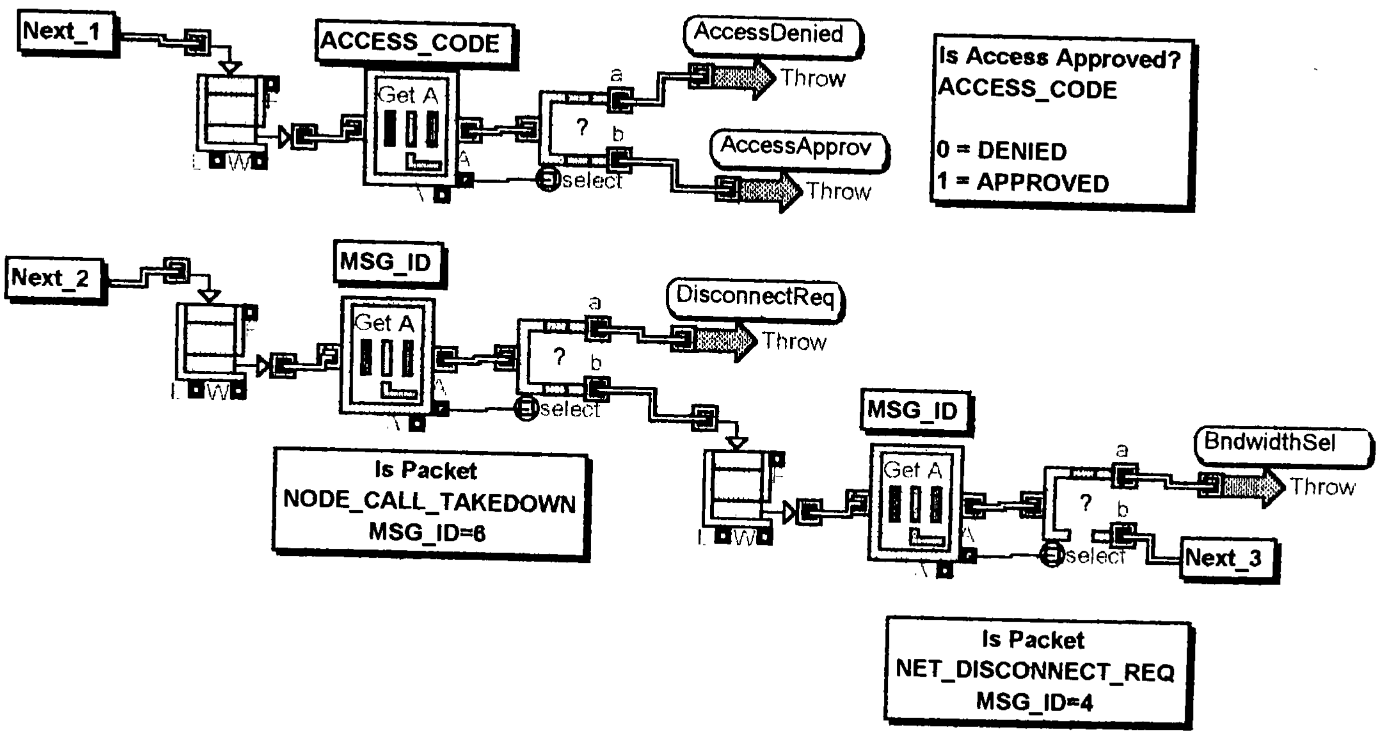
(Data Module cont.)



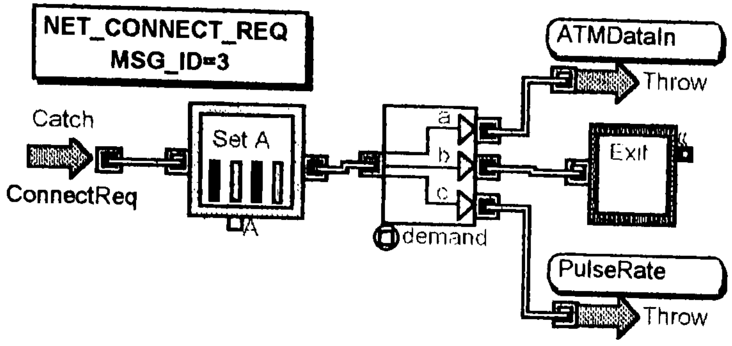
Block Diagram of Management Module



(Management Module cont.)



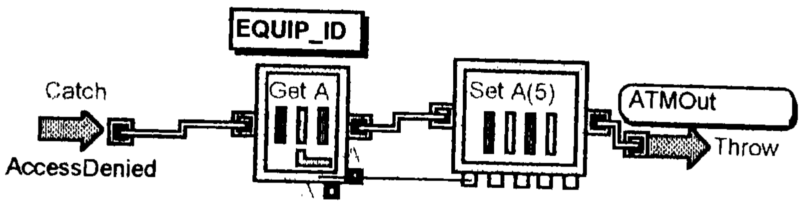
(Management Module cont.)



NET_CONNECT_REQ:

FM_NODE_ID = 1
TO_NODE_ID = 1
EQUIP_ID
EQUIP_TYPE
CELL_ID = 2
MSG_ID = 3
MIN_RATE
MAX_RATE
REQ_SLOTS
MSG_RATE = 1

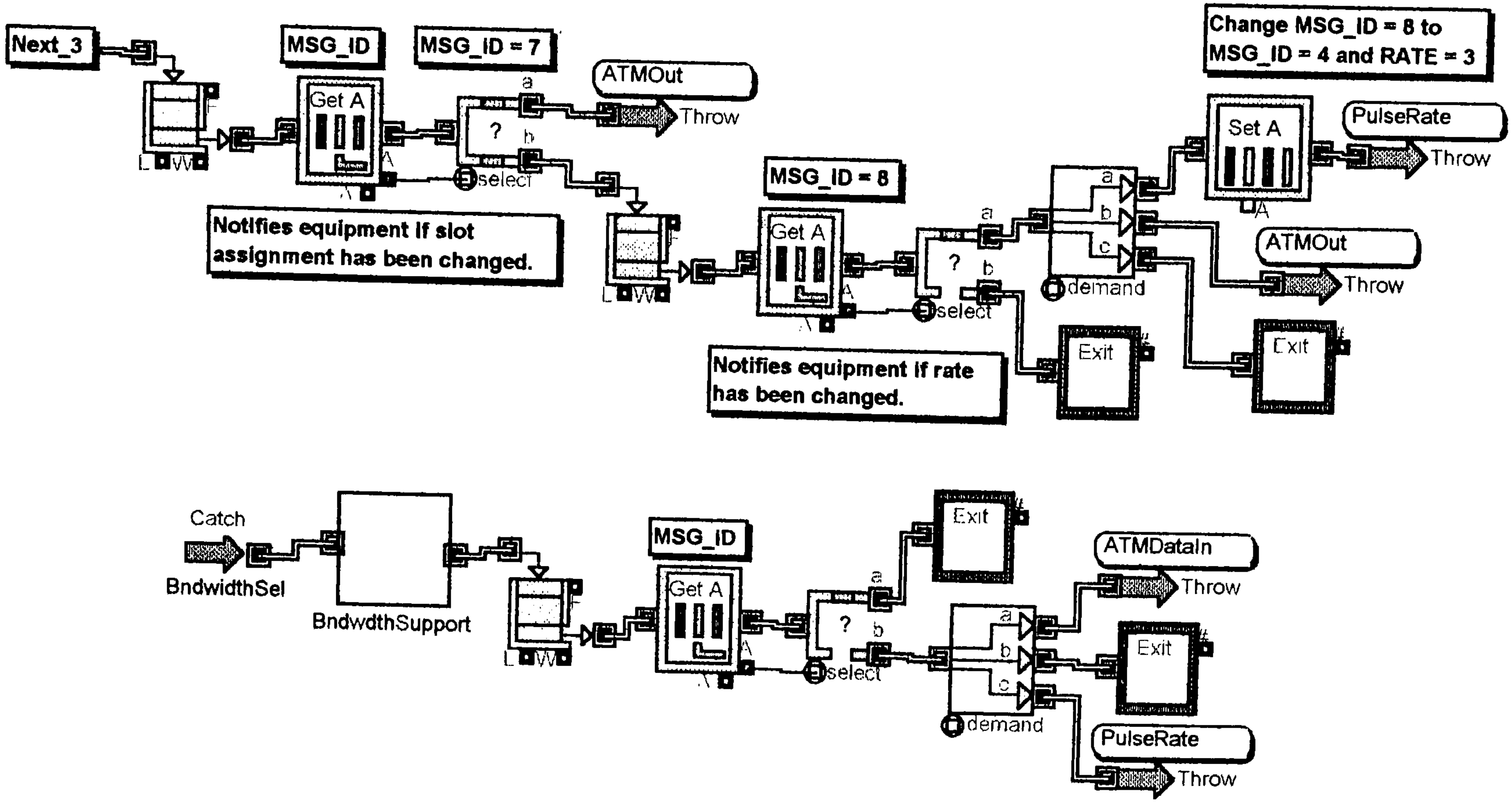
Is Packet
NET_ACCESS_MSG
MSG_ID=5

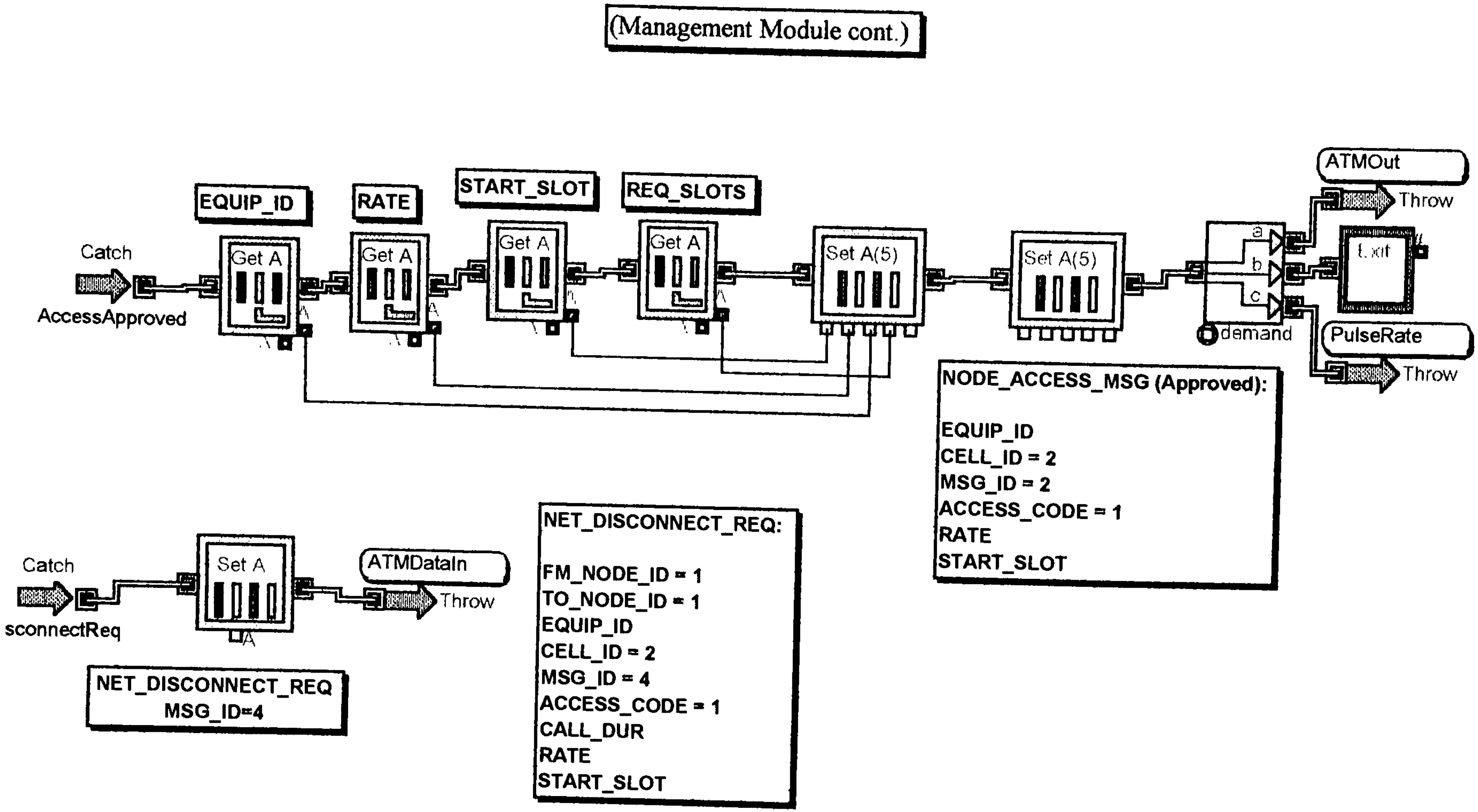


NODE_ACCESS_MSG (Denied):

EQUIP_ID
CELL_ID = 2
MSG_ID = 2
ACCESS_CODE=0

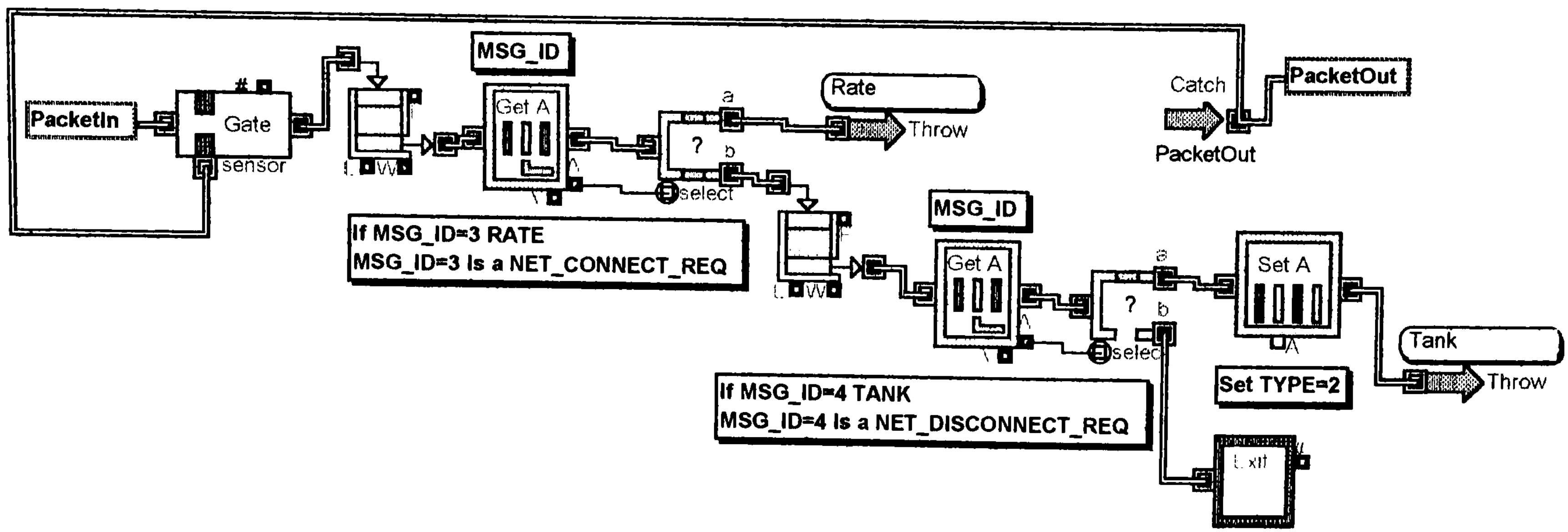
(Management Module cont.)



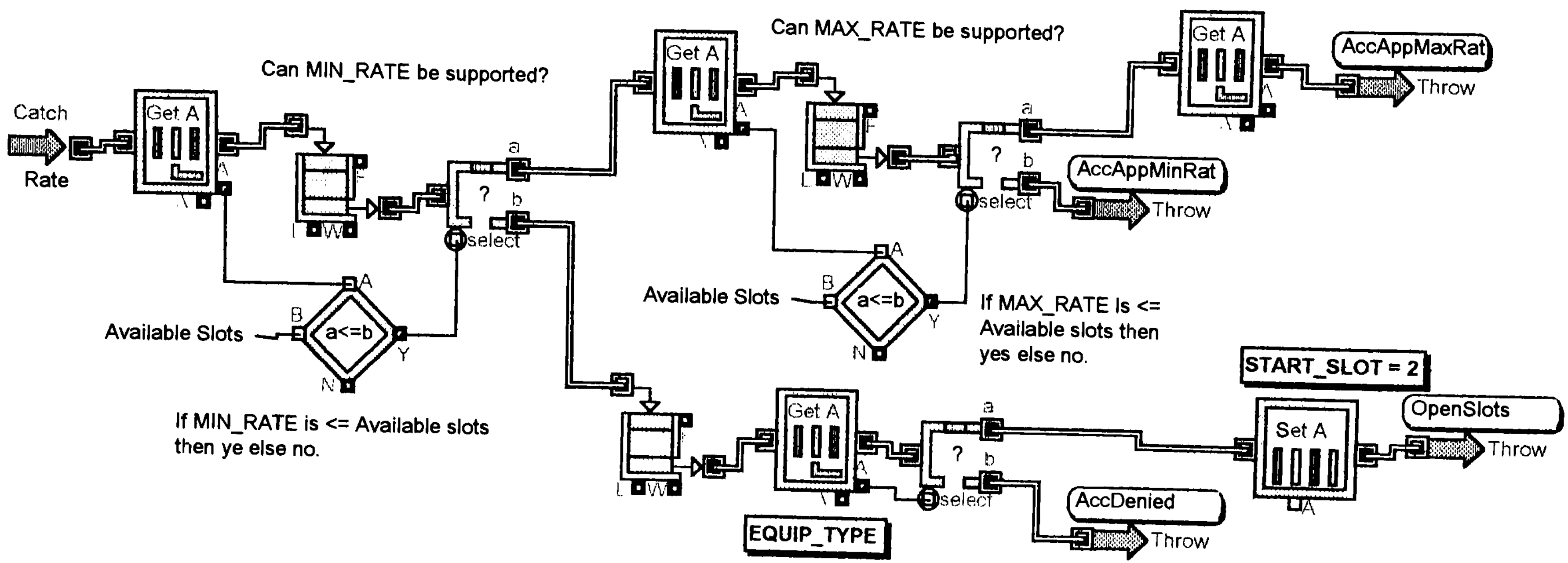


(Management Module cont.)
Catch
AccessApproved
EQUIP_ID
Get A
RATE
Get A
START_SLOT
Get A
REQ_SLOTS
Get A
Set A(5)
Set A(5)
a
b
c
demand
ATMOut
Throw
Exit
PulseRate
Throw
NODE_ACCESS_MSG (Approved):
EQUIP_ID
CELL_ID = 2
MSG_ID = 2
ACCESS_CODE = 1
RATE
START_SLOT
Catch
sconnectReq
Set A
A
ATMDataIn
Throw
NET_DISCONNECT_REQ
MSG_ID=4
NET_DISCONNECT_REQ:
FM_NODE_ID = 1
TO_NODE_ID = 1
EQUIP_ID
CELL_ID = 2
MSG_ID = 4
ACCESS_CODE = 1
CALL_DUR
RATE
START_SLOT

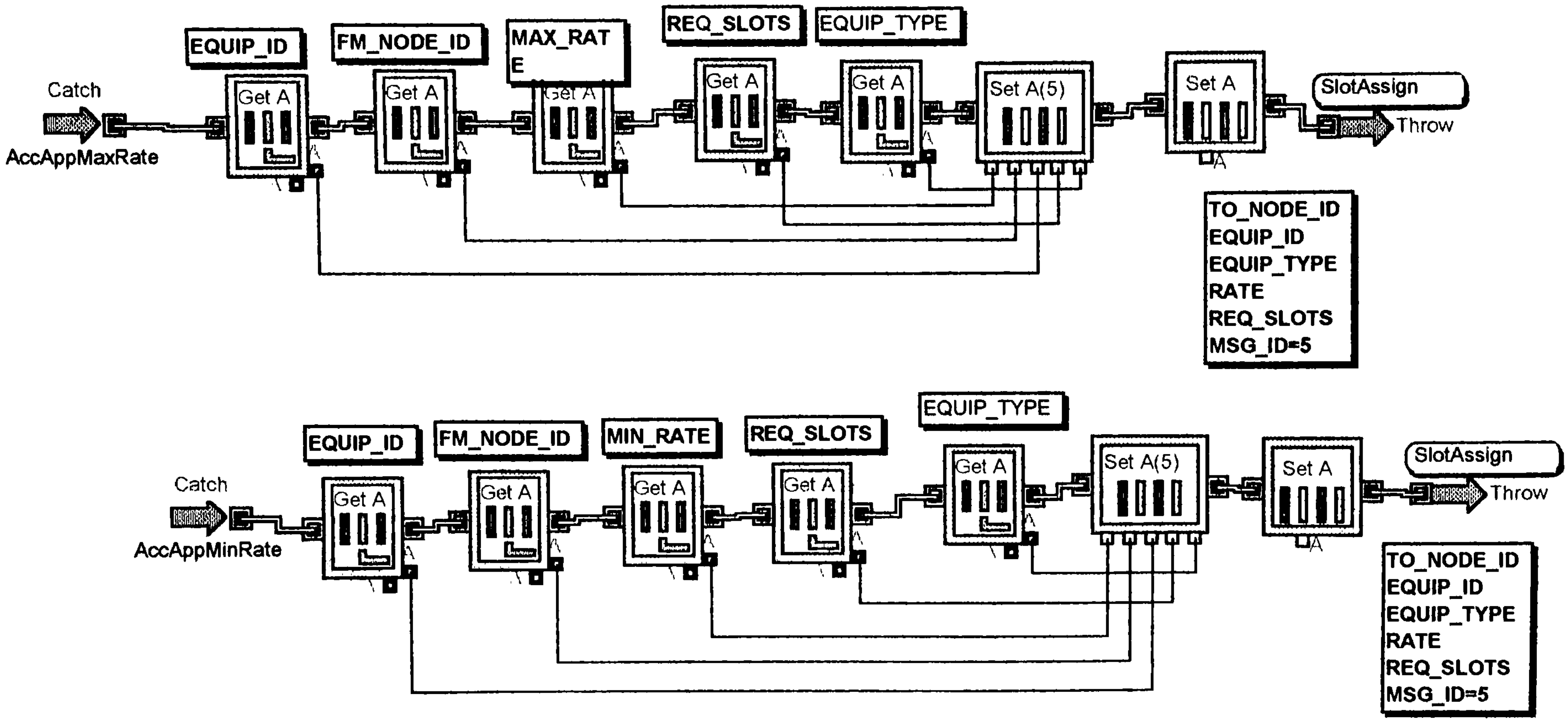
Block Diagram of Bandwidth Support Module



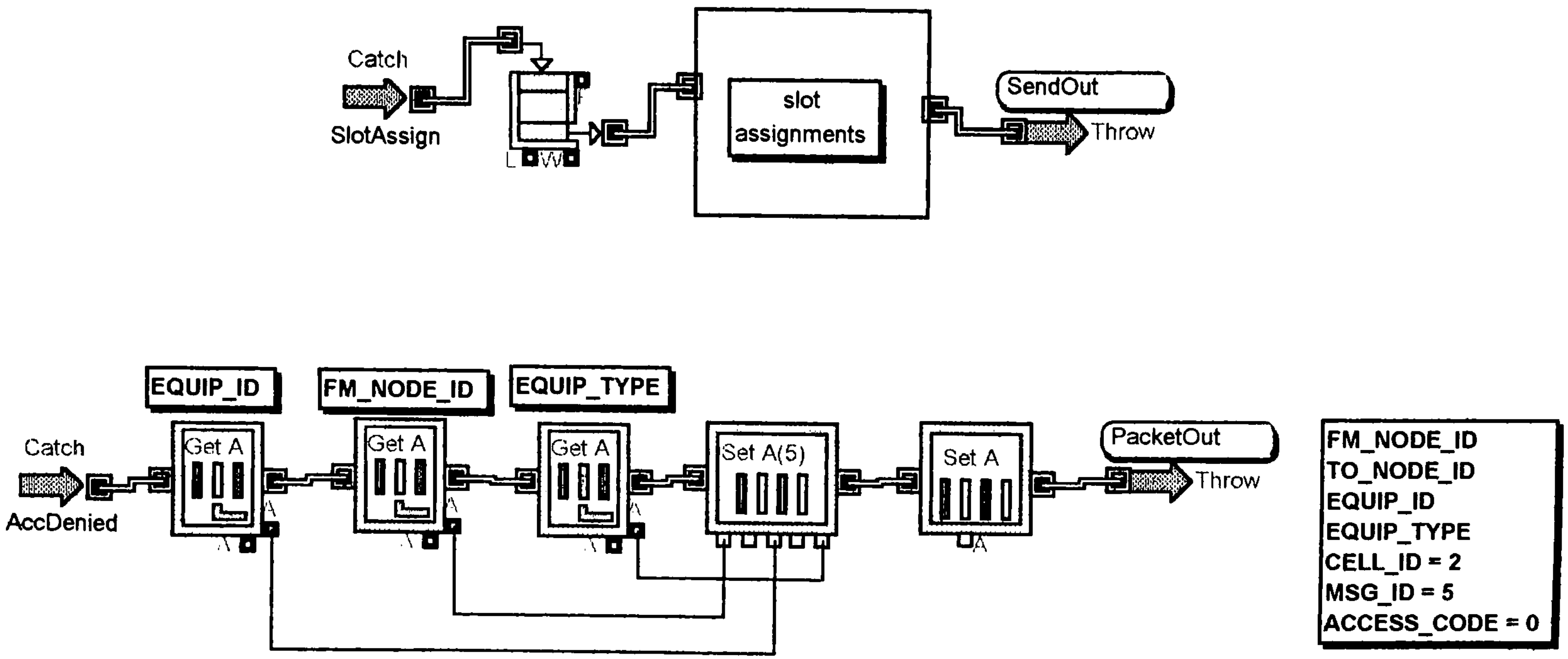
(Bandwidth Support Module cont.)



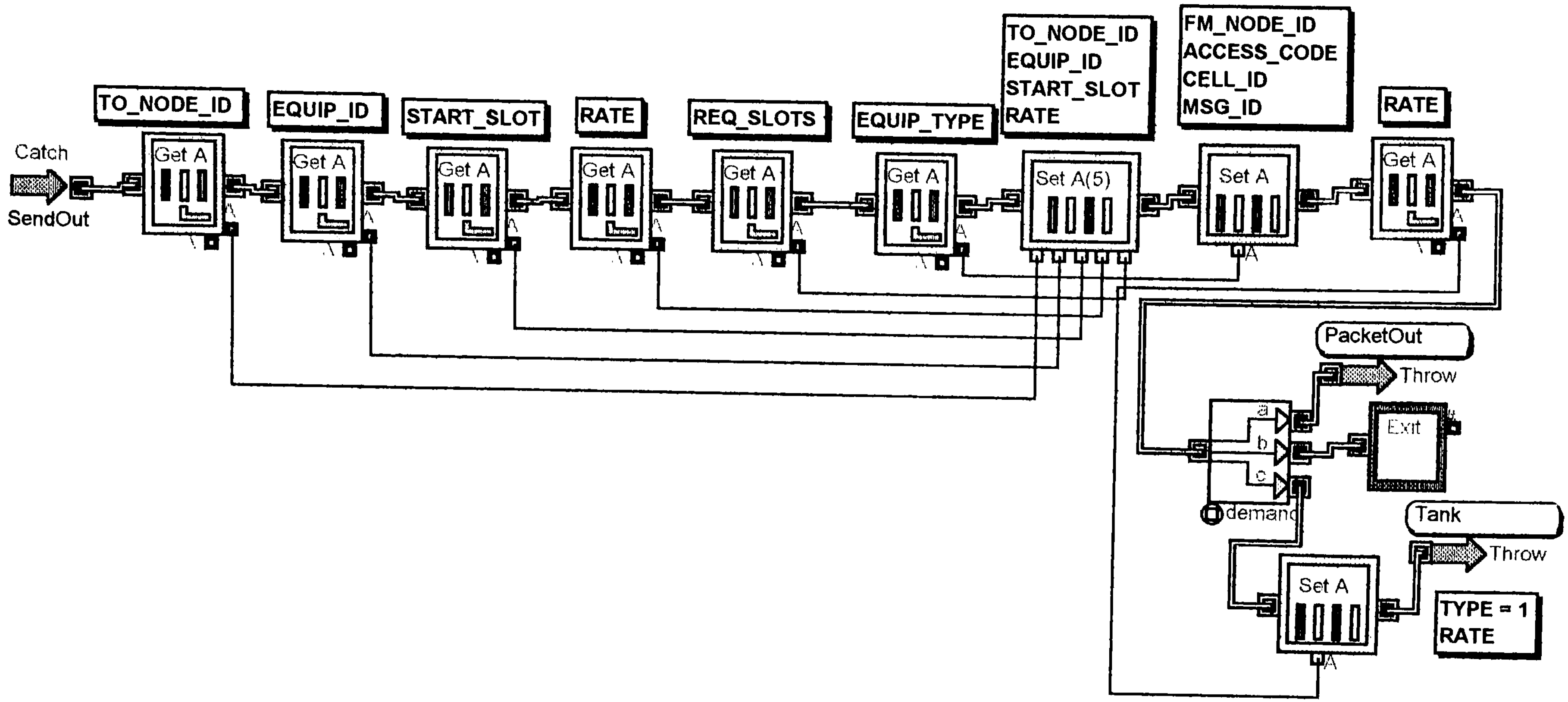
(Bandwidth Support Module cont.)



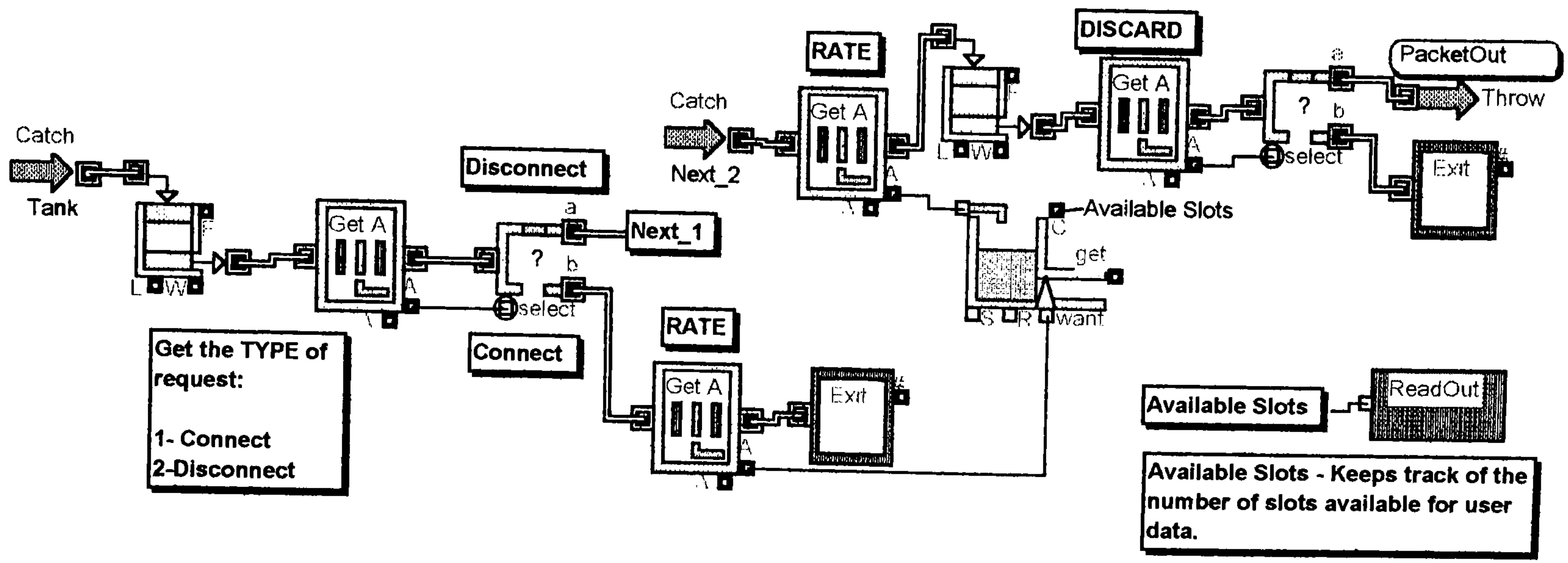
(Bandwidth Support Module cont.)



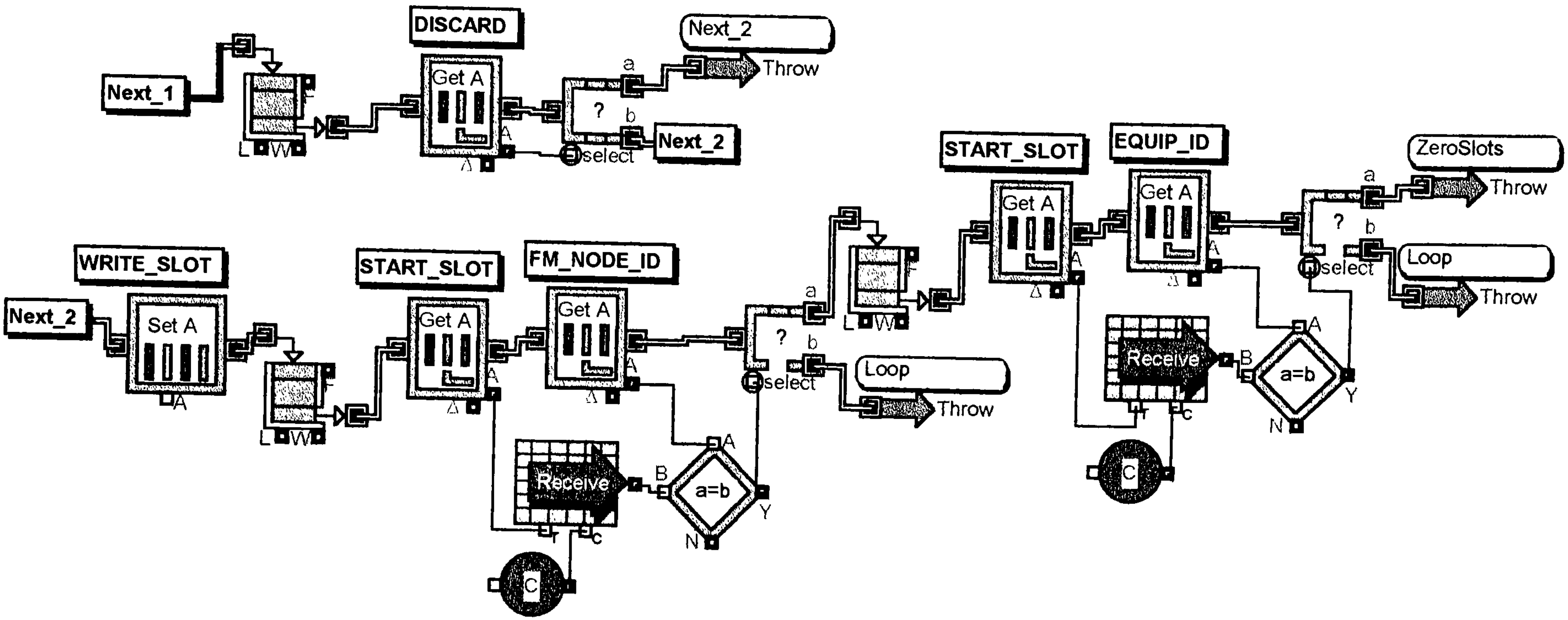
(Bandwidth Support Module cont.)



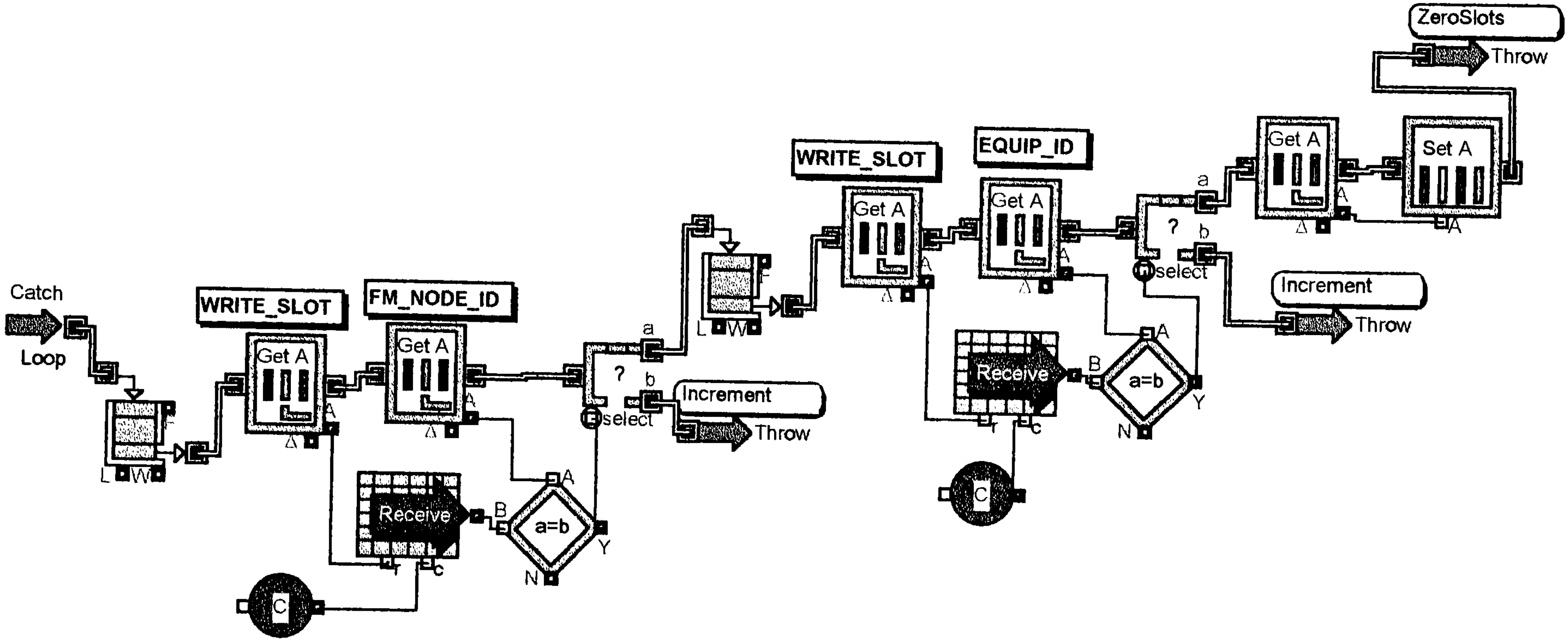
(Bandwidth Support Module cont.)

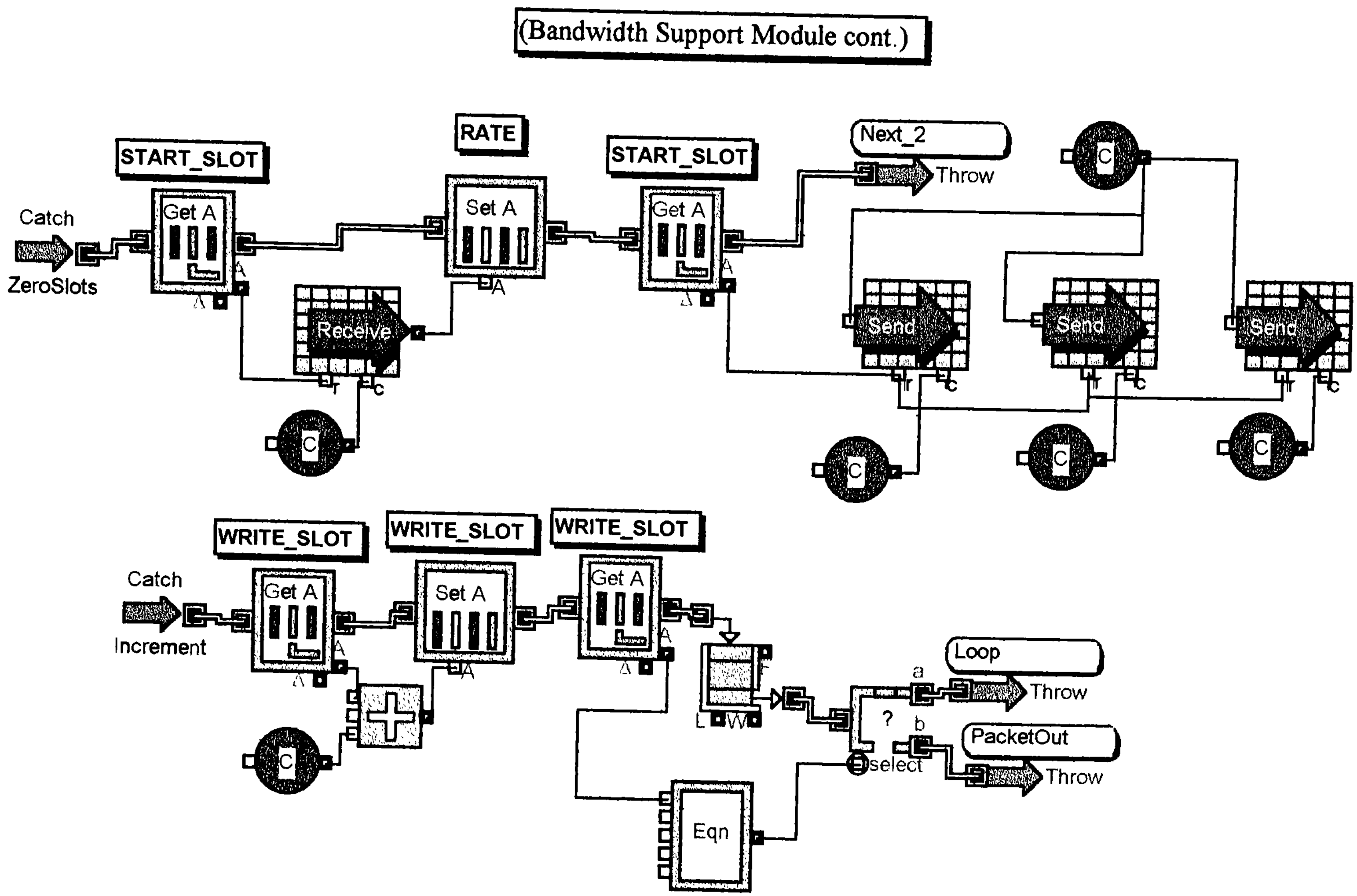


(Bandwidth Support Module cont.)

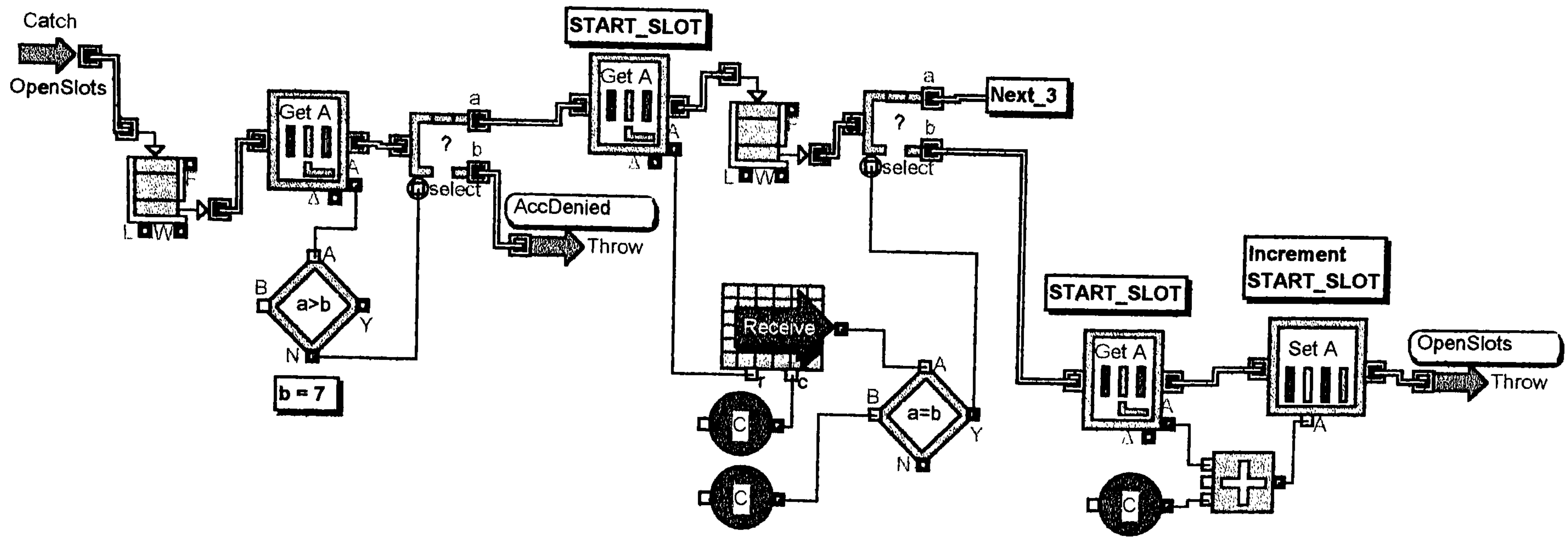


(Bandwidth Support Module cont.)

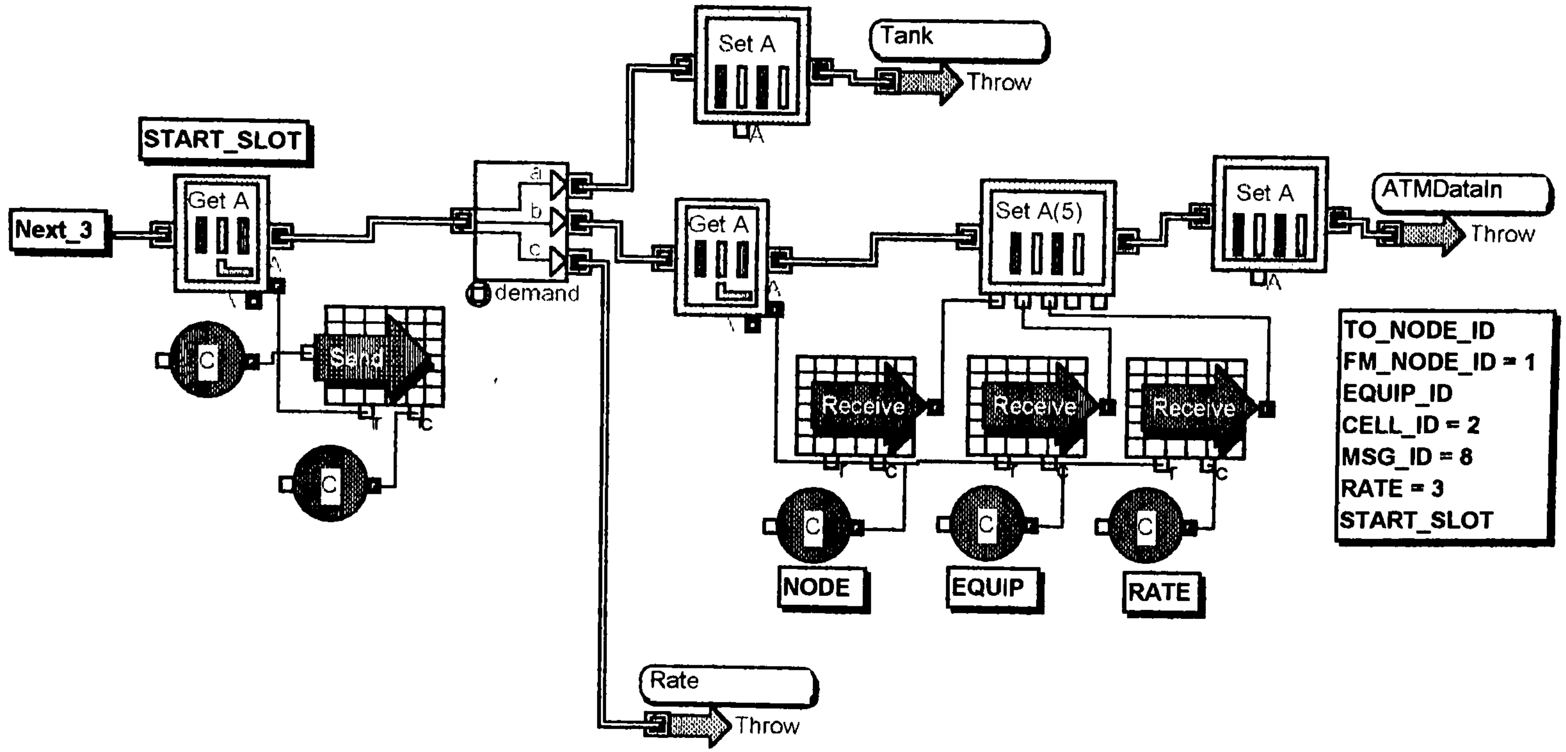



(Bandwidth Support Module cont.)
Catch
ZeroSlots
START_SLOT
Get A
RATE
Set A
Receive
START_SLOT
Get A
Next_2
Throw
Send
Send
Send
C
WRITE_SLOT
Get A
Catch
Increment
WRITE_SLOT
Set A
WRITE_SLOT
Get A
Eqn
Loop
Throw
PacketOut
Throw
select

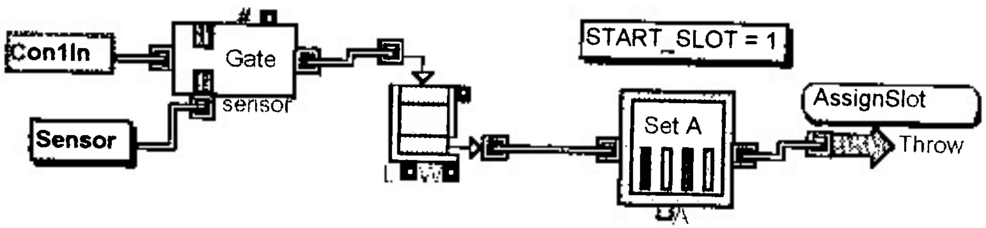
(Bandwidth Support Module cont.)



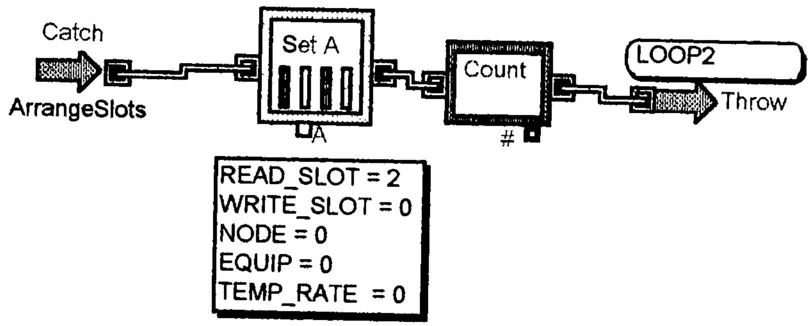
(Bandwidth Support Module cont.)

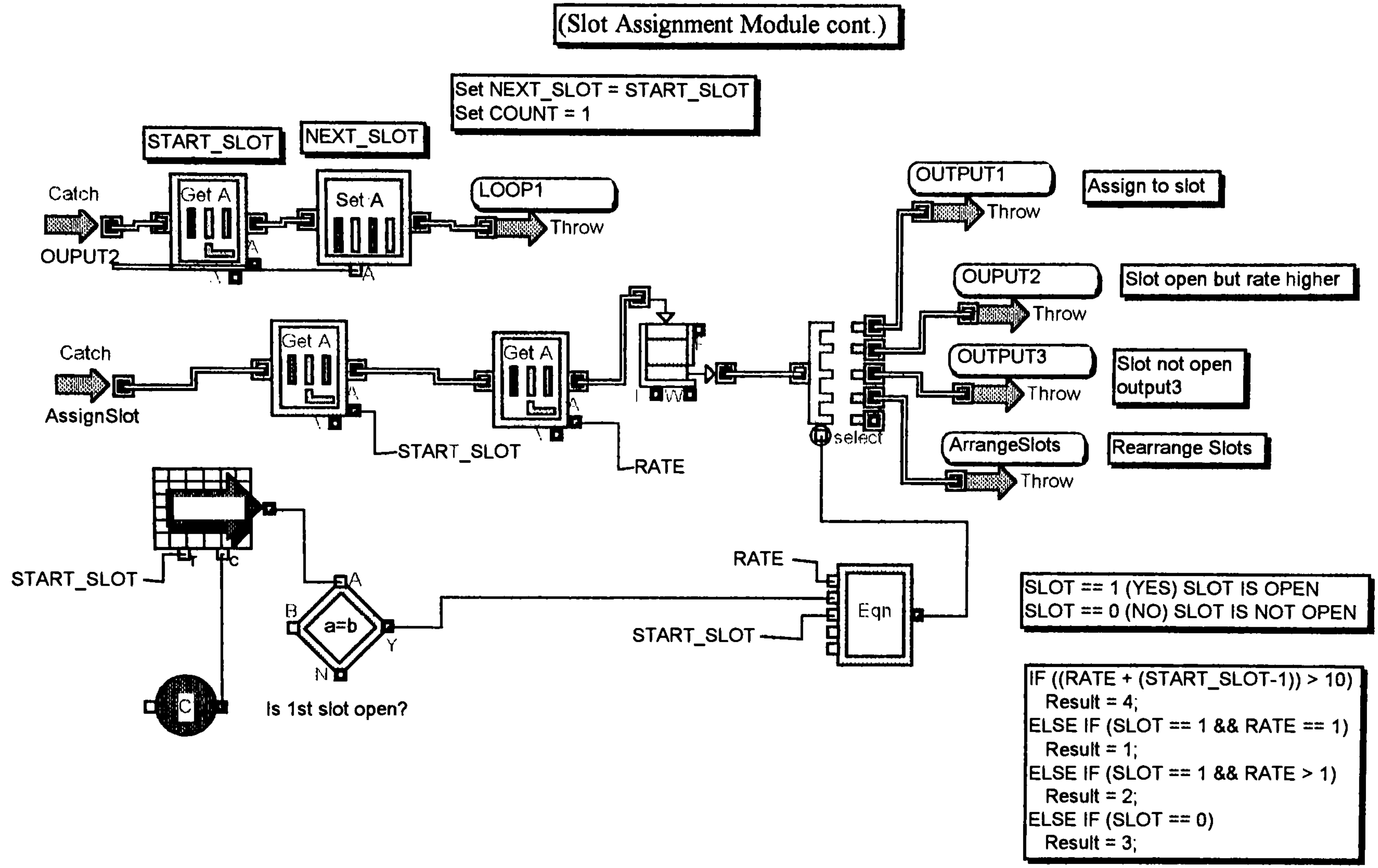


Block Diagram of Slot Assignment Module



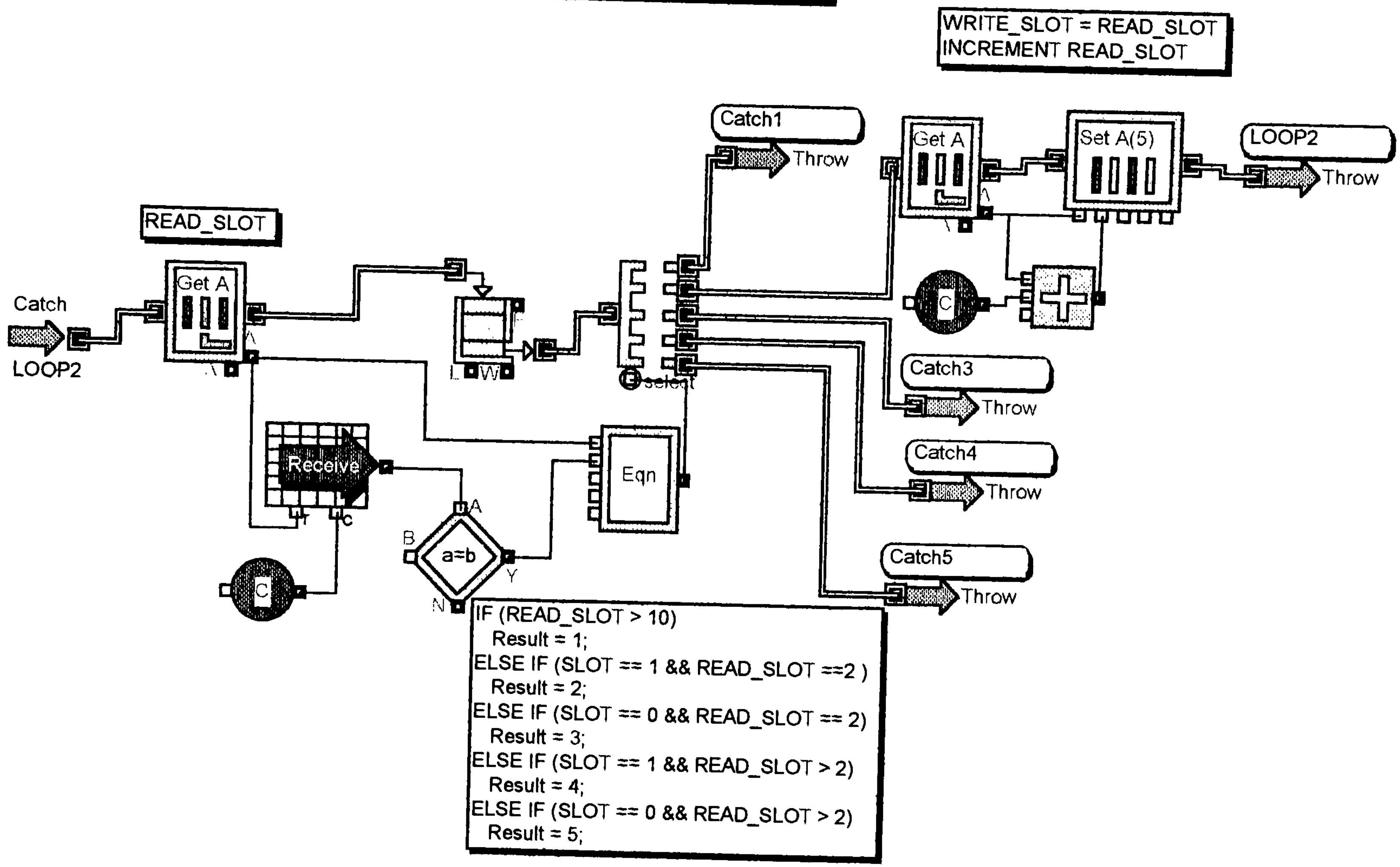
(Slot Assignment Module cont.)

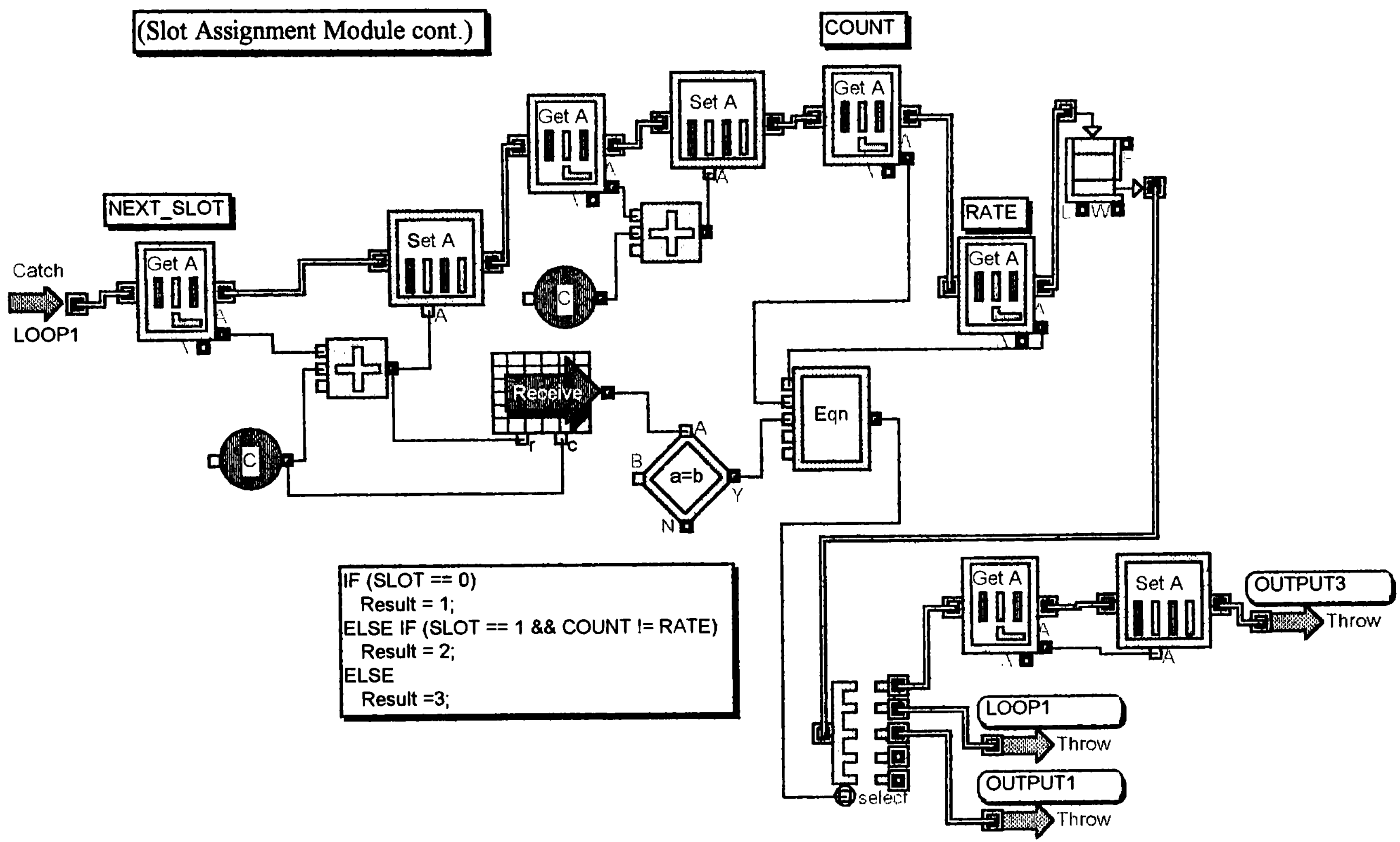




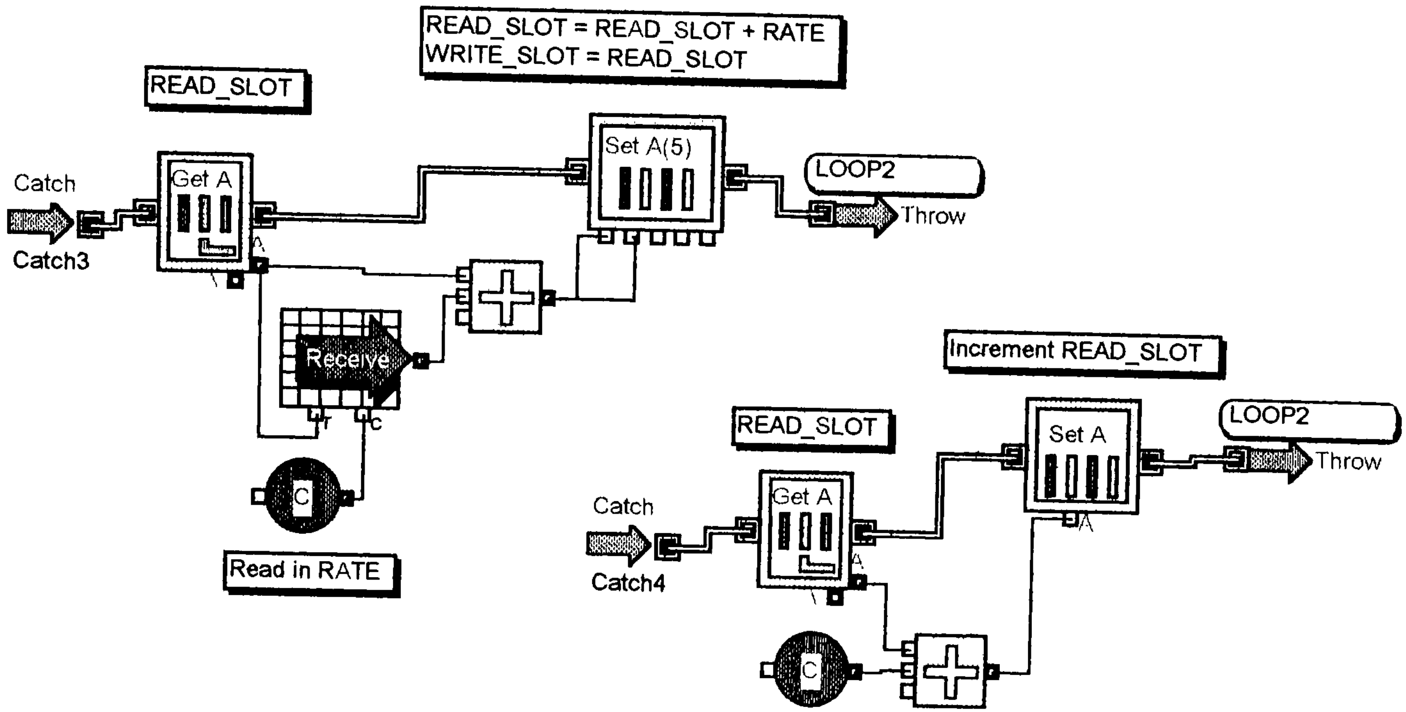
(Slot Assignment Module cont.)
Set NEXT_SLOT = START_SLOT
Set COUNT = 1
START_SLOT
NEXT_SLOT
Catch
OUPUT2
Get A
Set A
A
LOOP1
Throw
Catch
AssignSlot
Get A
START_SLOT
Get A
RATE
select
OUTPUT1
Throw
Assign to slot
OUPUT2
Throw
Slot open but rate higher
OUTPUT3
Throw
Slot not open
output3
ArrangeSlots
Throw
Rearrange Slots
START_SLOT
a=b
A
B
Y
N
Is 1st slot open?
C
RATE
Eqn
START_SLOT
SLOT == 1 (YES) SLOT IS OPEN
SLOT == 0 (NO) SLOT IS NOT OPEN
IF ((RATE + (START_SLOT-1)) > 10)
Result = 4;
ELSE IF (SLOT == 1 && RATE == 1)
Result = 1;
ELSE IF (SLOT == 1 && RATE > 1)
Result = 2;
ELSE IF (SLOT == 0)
Result = 3;

(Slot Assignment Module cont.)

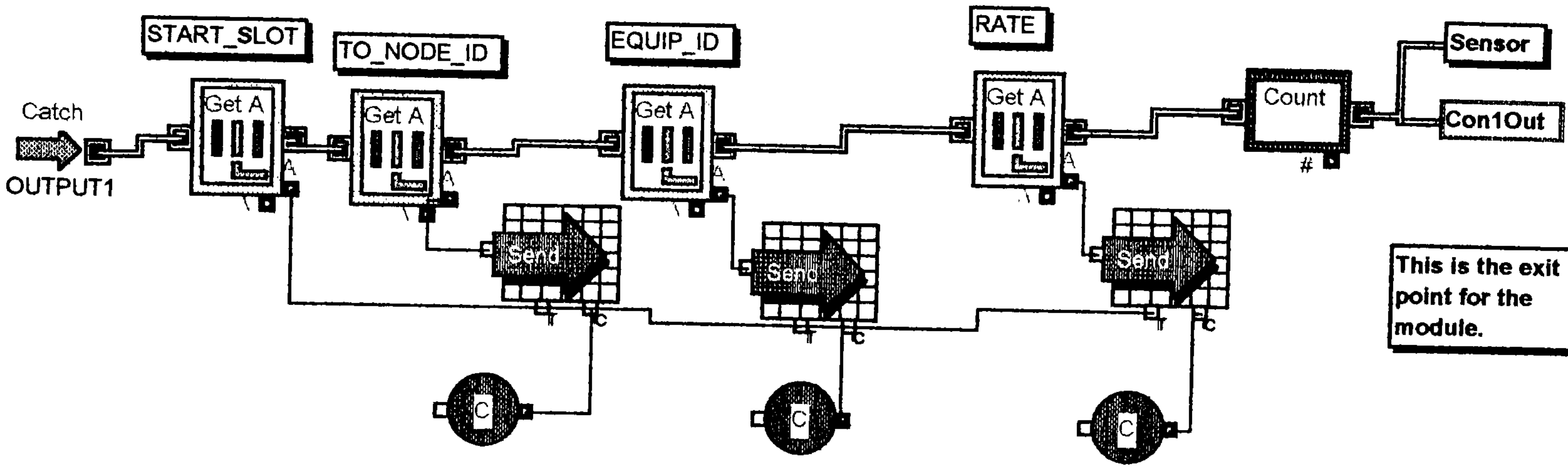



(Slot Assignment Module cont.)
Catch
LOOP1
NEXT_SLOT
Get A
Set A
Get A
Set A
COUNT
Get A
RATE
Get A
C
C
Receive
r
c
A
B
a=b
Y
N
Eqn
IF (SLOT == 0)
Result = 1;
ELSE IF (SLOT == 1 && COUNT != RATE)
Result = 2;
ELSE
Result =3;
Get A
Set A
OUTPUT3
Throw
LOOP1
Throw
OUTPUT1
Throw
select

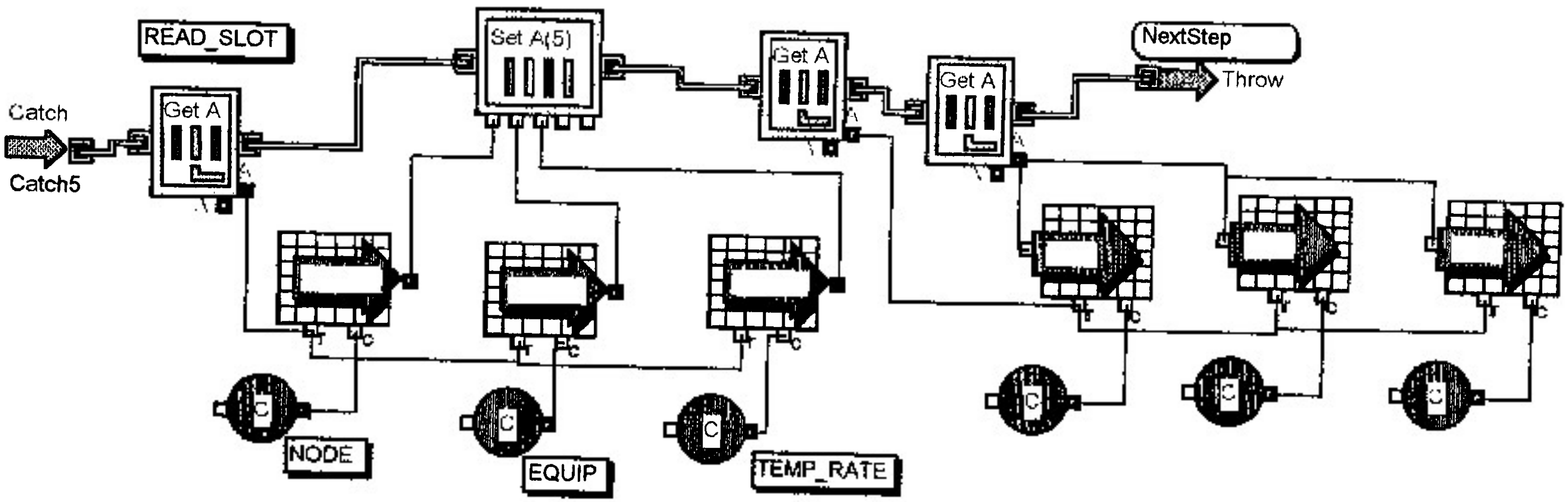
(Slot Assignment Module cont.)



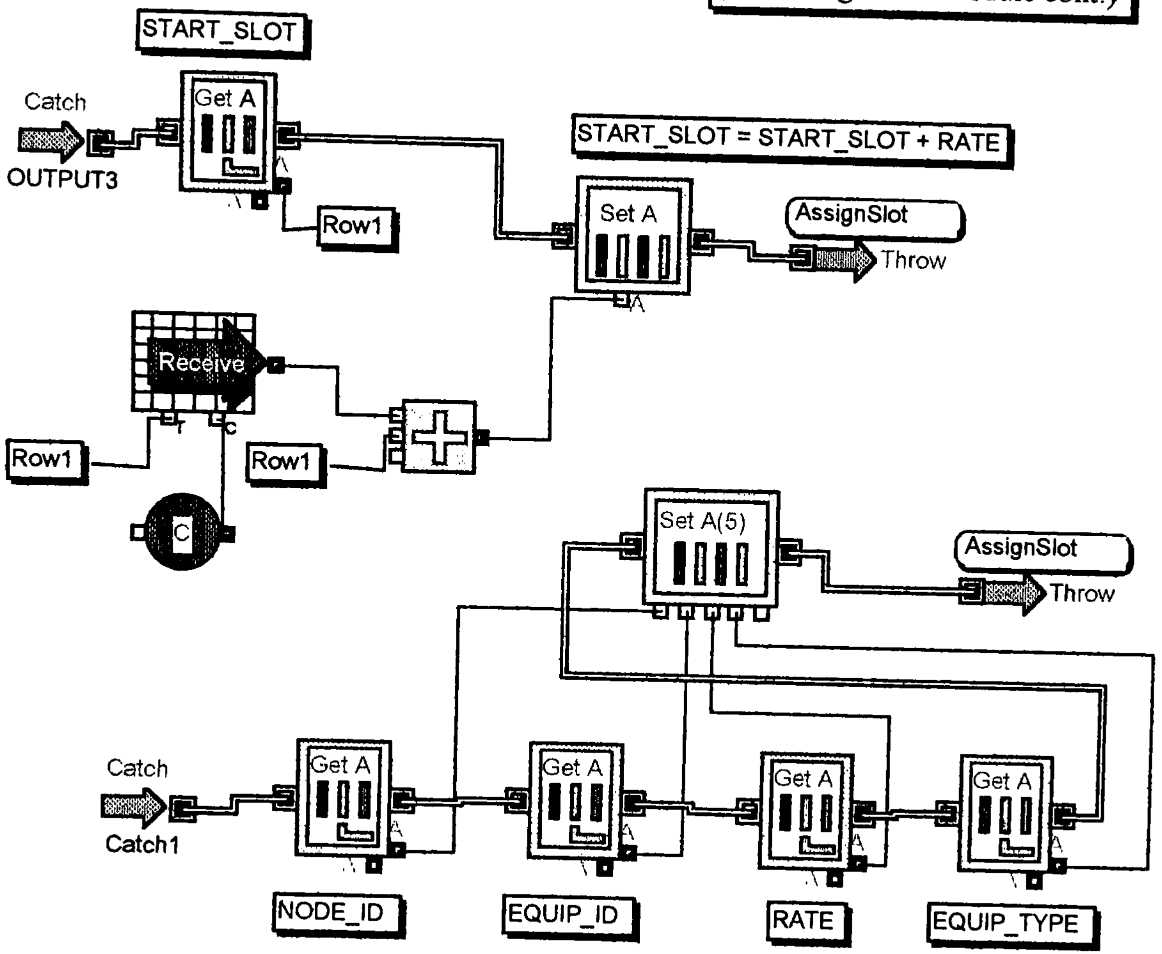
(Slot Assignment Module cont.)



(Slot Assignment Module cont.)



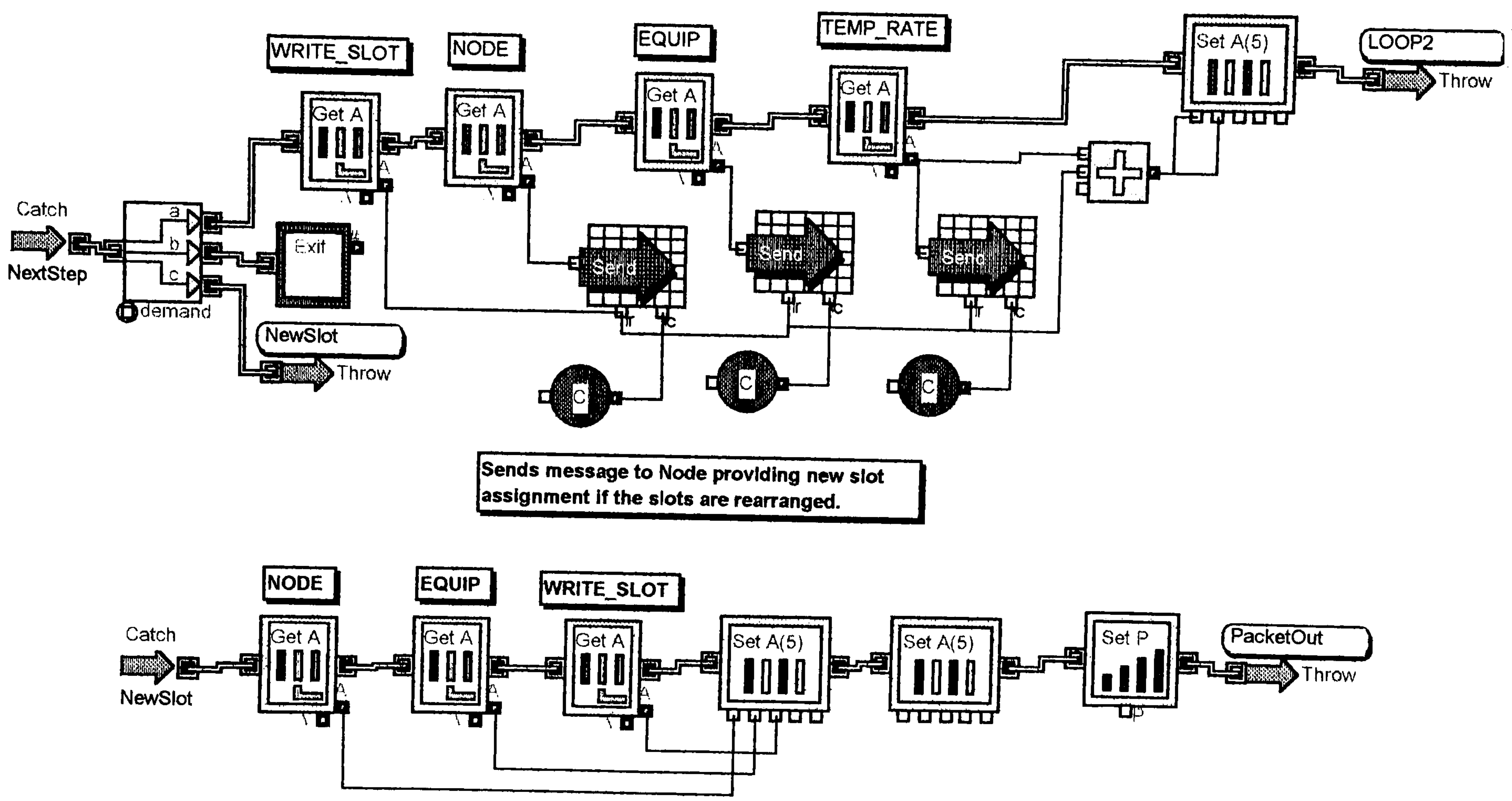
(Slot Assignment Module cont.)



Reset the packet back to the orignal:

TO_NODE_ID
EQUIP_ID
EQUIP_TYPE
START_SLOT
RATE

(Slot Assignment Module cont.)



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Glossary

NET_ACT State of the network when media has been established and there is activity

NET_IDLE - State of the network when there is no media or activity

NODE_ACT State of a node when it is active on the network

NODE_IDLE - State of a node when the net is NET_IDLE

NODE_INACT State of a node when the net is NET_ACT but the node itself has no active connection.

NODE_STARTUP - The State a node goes into when starting a network. See STARTUP.

ATM/HF - Network architectural design for operating ATM over HF radio.

RDRN A mobile ATM network using UHF frequencies and beam antennas.

ATM ATM is a high speed transport and switching system that allows the integration of voice and data applications including multimedia.

R-ALOHA Reservation ALOHA is based upon the S-ALOHA MAC protocol. The difference is that the station that successfully gains access to the slot maintains control of the slot for the duration of the transmission of data. Once transmission stops, the slot becomes available for contention.

S-ALOHA- Slotted ALOHA is based upon the ALOHA CSMA/CD MAC protocol. The difference is that transmission can only begin at the start of a slot.

PRMA - PRMA is based upon R-ALOHA. In PRMA a station can request permanent slot assignment for a fixed number of slots for the duration of the call.

DPRMA - DPRMA is based upon PRMA. In DPRMA a station provides information on minimum, maximum, and current data rates so that bandwidth can be dynamically adjusted. This protocol is proposed for use with new VBR applications.

PRMA/DA PRMA/DA is based upon PRMA. In PRMA/DA there are available slots used for access contention. The number of available slots is dynamically adjusted based upon the number of access attempts. The purpose is to reduce access delay and improve throughput.

CBR CBR applications require a constant bandwidth.

ABR ABR applications do not require constant bandwidth and are usually asynchronous in operation. They are assigned available bandwidth.

UBR - UBR applications do not specify a bit rate and are asynchronous in operation.

VBR- VBR applications require dynamic bandwidth allocation for the duration of the call. The bandwidth dynamically adjusts to accommodate the changing bit rate.

AAL - An AAL is an interface between an application and the ATM network.

STARTUP - The procedure a node follows for establishing a network. STARTUP is followed is the network is NET_IDLE.

Dynamic Bandwidth Allocation - A dynamic capability which allows the controlling node to vary the bandwidth of a connection upon request of the user application. Also allows the controlling node to change the bandwidth of a connection to accommodate higher priority requests.

CSMA/CA The IEEE 802.11 recommended protocol for network access with collision avoidance. Uses DIFS to avoid collisions.

Distributed InterFrame Space - The period of time a node senses the carrier before transmitting. In ATM/HF this period is a random multiple of the propagation delay.

Short InterFrame Space - The period of time within the DIFS in which other nodes can broadcast a collision message notifying other nodes that their access attempt has failed. Based upon the premise that transmitting nodes can not tell if a collision has occurred.

Dynamic Call Reservation A MAC protocol where access messages are embedded in the headers of existing connections.

Dynamic Available Slots - In ATM/HF these are the frame slots available for network access from the NODE_IDLE State.

Call Blocking - When a call is denied access to the network. The percent of calls blocked is used as a measure of the efficiency of the network and MAC protocol.

Throughput - A measure of the efficiency of a network. In ATM/HF it is the number of slots used compared to the number of slots available for use.

Controlling Node The node which maintains network timing via the sync signal and which controls network access, and rate and slot assignments.

Acronyms

AAL - ATM Adaptation Layer
ABR Available Bit Rate
ALE - Automatic Link Establishment
ATM Asynchronous Transfer Mode
ATM/HF ATM over HF Radio
BER Bit Error Rate
CBR - Constant Bit Rate
CLR - Cell Loss Ratio
CMISE - Common Management Information Service Element
CORBA - Common Object Request Broker Architecture
COTS - Commercial Off The Shelf
CRC - Cyclic Redundancy Check
CS Convergence Sublayer
CSMA/CA - CSMA with Collision Avoidance
CSMA/CD - Carrier Sense Multiple Access with Collision Detection
DAS -Dynamic Available Slot
DBA Dynamic Bandwidth Allocation
DCR Dynamic Connection Request
DIFS Distributed InterFrame Space
DRES Dynamic Reservation
FEC Forward Error Correction
FIFO - First In First Out
GT Guard Time
GUI - Graphical User Interface
HF High Frequency
HiFIN High Frequency IntraNet
LAN Local Area Network
LOS - Line of Sight
MAC Media Access Control
NIC Network Interface Card
OOP - Object Oriented Programming
ORB - Object Request Broker
OSI Open Systems Interconnect
OTH - Over the Horizon
PDU Protocol Data Unit
PODA Priority Oriented Demand Access
PRMA Packet Reservation Multiple Access
PRMA/DA PRMA with Dynamic Allocation
PSTN Public Service Telephone Network
QoS - Quality of Service
R-ALOHA Reservation ALOHA
RDRN Rapidly Deployable Radio Network
S-ALOHA Slotted ALOHA
SAR - Segmentation and Reassembly

SATCOM Satellite Communications
SIFS - Short InterFrame Space
SSB - Single Side Band
SYNC - Sync signal
TINA Telecommunications Information Networking Architecture
UBR Unspecified Bit Rate
UHF Ultra High Frequency
UNI User to Network Interface
VBR Variable Bit Rate
VHF Very High Frequency
ViVID Video, Voice, Image, Data
WATM Wireless ATM