IMPROVED PERFORMANCE FOR INTEGRATED VOICE AND DATA IN A TACTICAL PACKET RADIO NETWORK

Ronald D. Bowman†, Robert D. Martin‡, James R. McChesney‡, N. P. Robinson‡‡, and Harlan B. Russell†

†Techno-Sciences, Inc.
511 Westinghouse Rd.
Pendleton, SC 29670

‡ITT Industries
1919 W. Cook Rd.
Fort Wayne, IN 46801

‡‡U. S. Air Force
Air Force Research Lab/IFGB
Rome, NY 13441-4505

ABSTRACT

A new programmable radio platform (PRP) has been designed by ITT, and two of its significant features are a new frequency-hop modem and a general purpose processor. This manuscript describes a joint project to design and simulate a new channel-access protocol that is targeted for the capabilities of the modem. Our channel-access protocol exploits the improved modem design and a low bit rate voice codec to provide integrated voice and data communications in a military frequency-hop packet radio network. The protocols are implemented in OPNET, a computer aided design tool, and they will be ported to the general purpose processor in the PRP. We demonstrate that an open design for a radio system that uses general purpose processors allows new networking protocols to be quickly developed, tested, and implemented in actual hardware.

INTRODUCTION

Increased information throughput is necessary to meet evolving multimedia communication requirements for concurrent delivery of voice and data traffic in tactical packet radio networks. Contention for the RF communication channel and inefficient voice coding and modulation in current communication networks severely limits the integration of data traffic into traditional voice only systems, such as SINCGARS and HAVEQUICK. The data capacity, or throughput, of these packet radio networks is well below what is required to meet emerging performance requirements.

An improved waveform along with a new channel-access protocol has been developed that increases the data throughput of existing tactical systems. These new communication protocols provide integrated voice and data communications while greatly increasing the network information capacity and communication range. The target radios utilize MELP speech compression and improved modulation capabilities, and the waveform format we are specifying exploits these new capabilities. Channel-access protocols are devised that decrease contention for the channel and provide for the integration of voice and data to allow multiple messages to be delivered concurrently.

ITT Aerospace/Communications Division has developed a programmable radio platform (PRP) that allows various subsystems in the radio to be easily upgraded or replaced. The PRP has capabilities that are similar to the Programmable Modular Communications Systems (PMCS) suite of equipment currently under development. A commercially available bus allows either custom or commercially available cards to be inserted into the PRP. ITT has developed custom RF and modem modules, and has purchased general purpose DSP and processor cards. The DSP card controls the modem while the processor cards handle tasks such as I/O, voice coding and decoding, and custom control software. For example, one of the cards implements a MELP speech module to provide low bit rate voice encoding. While a variety of modem modules are planned by ITT, the modem utilized for this project provides a frequency-hop (FH) spread spectrum waveform.

Techno-Sciences, Inc. in cooperation with ITT and Rome Laboratory has developed a new channel-access protocol that exploits the features of the new FH modem. We have implemented the protocol design in OPNET, a computer aided design tool, and we plan to port the channel-access protocols to one of the general purpose processors hosted in the PRP. The OPNET simulation models the FH modem performance and allows us to verify and demonstrate the performance of the channel-access protocols before they are implemented in the hardware. The open definition of the modem and the general purpose processor allows us to easily port and test the new channel-access protocols in the PRP.

In this manuscript we first describe the capabilities of the modem, followed by a description of the channel-access algorithm. We present results for a fully connected network with one radio as the source of voice packets, and we show the performance improvement when the integrated voice and data channel-access capability is utilized. For this manuscript, our design focuses on a fully connected network of radios. In a companion paper, [1], additional results are presented on extending the channel-access protocols for a mobile, multiple-hop packet radio network.
WAVEFORM FORMAT FOR THE PRP FH MODEM

ITT has developed a new FH modem that extends the capabilities of existing FH modems such as SINGCARS and HAVEQUICK. The modem supports two waveform formats: one format is designed to carry data messages only, and the second format is designed to carry voice and, optionally, data messages. The second waveform supports the integrated voice and data channel-access protocol, and is called the V+D waveform. There are many improvements in the new modem design but only the features that impact the channel-access protocol design are discussed here.

If a radio is not transmitting or receiving, it is unlikely that will be able to maintain frequency-hop synchronization with other radios. Hence, all waveforms begin with a synchronization preamble. The synchronization preamble, which utilizes a pattern known to all radios, allows a potential receiver to adjust its hop timing to that of the transmitting radio. Furthermore, a small amount of additional information can be included with the synchronization preamble, including the type of waveform, and if applicable the source and destination radio identification numbers.

There are two variations of the data only waveform, and they are illustrated in Figures 1 and 2. A data message is encoded into one or more error control blocks (ECB’s), and the number of ECB’s depends on the size of the data message and the code rate. For this modem each ECB is comprised of 24 codewords from a (32,12) Reed Solomon code. The codewords are block interleaved across 32 hops with each hop containing one symbol from each codeword. The coding and ECB structure is similar to one of the methods available in the SINGCARS radio. For the results presented in this manuscript we limit the number of ECB’s in a single data message to one. The data only waveform is utilized for a transmission that is to be broadcast to all radios that are within range of the transmitting radio.

The data-RTS waveform is utilized for a transmission that is intended for a particular neighbor of the transmitting radio. For this waveform the eight hops after the data synchronization preamble are reserved for the intended receiver of the packet to respond with a clear-to-send packet (CTS). Also, after receiving the final ECB the receiver responds with an acknowledgment packet (ACK), and the ACK is transmitted in eight hops. The waveform format includes a delay between the reception of the final ECB and the generation of the ACK to allow for the packet to be decoded before the ACK is transmitted. However, our simulation model does not include the decoding time.

The availability of the MELP codec in the PRP allows for a new waveform format because a radio with voice information to transmit need not transmit continuously. By utilizing the V+D waveform a radio can transmit data messages during the intervals in which voice packets are not transmitted. The V+D waveform structure is illustrated in Figure 3. A voice synchronization preamble provides FH synchronization, and the remainder of the transmission is segmented into 48 hop super-frames. Each superframe hosts two frames with the first frame reserved for voice and the second frame allocated for data.

![Figure 3. Voice + Data waveform.](image)

The waveform is utilized when one radio has voice to transmit and the channel-access algorithm has determined that the channel is idle. Initially, only the voice frames are occupied. After a voice transmission has begun, the radio transmitting the voice or any radio receiving the voice may utilize the data frames. The channel-access protocol coordinates access to the channel and it is described in greater detail in the next section.

CHANNEL-ACCESS PROTOCOL

Our channel-access scheme is designed specifically for a distributed packet radio network, and it does not require any type of central control. Only those features of the channel-access protocol that are required for a fully connected network are described in this manuscript. Extensions to the modem, waveform, and channel-access protocol that enable efficient support for a multiple-hop packet radio network in which packets may be relayed is described in [1].

The channel-access protocol is organized into four major states, as illustrated in Figure 4. The protocol is in the idle state if there are no transmissions or receptions currently in progress. The protocol changes states if there is a packet to transmit or if the radio receives a synchronization preamble from another radio. A radio is initially in the idle state, and it attempts to receive any synchronization preamble that is detected.

**Idle to data only state transition.** If a radio has a data message to forward, or detects a data synchronization preamble, the channel-access protocol transitions to the data only state. The data only waveform does not permit simultaneous voice messages, hence a radio must wait until the data message is completed before initiating another action. However, the length of data transmissions is limited, so the delay before a voice
transmission can be attempted is acceptable.

There are two waveforms that can be utilized for a data only transmission. The data only waveform is utilized for data that is broadcast to all neighbor radios, and the data-RTS waveform is utilized for data that is intended for a particular neighbor. If a radio detects a data only synchronization preamble, it attempts to receive the data packet. On the other hand, if a radio detects a data-RTS synchronization preamble and determines that it is the destination, the radio transmits a CTS. All other radios return to the idle state. Only the radio that is the intended destination will continue to monitor the hopping pattern for this transmission, and it is the only radio that receives the packet. After the packet has been received and decoded, an ACK is transmitted provided the packet decoded successfully.

**Idle to voice only state transition.** If a radio has a voice message to forward, or detects the V+D synchronization preamble, the channel-access protocol changes to the voice only state. All radios attempt to receive the synchronization preamble and the voice message. The radio that is transmitting the voice utilizes the first sixteen dwells in each super-frame, and the remaining 32 dwells are idle. When the voice message completes (and assuming no data message is in progress) the protocol moves back to the idle state.

The voice only and data only states use an ALOHA channel-access protocol, and as such do not represent a new approach to channel access. The new feature of this protocol is the ability to initiate a data transmission while all radios are involved in a voice transmission. This capability is available because the modulation in the FH modem has been improved to provide a higher data rate than existing FH modems, and the MELP voice codec operates 2.4 kbps versus 16 kbps for typical fielded systems.

**Voice only to V + D state transition.** A radio transmitting or receiving voice can also initiate or receive a data transmission. A data message with a specific destination is transmitted using a point-to-point protocol while a data message with a broadcast destination is transmitted using the broadcast protocol. For either protocol, the source of the data message transmits an RTS, and all the radios that are receiving the voice transmission attempt to receive the RTS.

For the FH waveform implemented in the PRP modem, all radios monitor the same hopping pattern. Because of this, all radios that are receiving a voice transmission are also monitoring the same hopping pattern during the data frame. Thus, a synchronization preamble is not required for these radios to receive a RTS packet. The first sixteen dwell intervals of a data frame are used for the RTS/CTS transfer. The first nine dwell intervals are segmented into three three-dwell mini-slots, and the radio randomly chooses one of these mini-slots for the RTS transmission. The next four dwells are idle and allow the destination radio to process the RTS and prepare a CTS if necessary. The final three dwell intervals are reserved for the radio that is transmitting the CTS. The next sixteen dwell intervals in the data frame are reserved. A radio that does not acquire the synchronization preamble for a voice transmission will not synchronize to the V + D waveform. A protocol to utilize the reserved dwell intervals to synchronize a radio to the V + D waveform is discussed in [1].

More than one radio can transmit an RTS in a particular data frame, and the protocol for responding depends on the type of transmission that is requested. For a broadcast transmission request, the radio transmitting the voice selects the first RTS that it receives and transmits the CTS indicating which radio can transmit the data message. If the source of the voice transmission has a data packet to send it does not send an RTS, but transmits a CTS indicating that it will send the data message. In the next super-frame the source of the data begins the data message transmission.

The operation of the point-to-point protocol is similar. The source of the data message transmits an RTS and all radios will attempt to receive it. However, the radio that is the destination for the data message responds with the CTS. The source transmits the data packet but only the destination receives and decodes the transmission. Finally, if the packet is successfully decoded, the destination responds with an acknowledgment packet (ACK). The operation of the channel-access protocol when integrating point-to-point data into the V+D waveform is illustrated in Figure 5.

**Figure 2. Channel-access states**

**Figure 5. Voice and data integration with point-to-point data.**

The difference between integrating a point-to-point and a broadcast data transmission into the V+D waveform is that the radio selected to receive the packet responds with the CTS instead of the radio that is transmitting voice. In addition, because there is only one recipient of the data packet, a final acknowledgment is also generated. Because of the protocol for an RTS for point-to-point transmissions, it is possible that more than one CTS could be transmitted in the same data frame. However, in this case it is very unlikely that any of the CTS’s will
be successful and none of the radios with a data message will begin a transmission.

**V + D to either voice or idle state transitions.** In the V + D state, either the voice or data transmission may complete first. If transmission of the data message completes before the voice message, the protocol moves back to the voice only state. However, if the voice message ends before the data message completes, the V+D waveform is continued with the voice frames remaining idle. Once the radio transmitting the data message is finished, the protocol returns to the idle state.

After completing a transmission attempt, the radio will generate a random backoff time before another transmission can be initiated at this radio.

**RESULTS**

Computer simulations are used to compare the integrated voice and data channel-access protocol with a channel-access protocol that uses ALOHA for voice only or data only transmissions (i.e., for the ALOHA protocol the V + D state is not permitted). An OPNET simulation approximates the behavior of the ITT FH modem and includes the effects of multiple-access interference, path-loss variations, and the received signal-to-noise ratio. The method for modeling the FH signaling is nearly identical to the one specified in [2] and the references cited therein.

We consider a network of ten radios with one radio selected as the source of voice traffic. The traffic generation model for push-to-talk voice is a modified version of the standard talk-spurt model developed by Brady [3]. In the talk state, an exponential random variable is used to determine the length of the speech interval, and the expected value for the exponential distribution is ten seconds. However, the length of the speech interval is limited to a minimum value of one second and a maximum value of four seconds. An exponential random variable is also used to determine the time between talk spurts and we examine scenarios where the expected inter-arrival time varies from one-half to three seconds. We have focused on voice generation rates for which the fraction of the time that the source of voice traffic is busy transmitting varies from 50% to 80%.

The network performance is evaluated by measuring the throughput, delay, and success probability for both voice and data packets. 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 equal to the length of a super-frame and is approximately 0.2 seconds.

The data traffic source generates a single packet at a time and the packet size is fixed to correspond to the size of an ECB. An independent exponential distribution is used for each source to specify the interarrival time between packets. The rate that a traffic source generates packets is equal to the reciprocal of the expected interarrival time. The destination for a data packet is determined in one of two ways. If the source of the data has a specific destination, then all packets are forwarded to that radio. Furthermore, we mark these packets and monitor the throughput, delay, and success probability for each source-destination pair. The second method for specifying the destination is to randomly choose one of the other radios in the network.

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 eight times. Also, a radio discards a data packet if it is not successfully forwarded within twenty seconds.

For the results included in this manuscript, one radio generates voice messages. For the first set of results one other radio generates point-to-point data packets and these packets are marked. The fraction of data packets that are successful versus the data message generation rate is shown in Figure 6 for two

![Figure 6. Success probability for data packets with one source of data traffic.](image)

![Figure 7. Delay for data packets with one source.](image)
different voice traffic generation rates. The high voice rate corresponds to a generation rate of 0.5 for voice traffic and the low voice rate corresponds to a generation rate of 2.5. We show the performance for the channel-access protocol when utilizing the integrated V + D waveform versus ALOHA. The fraction of data packets that are successful is greater when the V + D waveform is utilized, and the performance improvement is larger at higher voice generation levels. At the higher voice level the radio that is the voice source uses the V + D waveform approximately 80% of the time.

The delay for the data packets is shown in Figure 7 for this scenario. Illustrated in Figure 8 is the success probability for all voice packets. Considerably lower delay is achieved with the V + D protocol because the radios are able to exchange data message while a voice transmission is active. Likewise the success probability for voice packets is also higher compared to ALOHA. The success probability for voice packets is nearly equal for both voice generation rates.

The second scenario considers the same network but with three radios generating data messages. One of the data sources generates marked data packets and the other two generate unmarked data packets where the destination is chosen randomly among all the radios. All three sources of data generate packets at the same rate. The results for the marked packets are shown in Figures 9 and 10. At the lower data generation rates the delay for data packets is considerably lower when the V + D protocol is utilized as compared to ALOHA.

Our new channel-access protocol takes advantage of the V + D waveform and allows the concurrent delivery of voice and data packets. In particular, delivery of data messages can occur while voice is in process, resulting in both lower delays for the data traffic, and a higher probability of success for the voice messages.

REFERENCES