Route Selection for Data Packets in a Tactical Packet Radio Network Using Integrated Voice and Data Transmissions

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Abstract—Integration of data networking into traditional voice only systems, such as SINCgars and HAVEQUICK, is necessary to meet evolving multimedia communication requirements. The success of this integration is dependent on the development of improved channel-access and network protocols that allow simultaneous voice and data information transfer. In previous work we investigated a system design that includes a new waveform and channel-access protocols to support integrated voice and data transmissions in a multiple-hop packet radio network. In this manuscript we present network protocols that support multiple-hop relaying of data packets in a mobile packet radio network. An adaptive forwarding protocol is defined that can respond quickly to local changes in the environment and traffic load. An OPNET simulation, which models the physical, channel-access, and network layers, is implemented to characterize the performance of the network protocols. We show the ability of the network to support relaying of data traffic, and we show the improvement in performance that is achieved through the use of the adaptive forwarding protocol.

I. INTRODUCTION

Support for multimedia communications in packet radio networks is critical for current tactical scenarios. In addition to traditional push-to-talk voice, many applications require support for data traffic. Furthermore, both traffic types need to be handled within one system. One of the challenging obstacles in tactical networks is maintaining reliable operation in situations when all radios are not within range of each other. To ensure a robust network, which does not depend on a subset of the radios to operate, distributed network control is often adopted. A key element in operating a distributed network is the ability of the network to adapt to the transmission environment and changes in the connectivity to neighboring radios. In this manuscript we present a distributed forwarding protocol that is able to quickly adapt to dynamic network conditions in mobile multiple-hop networks.

The radios that make up the network utilize frequency-hop spread spectrum signaling and to a large extent are based upon the SINCgars or HAVEQUICK radios [7]. An improved system design has been proposed for this type of system, and it includes a new waveform [6] and channel-access protocols [1, 2]. Increased efficiency in the support of integrated voice and data transmissions has resulted because of the new system design. In particular, the protocols for handling voice transmissions have been modified so that a radio does not need to transmit continuously during the entire duration of a voice message. Instead, the radio need only transmit voice during approximately one-third of a transmission period, and the remaining two-thirds of the transmission period are available for data packet transfer. The integrated voice and data channel-access protocol allows for data traffic to be forwarded during the idle periods of the voice message. Therefore, the utilization of this idle time for data forwarding vastly improves network throughput over a system such as SINCgars that must delay forwarding all data traffic while a voice message is being transmitted.

For the types of systems under consideration, three traffic types are supported. There is no change compared to the older system in the support for voice messages. A push-to-talk voice message may be generated at any radio in the network and all radios that are neighbors to the source attempt to receive the voice message. Two modes are supported for data traffic. In the local-broadcast mode a data message can be broadcast to all neighbors of a radio, and no link-layer acknowledgments are available. In the point-to-point mode a data message can be delivered to any other single destination, and a link-layer acknowledgment is included in the channel-access protocol.

In this manuscript we investigate a forwarding protocol that is able to enhance the ability of a radio to relay point-to-point data traffic when the network must support all three types of traffic. The routing protocol maintains multiple routes to each destination for relaying of point-to-point data. A key element in the forwarding protocol is that it selects, from among the available routes, the radio that should relay a packet. The selection is based both on information from the routing protocol about the quality of the links in the route and on the level of voice traffic at the neighbors of this radio. Extending the forwarding protocol to include information about the local level of voice traffic improves the ability of the network to support both voice and data traffic.

II. SYSTEM OVERVIEW

Certain details of the physical and link layers influence the design approach for the routing and forwarding protocols of the network layer. We briefly review the important features of the waveform format and the channel-access protocols, and additional details are given in [1] and [2]. The frequency-hop radios utilize two waveform formats depending on the type of information that is being transmitted. For the data-only format, the transmission begins with a 16 dwell interval...
synchronization preamble followed by an error control block that is interleaved over 32 dwell intervals. The voice-plus-data waveform format also begins with a 16 dwell interval synchronization preamble, but the waveform is divided into a voice frame that is 16 dwell intervals and a data frame that is 32 dwell intervals. During a voice talk spurt, a low-rate voice-encoding algorithm generates one voice packet every 216 milliseconds. This voice packet is transmitted in the voice frame of the voice-plus-data waveform, and the data frame remains unused until a radio reserves it for data transfer. Typically a voice message consists of multiple voice packets, so the two frames in the voice-plus-data waveform are repeated several times for each voice message.

III. CHANNEL ACCESS

The channel-access protocol is designed specifically for a distributed, mobile packet radio network, and it does not require any central control. There are several modes for the channel-access protocol depending if a radio is idle or transmitting voice, if the transmission type is broadcast or point-to-point, and if the information is voice or data. An idle radio with a broadcast message to transmit utilizes an ALOHA method to gain access to the channel. For these transmissions, a synchronization preamble is sent followed by the broadcast message. If the message is a voice transmission, the voice plus data waveform is used, and the data frame is initially unused. An idle radio with a point-to-point data packet to transmit gains access to the channel by performing a request-to-send/clear-to-send (RTS/CTS) protocol. A successful RTS/CTS exchange allows the data transfer to occur on a receiver-directed hopping pattern using the data-only waveform. For radios involved in a voice transmission, data transfer is allowed in the data frame of the voice plus data waveform. Contention for use of the data frames is handled using an RTS/CTS protocol during one of the data frames. After a radio reserves the data frames, data transfer occurs in successive data frames.

The channel-access protocol also handles the situations when a radio that is idle attempts to send data packets to a neighbor involved in a voice transmission and vice-versa. A radio that is transmitting or receiving voice has the timing of its hopping pattern, within the propagation delay time, synchronized to the group of radios involved in the transmission. We refer to this set of radios as the voice broadcast group. A radio in this situation is attempting to break into or out of a voice broadcast group, and an RTS/CTS protocol is utilized to control access. Further information on channel-access methods for the system is available in [1] and [2].

The waveform formats and channel-access protocol requires that a radio that is idle monitor the channel for all synchronization, RTS, and CTS packets. Based on the information contained in these packets, the channel-access protocol can track the times that a neighbor radio is involved in a voice transmission. The voice activity indicator (VAI) specifies if a neighbor radio is or is not handling a voice transmission. If both radios are involved in a voice transmission and the timing of the voice frames is aligned, we say the neighbor is involved in a compatible voice transmission. A compatible voice transmission implies that it is possible to forward data traffic to the neighbor while both radios are transmitting or receiving voice. If the timing of the voice frames is not aligned, it is not possible to forward data to the neighbor radio while voice is being handled by both radios.

The voice activity indicator time (VAIT) is the time at which a radio receives a packet from a neighbor that indicates the status of the neighbor radio. Because a radio can only update the VAI for a neighbor after receiving a transmission from that neighbor, the VAIT is utilized by the forwarding protocol to determine the age of the VAI information. The utilization of the VAI and VAIT by the network layer protocols is discussed in the forwarding protocol section.

IV. ROUTING PROTOCOL

An adaptive routing protocol is required to provide a fully distributed organization for delivering data packets to any radio in a multiple-hop network. We have implemented least-resistance routing (LRR), a routing technique that uses side information from the receiver to adapt the routes based on conditions in the network. In [3] LRR was introduced, and its performance was examined for a FH packet radio network in which there is multiple-access interference and partial-band jamming. A link metric is defined that characterizes the ability of a radio to forward a packet to a neighbor radio, and it accounts for the quality of the link to the neighbor radio. We utilize the EEA metric as described in [3].

The resistance for a route is the sum of the resistance for each link in the route. A Bellman-Ford type algorithm [8] is used to determine the routes with the lowest resistance. For each destination radio the outgoing link and the number of links in the route with the least resistance is stored in the routing table. This outgoing link is labeled as the primary outgoing link. In addition, a secondary outgoing link is maintained, and this corresponds to the route with the next lowest resistance among routes that do not use the primary link. As described in the next section, a forwarding protocol is employed to determine if a particular data packet is forwarded on the primary or secondary link.

V. FORWARDING PROTOCOL

A radio utilizes a forwarding protocol to select the next radio on the route to a packet’s destination. Destination-based forwarding [8] is utilized, and it requires each radio to make a forwarding decision based on a packet’s destination and the routing information available at the radio. Least-resistance routing will often calculate two routes to the destination, and the forwarding protocol can choose either of these routes based on local conditions and the status of a particular packet. An important feature of the forwarding protocol is the ability to forward data traffic to a neighbor while voice is being handled by both radios.
protocol is that it is not limited to choosing the primary outgoing link only, but can choose the secondary outgoing link if other conditions indicate this may be a better choice. Forwarding protocols for packet radio networks that utilize LRR have been examined in [3], and in this manuscript we extend the forwarding protocol to utilize information from the channel-access protocol about the status of voice transmissions at neighboring radios.

A radio executes the forwarding protocol to determine which outgoing link to use for transmitting a point-to-point data packet. After each attempt to transmit the packet, the channel-access protocol expects an acknowledgment packet from the neighbor radio. If an acknowledgment is received, the radio can discard the packet. However, if no acknowledgment is received, the packet is returned to the radio’s queue for a retransmission attempt. The number of forwarding attempts for the packet is monitored, and if after FTA attempts the packet has not been acknowledged, the packet is discarded.

We consider two forwarding protocols in this manuscript. We refer to the first protocol as the PS-forwarding protocol and it is described in [3]. This protocol does not incorporate information about status of integrated voice and data transmissions. For a packet that has been selected for a forwarding attempt, NRp is the next radio corresponding to the primary outgoing link for this packet’s destination, and similarly NRs is the next radio for the secondary outgoing link. Note that either or both of the next radios may be invalid if the routing table does not have a route to the destination at the current time. Furthermore, the lateral transmission protocol as described in [3] may prohibit the packet from being forwarded on one of the links, and if so, the corresponding next radio is considered to be invalid for this packet. A parameter, FPA, is used to determine how many forwarding attempts are made on the primary link with the remaining attempts made on the secondary link. The number of previous attempts to forward this packet at this radio is FA. The next radio (NRj) is chosen as follows:

\[
\text{If } (F_A \geq F_P A \text{ and } NR_j \text{ is valid}) \text{ then } NR_j = NR_p
\]

\[
\text{Else } NR_j = NR_s
\]

If NRj is invalid then a transmission is not attempted, and the packet is returned to the queue.

The second forwarding protocol extends the PS-forwarding protocol by considering the status of voice transmissions at the forwarding radio, the NR1, and the NR2 where NR2 is the next radio associated with the other outgoing link. We refer to this new protocol as the VAI-forwarding protocol. When required and with certain limitations, the VAI protocol selects a neighbor radio so that it is not necessary to break into a voice broadcast group. Note that this protocol differs from the PS protocol only if NR1 and NR2 are both valid. For this situation the VAIT is examined to determine the voice transmission status of the next radios NR1 and NR2. Therefore, the VAIT value for NR1 and NR2 is tested as follows:

If (current time - VAIT > EVML) NR is marked inactive
Else NR is marked according to the VAI

where NR is NR1 or NR2 and EVML is the expected voice message length which is a design parameter. After the voice transmission status of NR1 and NR2 has been determined, the current status of the channel-access protocol is examined to determine if the forwarding radio is active or inactive in a voice transmission.

Consider the situation in which the forwarding radio is active in a voice transmission. If NR1 is marked as active and it is involved in a compatible voice transmission, NR1 is selected for the forwarding attempt. If this test fails, NR2 is selected if it is involved in a compatible voice transmission. If there is still no resolution, NR1 is selected if it is marked as inactive. Otherwise, NR2 is selected if it is marked as inactive. Finally, if a neighbor has still not been selected, the VAI protocol defaults to utilizing NR1.

Next, consider the situation in which the forwarding radio is inactive in a voice transmission. If NR1 is marked as inactive for a voice transmission, the VAI protocol uses NR1. If this test fails and NR2 is marked as inactive, the route resistance associated with NR2 is tested as shown:

If (RR2 - RR1 \geq \delta) \text{ use } NR_1
Else \text{ use } NR_2

where RR1 is the route resistance for next radio NR1, RR2 is the route resistance for the next radio NR2 and \delta is a predefined limit. Even though the forwarding radio and NR2 are inactive, the alternative route is avoided if its quality is much worse than the first route. Therefore, if the quality of the alternative route is poor, we assume that it is better to break into a voice broadcast group by attempting the transmission on the first route.

Finally, if both NR1 and NR2 are marked as active in a voice transmission and the forwarding radio is inactive in a voice transmission, the forwarding radio must break into a voice broadcast group regardless of the route chosen. In this situation NR1 is selected as the next radio.

VI. RESULTS

A packet radio network is simulated with OPNET, and it utilizes the channel-access protocols, integrated voice and data waveforms, and interrupted frequency hopping patterns as described in [1] and [2]. The traffic generation model for push-to-talk voice is based on the standard talk-spurt model developed by Brady [5]. A two-state Markov model is utilized to generate voice packets. If the process is in the talk state, an exponential random variable is used to determine the number of voice packets to generate, after which the process changes to the quiet state. The length of time the process stays in the quiet state is determined by an exponential random variable using a different distribution. The length of
voice messages is limited to a minimum of five and a maximum of 75 voice packets. This limits the duration of a talk spurt to between one and fifteen seconds. Data messages are generated independently at each radio, and the length of a data message is one data packet. The inter-arrival times of data messages are generated from an exponential distribution, and the data generation rate is equal to the reciprocal of the expected inter-arrival time.

The performance of the forwarding protocols is examined in a thirteen-radio network, and the topology is illustrated in Figure 1. The radios in the shaded circle are all within range of each other, and the lines show the connectivity of the remaining radios. Radio 0 is the only source of voice messages, and radios 0-4 form a voice broadcast group.

![Figure 1. Thirteen-radio topology.](image)

The network performance is evaluated by measuring the end-to-end delay, and end-to-end success rate for data packets for four origin-destination pairs: (5, 6), (6, 5), (7, 8), and (8, 7). All packets generated at radios 5–8 are for the corresponding destinations only. These packets are marked and their flow through the network is monitored to determine the delay and success rate. Data packets are also generated at radios 1–4 and the generation rate is fixed at 0.04 packets/sec. at each radio. The destinations for these packets are randomly selected with a uniform distribution over all other radios in the networks. These packets are forwarded to their destinations but they do not contribute to the measured delay or end-to-end success rate. The end-to-end success rate is the fraction of marked packets that reach their destinations; the end-to-end delay is the time in seconds for those packets that reach their destinations to do so. Note that packets that are discarded do not contribute to the end-to-end delay in our simulation. Such packets must be detected by the end-to-end acknowledgment protocol, and they are resubmitted to the network at a later time. We have not included an end-to-end acknowledgment protocol in the simulation.

The success rate for voice packets is the fraction of voice packets that are received by radios 1–4. Because voice packets are given priority by the channel-access protocol, the delay for delivering voice is dominated by the encoding and interleaving delay. This is approximately 0.2 seconds.

A transmission attempt for a data packet is unsuccessful if an ACK is not received, and a radio can try forwarding a packet up to six times; \( F_{PA} \) equals 5. Also, a radio discards a data packet if it has not reached its destination within 20 seconds. The VAIT is 12 seconds and \( \delta \) is equal to 5 for the results included in this manuscript on the VAI forwarding protocol. We have considered other values for the VAIT and \( \delta \) and have found that the network performance is not very sensitive to the values of these parameters. Other details of the simulation model are described in [1] and [2] and the references cited therein.

The end-to-end success rate for marked data packets is shown in Figure 2. The level of voice traffic that is generated by radio 0 in this example results in radio 0 transmitting voice in the integrated voice and data waveform approximately 60% of the time. In this situation, the end-to-end success rate is larger for the VAI forwarding protocol than the PS protocol, particularly at lower data generation rates. Because it is likely that the radios in the voice broadcast group are busy, transmission attempts by radios 5–8 to radios in the group are more likely to be unsuccessful. Hence, even though a route from 5 to 6, for example, that avoids the voice broadcast group is likely to require more relays than a route through the group, the probability that the individual relays are successful is greater. Not only is a packet more likely to reach its destination in this example, but also the radios that are not in the voice broadcast group are likely to require fewer unsuccessful transmissions to forward marked packets. As a consequence, these radios are more likely to be available to relay other traffic.

The end-to-end delay for marked data packets is shown in Figure 3. The delays for the two forwarding protocols are very similar. The VAI forwarding protocol does result in a larger delay for data traffic, primarily because the packets typically require an additional relay. In Figure 4, the success rate for the voice traffic is illustrated. There is some improvement in the success of voice with the new forwarding protocol over the older approach.

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VII. CONCLUSIONS

The VAI forwarding protocol introduced in this manuscript has been designed to work in conjunction with the existing channel-access and routing protocols for an integrated voice and data system. The VAI forwarding protocol improves support for data traffic in a multiple hop network that must support both push-to-talk voice and data. Furthermore, a slight improvement in voice success rate is obtained. The VAI protocol has the greatest impact when the voice level is relatively heavy. At lower voice generation rates, the VAI protocol still improves the voice and data success rates. However, the improvement is not as large.