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# INTEGRATED VOICE/DATA PROTOCOLS FOR SATELLITE CHANNELS

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## ABSTRACT

In this paper, several integrated voice/data protocols for satellite channels are studied. The system consists of two types of traffic; voice calls which are blocked–calls–cleared and the data packets which may be stored when no channel is available. The voice calls are operated under a demand assignment protocol. We introduce three different data protocols for data packets. Under RAD, the ALOHA random access scheme is used. Due to the nature of random access, the channel utilization is low. Under DAD, a demand assignment protocol is used to improve channel utilization. Since a satellite channel has long propagation delay, DAD may perform worse than RAD. We combine the two protocols to obtain a new protocol called HD. The proposed protocols are fully distributed and no central controller is required. Numerical results show that HD enjoys a lower delay than DAD and provides a much higher channel capacity the RAD. We also compare the effect of fixed and movable boundaries in partitioning the total frequency band to voice and data users.

## 1 INTRODUCTION

In this paper, we propose distributed multiple access protocols for satellite channels. In [3,4], integrated voice and data multiple access protocols are studied. In [3] a central controller implements reservation–based protocols for voice calls and data packets in a land-mobile satellite network. Coviello and Vena [2] described the Slotted Envelope Network scheme for multiplexing voice and data traffic on a communication link using TDMA such that the number of slots reserved for voice calls in each TDMA frame varies from frame to frame. Similar concepts, called movable boundary schemes, are studied in [5,6].

For data service in satellite channels, the well-known ALOHA scheme is simple but suffers from low channel capacity. Demand assignment protocols, on the other hand, provide higher channel capacity but require channel access control. In addition, the delay of demand assignment scheme is larger than the ALOHA scheme when the offered traffic is light.

In this paper, a multiple–channel satellite providing both voice and data service is considered. For data packets, three different protocols are proposed. Under RAD, the ALOHA random access scheme is used. Under DAD, a demand assignment protocol is used to improve channel utilization. Since a satellite channel has long propagation delay, DAD may perform worse than RAD. We combine the two protocols to obtain a new protocol called HD. In fact, HD behaves like RAD when the offered traffic is low and like DAD when the traffic is high.

We also study a movable boundary scheme in which the data packets can use the idle voice channels. Note that all of the protocols proposed are fully distributed, that is, no central controller is necessary.

## 2 PROTOCOL DESCRIPTION

There are two types of traffic: voice calls and data packets. Voice calls will be blocked–calls–cleared, while data packets which arrive when all channels are busy may be stored. The system consists of a large number of voice and/or data users and a satellite which serves as a relay. The total bandwidth of the satellite is divided into multiple channels. There are R reservation channels and M message channels. The word "message" here refers to either voice call or data packet. All channels are slotted and the length of a slot time is equal to the transmission time of a data packet. A time slot in a reservation channel is further divided into $n$ minislots and the length of each minislot is equal to the transmission of a request packet. Thus there are a total of $N = nR$ minislots per slot time in which to make reservations. Among the $N$ minislots, the first $M_v$ of them, called voice request minislots, are for voice requests and the others, called data request minislots, are for data requests. Here $M_v$ is the maximum number of voice calls allowed in the system at the same time. $M_v$ is a system design parameter.

### 2.1 Voice Protocol

All potential voice users monitor the reservation channels at all times and voice calls are operated on a demand assigned basis. When a voice user attempts to communicate with another, he will first send a request on a voice request channel and then begin his voice call on a message channel if the request is successful. More specifically, suppose user $i$ has a voice call to user $j$. First user i checks all of the voice request minislots. If all of them are busy, then no channel is available at this time and the voice call will be blocked; otherwise, user $j$ picks at random one of the idle voice request minislots, say the $k$th voice request minislot, within the next slot and sends a request packet. The request packet cintains the I.D. of the destination node. It also sends a jamming signal on the $k$th voice request minislot of the S following slots, where S is the round-trip propagation of the satellite channel measured in slots. This ensures that both node $i$ and node $j$ will learn whether the request signal is successful or not. If this request packet is successful, that is, the packet does not suffer a collision, node $j$ will tune his receiver to the $k$th message channel and node $i$ will start his transmission on this channel when he hears his request packet. If the request packet is not successful, node $i$ will wait a random time and try again. Note that while node $i$ is transmitting, he will keep on sending jamming signal in the corresponding voice request minislot to prevent potential interference from other users.

## 2.2 Data Protocols

Three different protocols are developed for data transmissions. Suppose that there are $m$ message channels available for data.

### 2.2.1 Random Access Data (RAD) Protocol

The data packets are transmitted under a multi-channel slotted-ALOHA scheme and no reservation channel is required. If node $i$ wants to send a packet to node $j$, then node $i$ will send the data packet on the $j_m$th, $j_m = j \bmod m$, channel. In other words, multiple nodes may be receiving packets on the same channel, although not at the same time. Since there may be many channels available, it is not practical for a receiver to listen to all channels and wait for incoming packets. In this protocol, node j will only listen to the $j_m$th channel.

### 2.2.2 Demand Assignment Data (DAD) Protocol

The maximum throughput of m-channel slotted-ALOHA is $0.368m$. Here we propose a demand assignment protocol to improve it. Suppose node $i$ wants to send a packet to node $j$. Node $i$ will send a request packet on the reservation channels. In particular, node $i$ chooses one of the reservation minislots at random, and sends a request packet containing the destination I.D. If the request is successful, node $i$ puts this request on his request queue and waits for the scheduled time to transmit the packet. In this protocol, all users continuously monitor the reservation channels and track the successful reservation requests in the system. Upon hear a successful request, it puts the request in its request queue. Due to the broadcast nature of satellite channels, all nodes will receive the same request information and, thus, the request queue for every user is identical. According to the request queue, all packets with successful request are transmitted on a first come first served basis. At the beginning of a time slot, the user who has a request queued at the $k$th, $k \leq m$ position of the request queue will transmit the packet on the $k$th data channel. Up to $m$ packets can be transmitted in a slot time and the corresponding requests of the transmitted packets in the request queue will be deleted at the end of the slot. Since the receiver (node j) also has the transmission schedule, he can tune his receiver to the proper channel to receive the packet from node $i$. Note that a user needs only keep information on the length of the queue, the position of his own request, and the requests in which the corresponding data packets are destined to him. Thus we have a fully distributed scheduling scheme which does not require much bookkeeping.

### 2.2.3 Hybrid Random Access and Demand Assignment Access Data (HD) Protocol

The delay under DAD is larger than that under RAD when the offered traffic is light, especially on a long propagation delay channel such as satellite channels. However, we can combine these two protocols and get a hybrid protocol which behaves like a random access scheme when the traffic is light and like a reservation scheme when the traffic is heavy.

Again every user keep on monitoring the reservation channels so that everyone knows the system request queue at any time. Suppose node $i$ has a packet to node $j$ at slot $\beta$, it will send a request on the data request minislot. Whether this request is successful or not is unknown until one round–trip delay later, i.e., the scheduled time for this packet is at least S slots from now. Since all nodes have the system schedule information at slot $\beta$, node $i$ knows whether there will be some idle channels at the next slot or not. If there are some, say $k$, idle channels, node $i$ can send the packet, called an R–packet, on the $j_k$th channel at the next slot; otherwise, the operation of the protocol is exactly the same as DAD. Note that node $I$ still needs to make a reservation for this packet, since the R–type packet may result in a collision.

## 2.3 Integrated Voice and Data Protocols

We can mix the voice protocol with any one of the three data protocols mentioned above to provide an integrated voice and data service. The $M$ message channels are divided into two groups. On group, containing $M_v$ channels, is allocated for voice calls; the other group, containing $M_d = M - M_v$ channels, is for data packets.

### 2.3.1 Fixed Boundary Strategy

In this strategy, the data packets are not allowed to use the $M_v$ voice channels even if some of them are idle.

### 2.3.2 Movable Boundary Strategy

The difference between the fixed and movable boundary strategies is that in the latter the data packets may occupy any of the voice channels not currently in use. An arriving voice call, however, has higher priority to receive service in the voice channels. Since the voice calls have higher priority, the operation of the voice users is the same as the voice protocol mentioned above. However, the operation of the data user needs some minor modifications. In this strategy, a data user must check the status of the voice request minislots in every slot. Since an active voice call will keep on sending jamming signal on the corresponding voice request minislot, the data user can find out how many and which voice channels are idle by checking the voice request minislot. If there are $v$ voice channels free, then the data users will operate as if there are $M_d + v$ data channels at the next slot.

## 3 PERFORMANCE ANALYSIS

We now analyse the performance of the multipleaccess schemes proposed in the previous section.

### 3.1 Model Assumptions

The following assumptions will be made in the analysis:

1. The users collectively generate Poisson data packet traffic at rate $\lambda_d$ packets/second and Poisson voice traffic at rate $\lambda_v$ calls/second.
2. The data packets are of fixed length. The voice call duration is exponentially distributed with mean $1/\mu_v$ seconds.
3. Channels are slotted. Let T denote the slot length which equals the transmission time of a packet and S be the round–trip delay of the channel measured in slots.
4. The retransmission delay for a request or a random access data packet is uniformly distributed between 0 and $K$ slots.

### 3.2 Voice Blocking Probability

The average voice duration $1/\mu_v$ is expected to be much longer than the slot time $T$ and for a reasonable blocking probability, $\lambda_v T$ should be much less than one. The probability of collision of voice requests is negligible. In addition, since the channel is slotted, a voice call of length $X$ sec. will occupy $\lceil \frac{X}{T} \rceil$ slots $= \lceil \frac{X}{T} \rceil T$ sec. We shall approximate this by $X$ sec. The error introduced is negligible. The voice channels can be modeled as as an M/G/s/s s-server loss system. The blocking probability for an s-channel system is given by the Erlang B formula,

$$P_B = \frac{(\frac{\lambda_v}{t_v})^s/s!}{\sum_{k=0}^{s}(\frac{\lambda_v}{t_v})^k/k!} \quad (1)$$

where $1/t_v$ is mean call duration time $1/\mu_v$ plus the round–trip delay time $ST$. Furthermore, for typical voice calls, the call duration is much larger than the round–trip delay. We can futher simplify the system to an M/M/s/s queue and the probability that a system with s channels has $n$ active voice calls is given by

$$\pi_v(n) = \frac{(\frac{\lambda_v}{\mu_v})^n/n!}{\sum_{k=0}^{s}(\frac{\lambda_v}{\mu_v})^k/k!} \tag{2}$$

### 3.3 Data Channel Analysis

#### 3.3.1 Random Access Data (RAD) Protocol

Define $p_{suc\_data}$ to be the probability that a data packet will be successful on a data channel RAD. Clearly $p_{suc\_data} = \lambda_d T/m \times e^{-\lambda_d T/m}$ for an $m$ data channel system. The throughput of such a system $\eta_{RAD}$ is $p_{suc\_data} m = \lambda_d T e^{-\lambda_d T/m}$. The average delay under RAD can be obtained from

$$D_{RAD}(m) = (1.5 + S + (e^{\lambda_d T/m} - 1)(1 + S + \frac{K-1}{2}))T \tag{3}$$

where $e^{\lambda_d T/m} - 1$ is the average number of retransmissions required for the data packet.

#### 3.3.2 Demand Assignment Data (DAD) Protocol

The analysis of the reservation channels, which operates under random access, is similar to that of the data channel under RAD. Define $p_{suc\_req}$ to be the probability that a request packet will succeed in a reservation minislot. Then, $p_{suc\_req} = \lambda_d T/l \times e^{-\lambda_d T/l}$ and the throughput of the reservation channel $\eta_{rs}$ is $l \times p_{suc\_req}$ per slot, where $l$ is the number of data request minislots per slot. The delay of a successful request is $t_{rs} = (1 + S + (e^{\lambda_d T/l} - 1)(1 + S + \frac{K-1}{2}))T$. The maximum throughput of the reservation channel is $0.368l$ requests/slot. If there are $m$ data channels, then $l = 3m$ is enough to achieve the maximum utilization of the data channels.

In the analysis of the data channels, we use $p_i$ to denote the probability that $i$ requests succeed in a slot and define $\pi_i$ to be the probability of having a total of $i$ successful requests enqueued in the system at the beginning of a slot, then

$$p_i = \binom{l}{i} p_{suc\_req}^i (1 - p_{suc\_req})^{l-i} \tag{4}$$

and from [1] we get

$$\Pi(z) = \frac{P(z)[\Pi_m(z) z^m - \Pi_m(z)]}{z^m - P(z)} \tag{5}$$

where $\Pi(z) = \sum_{i=0}^{\infty} \pi_i z^i$, $\Pi_m(z) = \sum_{i=0}^{m-1} \pi_i z^i$ and $P(z) = \sum_{i=0}^{\infty} p_i z^i$. Note that $\Pi'(1)$ is the average queue length.

The total packet delay $D_{DAD}$ consists of three parts: the request packet delay $t_{rs}$, the queueing delay $t_q$ and the propagation delay S. The queueing delay, given by Little's formula, is $\Pi'(1)/\eta_{rs}$. Now the total packet delay under DAD is

$$\begin{aligned} D_{DAD} &= t_{rs} + t_q + (S+0.5)T & (6) \\ &= \{1.5 + 2S + \frac{\Pi'(1)}{\lambda_d T e^{-\lambda_d T/l}}(e^{\lambda_d T/l} - 1)(1 + S + \frac{K-1}{2})\}T & (7) \end{aligned}$$

### 3.3.3 Hybrid Data (HD) Protocol

Under HD, the time slots which are not scheduled for packets can be used to transmit packets in a random access manner. The random access type transmission can be used only when the system state, defined as the number of requests enqueued in the system at the beginning of a slot, is less than the total number of data channels. A packet transmitted in the random manner is called an R–packet. Let $\lambda_d^R T$ be the average number of successful R–packets per slot. Then,

$$\lambda_d^R T = \sum_{i=0}^{m-1} \pi_i \lambda_d T e^{-\frac{\lambda_d T}{m-i}} \qquad (8)$$

Note that R-packets will be transmitted again during their normal scheduled time. This is necessary since the packets transmitted under random access may collide. The delay of successful R–packets if successfully transmitted under random access id $D^R = (S + 1.5)T$ and the delay of R–packets if transmitted at the scheduled time is $D^{DR}$ which is greater than $(2S + 1.5)T$. Let us denote the average delay of packets other than R-packets by $D^N$. We get

$$\begin{aligned} D^N &= \frac{D_{DAD} \times \lambda_d - D^{DR} \times \lambda_d^R}{\lambda_d - \lambda_d^R} T & (9) \\ &\leq \frac{D_{DAD} \times \lambda_d - (2S+1.5)\lambda_d^R}{\lambda_d - \lambda_d^R} T & (10) \end{aligned}$$

Now we can get the upper bound of the average delay under HD with m data channels:

$$\begin{aligned} D_{HD}(m) &= \frac{\lambda_d^R}{\lambda_d} D^R + \frac{\lambda_d - \lambda_d^R}{\lambda_d} D^N & (11) \\ &\leq D_{DAD}(m) - ST\frac{\lambda_d^R}{\lambda_d} & (12) \end{aligned}$$

## 3.4 Analysis of Integrated Protocols

Since the voice calls have higher priority in their own channels, the blocking probability of voice call is not affected by data packets and can be obtained from section 3.2 directly. The following analyzes the delay of data packets in the integrated protocols.

### 3.4.1 Fixed Boundary Intergrated Protocols

Under the fixed boundary strategy, the data packets are not allowed to use the voice channel. The transmissions of voice calls and data packets do not affect each other. Thus, the performance analyses are the same as in sections 3.3.1, 3.3.2, and 3.3.3.

### 3.4.2 Movable Boundary Integrated Protocol

The data packets can use the idle voice channels under this strategy. The typical call duration is about 100 seconds and the typical slot time is in the order of $10^{-2}$ seconds or smaller. To simplify the caculations, we can assume the data queues reach their stationary state when $v$, $0 \leq v \leq M_v$ voice calls are active. Then the average packet delay is

$$Delay = \sum_{k=0}^{M_v} \pi_v(k) D_{data}(M - M_v) \tag{13}$$

$D_{data}$ is obtained from either (3), (7) or (12), depending on which data protocol is used.

## 4 NUMERICAL RESULTS

Figure 3 shows the delay–throughput relationship for the three data protocols under the fixed and the movable boundary strategies. We consider a system with $M = 10$, $M_v = M_d = 5$, $S = 20$, and a voice call arrival rate and service rate corresponding to a blocking probability of 0.02 for five voice channels. As expected, the delay of each protocol increases as the throughput increases, approaching infinity at the maximum throughput. Under RAD, the maximum throughput is $0.368 \times M_v$, while the maximum throughput under DAD and HD is $M_v$. For light traffic, the delay under RAD is smaller than that under DAD. The hybrid protocol HD has the best delay characteristics for medium to heavy traffic and is only slightly inferior to RAD at light traffic. We conclude that most of the improvement for HD is due to R–packets. When the traffic is high the scheduled packets dominate the system and the improvement is small. We also find that the movable boundary strategy is superior to the fixed strategy, but the improvement is very small under DAD.

## 5 CONCLUSIONS

We have proposed several distributed protocols for integrated voice/data service in satellite channels. We find that random access performs best under light traffic, while a hybrid random access-demand assignment protocol performs best for medium to heavy traffic. In addition, a movable boundary strategy performs better than a fixed boundary strategy.

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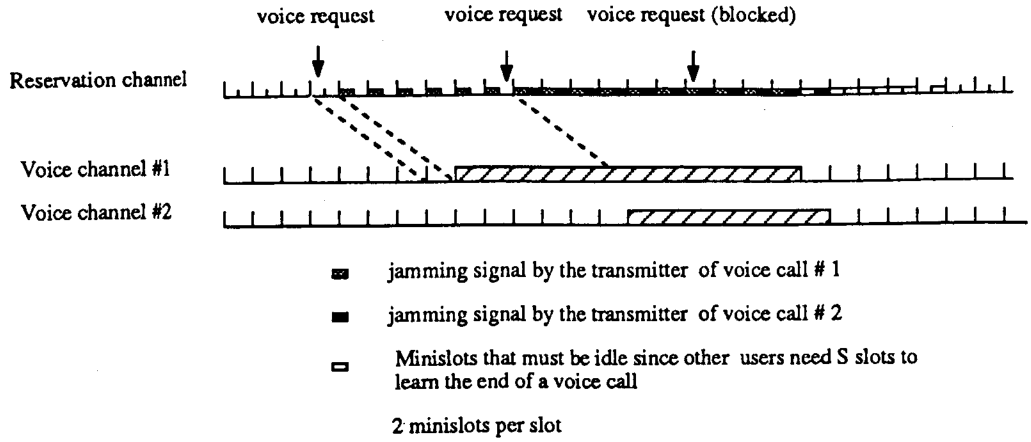


Figure 1. Time diagram for voice calls

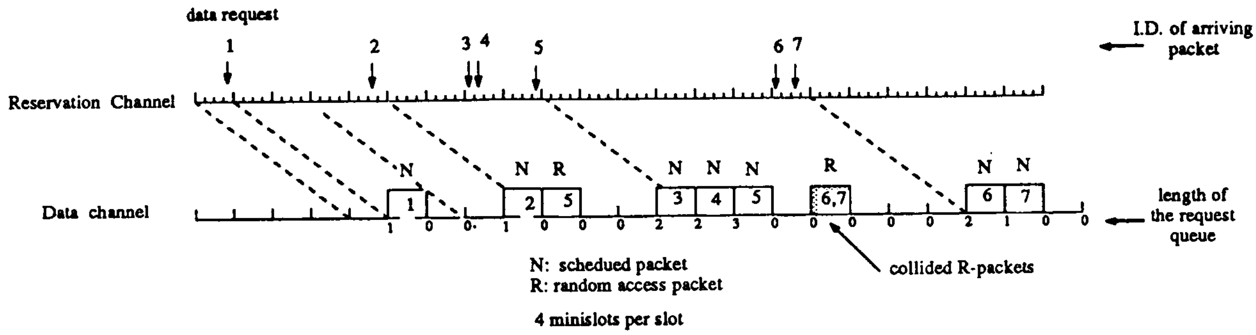


Figure 2. Time diagram for the hybrid data protocol

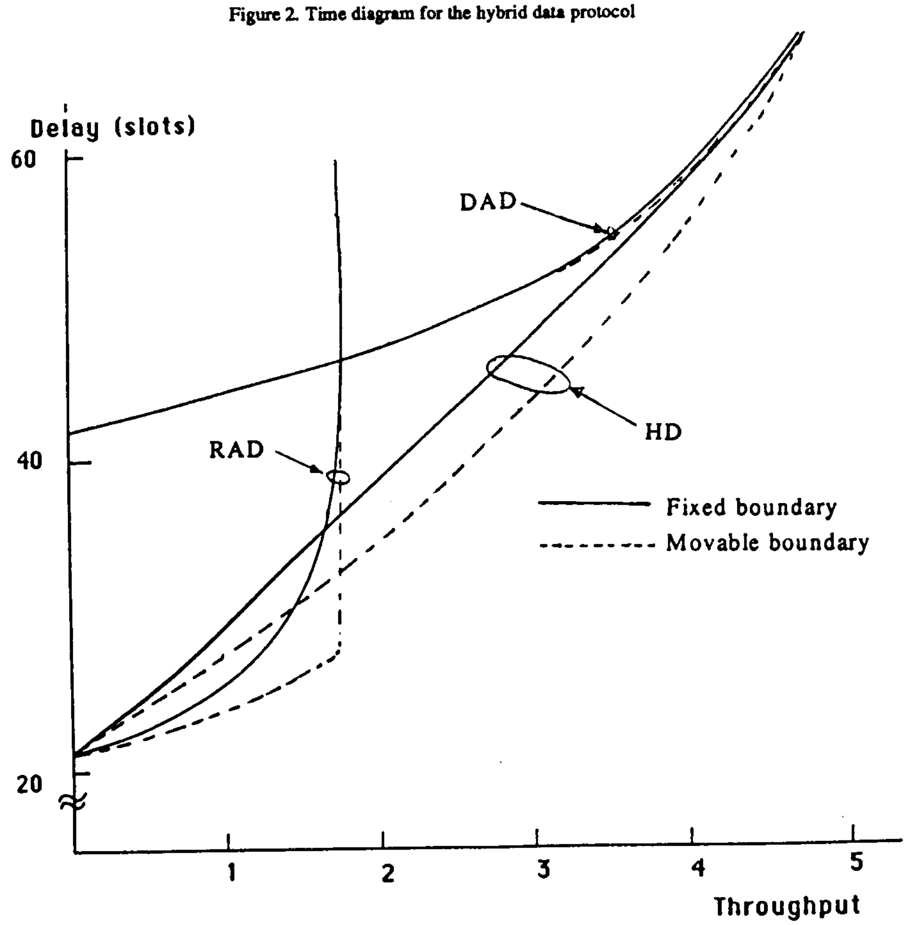


Figure 3. Performance comparisons of the three data protocols under the fixed and movable boundary strategies.