An Adaptive VHF/UHF System for the Next Generation Tactical MANETs

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ABSTRACT

Tactical Mobile Ad Hoc Networks (MANETs) have a continued growth in bandwidth demand mainly driven by the introduction of new user services and applications. Everything over IP is one of the main requirements of the next generation tactical MANETs. As part of an initial investigation to provide enhanced tactical IP networking capabilities, an adaptive VHF/UHF system that attempts to satisfy the mobility, capacity and adaptability requirements of the new tactical environment was developed. The solution includes a multi-rate Orthogonal Frequency Division Multiplexing (OFDM) VHF/UHF line-of-sight (LOS) radio modem and a fully distributed multi-channel media access protocol that supports integrated voice and data. A description of the two technologies is provided along with the main features and novel functionalities of the system.

INTRODUCTION

Tactical networks have a growing demand for bandwidth driven by the introduction of new user services and applications. Everything over IP is one of the main requirements of the next generation of tactical MANETs. VHF/UHF network users are now expecting to be able to share, in addition to voice, a variety of IP-based data services such as chat, email, file transfers and Situational Awareness (SA). This requires a system that can provide increased throughput without compromising robustness and long range signal coverage, two important characteristics of VHF/UHF communications.

As part of an initial investigation to provide enhanced tactical IP networking capabilities, an adaptive VHF/UHF system that attempts to satisfy the mobility, capacity and adaptability requirements of the new tactical environment was developed. An OFDM VHF/UHF modem based on Software Defined Waveform (SDW) technology was combined with a fully distributed multi-channel media access protocol. The two technologies operate within a highly flexible framework that conforms to Software Defined Radio (SDR) principles. The communication occurs via abstract generic interfaces. This paper presents a description of the two technologies along with their main features and novel functionalities.

TACTICAL MANETS

Tactical MANETs are formed of broadcast radios where spectrum is shared among the users. The communication is half-duplex, i.e., at any point in time, a radio system can either transmit or receive but can never perform the two operations simultaneously. Networks are deployed with no fixed infrastructure. In addition, each node is mobile and can move in and out of the transmission range of another node. Consequently, networks may include multi-hops where each node may play the role of a relay node.

Particular to VHF/UHF tactical MANETs is the fact that nodes communicate with each other across bandwidth-constrained radio links. In the VHF/UHF bands, the spectrum availability is limited. For the networks we consider, a channel allocation typically varies from 25 kHz to 200 kHz of bandwidth. In this bandwidth limited environment, the challenge lies in providing increased capacity for the network operations while maintaining robust and long range signal coverage.

The architecture of a tactical MANET node is shown in Figure 1.

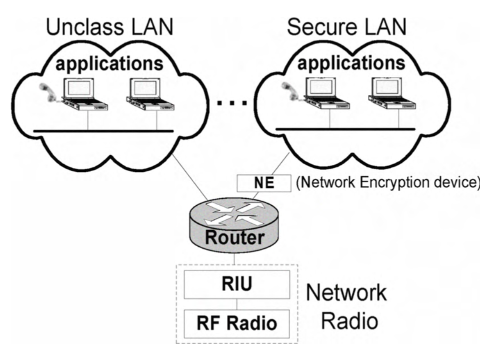

Figure 1: Tactical MANET node architecture

This typical MANET node architecture is found for example in the maritime environments with ships at sea or in land tactical domains with deployed ground vehicles. The node is composed of many devices. It includes a network router to which one or multiple LANs of possibly different security classifications are connected. The LANs host the various IP-based applications and data services used during operations. The network router is connected to a network radio device. The network radio device is typically made of two components: a Radio Interface Unit

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(RIU) and the RF unit of the radio. The RIU interfaces between the IP data router and the RF unit. The RIU essentially implements the protocol stack that provides IP data transfer in a multiple-node, multiple-hop dynamic network.

An RIU has been developed that includes a multi-rate OFDM VHF/UHF radio modem and a fully distributed media access protocol that supports integrated voice and data.

THE RIU PROTOCOL STACK

An overview of the RIU protocol stack is shown in Figure 2.

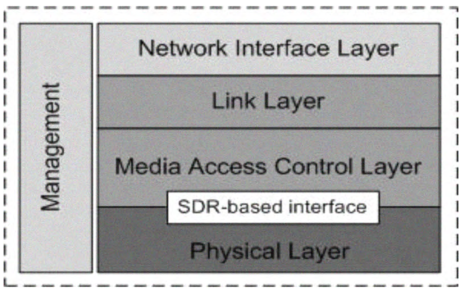

Figure 2: RIU protocol stack

The RIU is a system that allows VHF, UHF and potentially other types of bearers to be made available for the transport of IP data. It includes a set of protocol layers that provide a number of services: the network interface layer communicates with the attached network. It essentially performs the translation of the information between the network layer and the link layer. The link layer acts as an adaptation layer, responsible for organizing the network data into logical blocks for transmission by the media access control (MAC) layer. It manages the queuing strategy in support of the Quality of Service (QoS) scheme. It also performs packet segmentation/reassembly, duplicate detection and may perform data compression/decompression functions. If required, the link layer also ensures a reliable transfer of data across the link. The MAC layer coordinates transmissions on the shared channel and enables ad-hoc operations. Finally, the Physical (PHY) layer interfaces with the RF unit. It ensures the transmission/reception of the data over the on-air bearer.

SYSTEM REQUIREMENTS AND DESIGN GOALS

The design of the system was guided by the objective of developing an enhanced wireless solution that satisfies the requirements of future tactical MANETs. VHF/UHF tactical networks are relatively small networks. The total number of nodes is not expected to be greater than 30. Tactical networks are designed to operate in a variety of environments ranging from the open sea to urban areas. A variety of scenarios, such as fixed, mobile and multi-hop, must be supported. A highly flexible and adaptable tactical network leads to higher complexity in network formation. Self-configuration with dynamic node joining/leaving becomes a necessity.

The network must be IP capable. This implies the support of a variety of mainstream services such as real-time voice, video, chat, SA data, command and control data, web browsing, file transfers, messaging, email and database access and replication. Unicast, multicast and broadcast communications are expected and thus support for these various addressing schemes must be included. Given the relatively low-bandwidth environment, priority and preemption mechanisms are necessary to ensure resource availability for the more important traffic.

Next generation tactical radio networks will need to be deployed in various radio bands and thus will be required to support a variety of bandwidths and data rates. Such conditions may either be imposed by spectrum policy or dynamically derived by cognitive/context aware network protocols. The latter implies the use of cross-layer information exchange.

Physical layer modems built on the SDR concepts and offering highly flexible and adaptable waveforms are well suited for the requirement. Furthermore, to satisfy the real-time network constraints, the modem should exhibit excellent short burst mode characteristics such as robust and accurate synchronisation, auto-baud agility, low overhead and minimal transmit/receive processing delays.

THE MAC PROTOCOL

Because the radio channels are shared amongst the participating nodes, a scheme that coordinates access to the channels is required. A distributed multi-channel media access protocol that supports integrated voice and data has been developed. Named the *Multiple Access for Tactical Radios with Integrated Quality of Service* (MATRIQS), the protocol is based on synchronous Time Division Media Access (TDMA). Many distributed TDMA-based channel access schemes for ad hoc networks have been proposed previously. Most of them however, satisfy only partly the tactical MANET requirements. In [1], the proposed scheme does not address multi-hop networks, eliminating the difficulty of addressing the hidden node problem. Other solutions such as [2] and [3] offer interesting multi-hop self-configuring TDMA schemes but lack in providing support for traffic other than non real-time. While [4] and [5] propose solutions that seem to cover many of the desired tactical MANET functionalities including support for integrated voice and data, the support for dynamic bandwidth on-demand of non real-time traffic seems limited and there is no mention of priority and preemption support for real-time traffic flows. The MATRIQS protocol attempts to bring together all these desired capabilities. In particular, the solution offers:

- Self-forming/self-configuring networks
- Support for mobility (adapts to topology changes)
- Hidden node/Exposed node tolerance
- Optimization of bandwidth usage
- Support for scalability (multiple frequencies)
- Ability to support concurrent voice and data traffic
- Support of priority & preemption
- Provision for QoS support.

The following sub-sections describe, in overview, how the above capabilities are addressed by the protocol.

A. Real-time and non real-time integrated services

One of the main key features of the MATRIQS protocol is its ability to handle a variety of IP services and in particular, successfully integrate services such as voice and data. The challenges in doing so lie with the ability to support the different transmission characteristics and requirements of the two traffic types. Such considerations need to be taken when defining the TDMA structure. Figure 3 shows the TDMA cycle structure used by the MATRIQS protocol.

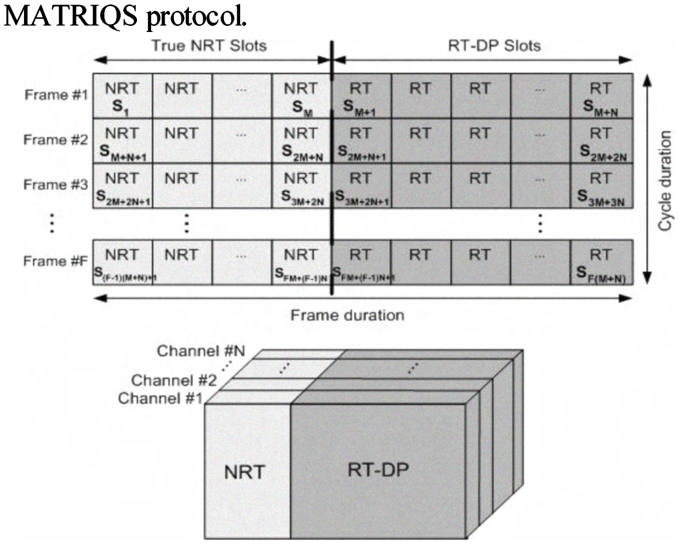

Figure 3: MATRIQS TDMA cycle structure

The protocol defines two types of slots: the true Non Real-Time (NRT) slots and the Real-Time Dual Purpose (RT-DP) slots. The true NRT slots are strictly used for the transmission of traffic other than real-time while the RT-DP slots are reserved for the transmission of real-time (voice) flows. In this structure, a voice flow, when present, occupies all the slots of a RT-DP column, i.e., data of a given voice flow is transmitted on the same RT-DP slot index at every frame. This mapping ensures a deterministic access delay that guarantees to meet the strict timing requirement of voice. The maximum number of supported RT flows per frequency is thus equal to the number of RT-DP columns of the matrix. When multiplexed over multiple frequencies, this slot structure offers an interesting scalability option.

To optimize bandwidth usage, the RT-DP slots are dual purpose, i.e., whenever unused by RT traffic, these slots can be used to transmit NRT traffic, in which case, they effectively become temporary NRT slots. Because NRT traffic is allowed to borrow bandwidth dedicated to RT, the total bandwidth actually used by the NRT traffic will dynamically grow and shrink depending on the number of RT flows present at any one time in the network. Care must be taken to ensure that NRT traffic is never completely starved, that is, the total bandwidth occupied by RT application flows at any point in time is limited to some fraction of the total capacity of the network. To enforce this, the system defines a *minimum true NRT slot allotment* value. Essentially, each node maintains a minimal number of NRT slots as specified by the *minimum true NRT slot allotment* value. This value cannot be less than one. It is preferably set to two or more and is policy-defined. This value is used to derive the ratio of NRT slots to RT-DP slots which effectively determines the minimum and maximum bandwidth guaranteed to NRT and RT traffic respectively.

As the various application data packets arrive from the network, they are queued at the link layer according to their traffic type (RT/NRT) and priority. When a transmission slot comes up, the slot is packed with the appropriate data. Since NRT includes different types of traffic ranging from near real-time (e.g. routing data, positioning reports) to best effort (email, background data transfers), the packing of NRT slots will always favor the higher priority traffic over the less important traffic.

B. Self-configuring and self-forming networks

The system is completely self-configurable and thus, the format of the cycle structure shown in Figure 3 is not determined in advance. At start time, the system derives the TDMA structure values based on a number of parameters it obtains either via some user-defined input or directly from the various devices it operates with. For example, the type of vocoder used for the voice application drives the frame duration computation. The radio key on and key off delay values as well as the modem waveform parameters directly affect the slot overhead and consequently impact the slot duration computation. Whenever possible, these parameters are obtained automatically, keeping the complexity hidden from the user. This is the case, for example, for all of the modem/radio parameters which are exchanged via an SDR-based interface between the PHY and the MAC layers. This design approach offers the benefit of providing a highly adaptable system that can be deployed in different environments with minimal reconfiguration effort.

The MATRIQS protocol defines five states of operation. The state diagram and its transitions are shown in Figure 4.

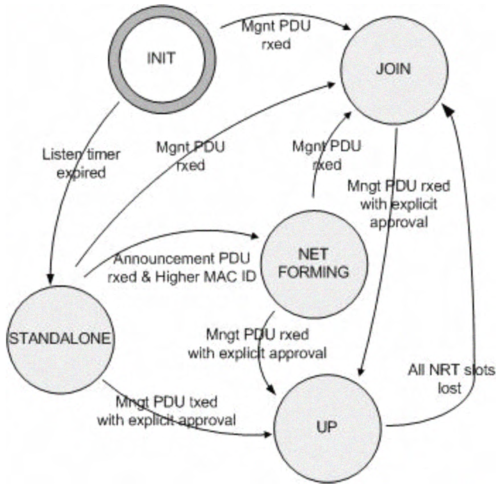

Figure 4: MATRIQS protocol state diagram

The *INIT*, the *STANDALONE*, the *NET FORMING* and the *JOIN* states all relate to the capability of the protocol to support self-forming/self-organizing networks. The *INIT* state is entered immediately after start-up. In this state, the node listens to gain local information about the network. It listens for a (configurable) minimum of 1 cycle duration. This is to allow for existing network discovery. If no existing network is detected before the end of the listening period, the node enters the *STANDALONE* state. In this state, the node announces its presence. Once per cycle, the node selects, at random, an NRT slot in which it transmits an announcement message. The node will go on transmitting announcement messages until it either hears an announcement message from another node or detects the presence of an existing network. In the latter case, the node will transition to the *JOIN* state. In the former case, the node will enter the *NET FORMING* state and will attempt to form a network with the remote peer.

In both, the *JOIN* and the *NET FORMING* states, a node attempts to gain ownership of at least one NRT slot by transmitting join request messages. The protocol does not actually set aside any special slots in support of the joining process. That approach, often adopted in TDMA-based schemes [2][4][5][6], was avoided here as considered too wasteful of precious network resources. Instead, it was chosen to rely on the highly dynamic slot request and release scheme that is implemented by the protocol. Essentially, the node makes use of slot information reported by its neighbors (described later) to select an available slot in which to issue the join message as well as to select the slot to request in the message. Situations may occur in which the node does not have enough information to select which slot to request. In such an event, a node will issue a request with an assist flag. The requestor then relies on the neighbor(s) to propose back an assignment to use. A slot request made as part of a join message is treated with priority critical by the receiving neighbors. Critical requests can preempt an existing slot allocation. This mechanism enables a new node to gain ownership of a true NRT slot in a timely manner and as a result, ensures a rapid joining process.

The *UP* state is the desired end state. It is the normal state of operation. When a node enters this state, it is part of an organized network. This means the node has at least one neighbor and has gained ownership of at least one true NRT slot. Once in this state, a node will begin to regularly exchange information with its one-hop neighbors by transmitting Management Information (MI) messages. MI messages are control messages that carry all the necessary information in support of the protocol operations. Nodes periodically report information such as the slots they are using for transmission, the slots on which they are available to receive and NRT load demands. As this information is propagated, nodes build up and maintain their view/knowledge of the network (over a 2-hop neighborhood). This information is compiled and used to guide their decisions and assist them in the slot allocation procedure.

The MI message is sent once per cycle and always included as part of an NRT transmission (whether in a true NRT slot or in a RT-DP slot that the node uses for NRT transmissions). For robustness as well as to help in the node joining/network merging process, the slot assignment in which a node transmits its MI message changes periodically during operations. The assignment is picked amongst all the NRT slots (true NRT or temporary NRT) owned by the node.

After joining an organized network, each node will acquire and maintain a minimum number of true NRT slots (specified by policy). This not only guarantees the NRT traffic with a minimum bandwidth value but it also ensures that a node always has opportunities to transmit its control traffic. As in the join process, a node with less than its minimum true NRT slot allotment value will issue NRT requests with priority critical.

Once the nodes have acquired their minimum true NRT slot allocation, the remaining slots (NRT and DP-RT) are shared on-demand. This is performed via a fully distributed dynamic slot request and release scheme as described in the following section.

C. The dynamic slot allocation scheme

Nodes gain slot assignments by making requests. The protocol supports two types of slot requests: RT and NRT.

The mechanism that arbitrates the ownership of additional NRT (true NRT and temporary NRT) assignments attempts to respect the principle of fair share. Slots are divided amongst the nodes according to their relative NRT traffic needs (NRT traffic queued for transmission). To support this, each node reports, as part of every MI transmission, information on the NRT load. Essentially, each node includes the following information: the node's local slot demand and the sum of all of its neighbors' slot demand. A node makes use of its local NRT load information as well as the NRT load information reported by its neighbors to compute a target slot allocation. When preparing to send an MI message, a node evaluates whether it should try to gain additional NRT slots or release some. Typically, a node whose slot allotment is less than its computed target allocation will try to request additional slots. Alternately, a node whose slot allotment is greater than its computed target allocation will release slots. This highly dynamic NRT request and release scheme provides the system with a true adaptive bandwidth on-demand capability which guarantees maximum channel utilization.

While slot-request information is explicitly carried inside an MI message, slot releases are performed implicitly. This is to keep the signaling overhead as low as possible. Nodes simply remove the released assignment from their reported slot ownership information.

When a node has a voice flow to send, it needs to reserve the slots of a RT-DP column. It does this by issuing an RT request. As opposed to the NRT requests which are always destined to everyone, RT requests support both unicast and broadcast destinations. Since RT traffic always has precedence over NRT traffic, RT requests will naturally result in the preemption of temporary NRT slots if necessary.

The protocol also supports priority and preemption between RT flows. Four RT priority levels are currently supported. As part of its MI messages, a node will report the priority values of the RT sessions it takes part in (either unicast or broadcast). The reported priority information is used to arbitrate between RT requests and may lead to RT preemption decisions when the maximum capacity of the network has been reached (maximum RT capacity ratio has been attained). RT slot reservations are valid for the duration of the corresponding voice flow/session. As in the case of NRT, RT slot releases are implicit and will be performed as soon as the voice session terminates.

Nodes select the slots to request based on the distributed slot information they maintain. To support this, a node reports as part of every MI transmission, information on slot ownership. Essentially, each node includes: the set of all slots the node owns for transmission (*tx set*) and the set of all slots the node is available to receive on (*available for rx set*). To keep a low overhead, this slot information is expressed using a 3-bit slot vector where each bit represents respectively: the tx set, the available for rx set and the type (NRT or RT).

The slot selection process ensures that nodes request non-conflicting transmission allocations over a 2-hop neighborhood (hidden node problem) while taking advantage of simultaneous transmissions whenever possible (exposed node problem). It also achieves spatial slot re-use when nodes are at least 3-hops apart. The algorithm used to perform the selection of the slot to request depends on the type of the request (NRT or RT). The process is more complex for the RT request type than it is for NRT. The RT priorities, the support of both unicast and broadcast destinations, as well as the fact that RT slots may temporarily be used by NRT transmissions are factors that contribute to the increased complexity of the RT slot selection algorithm.

To achieve greater efficiency, the slot selection algorithm can also offer the slot merging optimization option. Slot merging allows for greater data carrying capacity by saving on the slot overhead. Slot merging applies to the NRT slot selection only. When enforced, the algorithm will favor the selection of slots that are adjacent to the ones already owned by the node.

A node will not take ownership of a requested slot until all of its neighbors have had a chance to respond. The request will thus be pending for a maximum of 1 cycle duration. If no neighbor objects, the requesting node will verify if approval can be claimed. The condition to claim approval of a request is dependent on whether the request is a unicast or a broadcast destination. In the case of a broadcast request, if after 1 cycle, the approval from any of the neighbors is missing, the node will still consider the request as approved. In the case of a unicast request, the approval from the destination node must be received (within the cycle) to declare the request approved.

While objections to slot requests are explicit, approvals are implicit, i.e., as part of its MI message, a node indicates approval by removing the requested slot assignment from its reported *available for rx* information set. This procedure reduces the management information overhead.

Because of mobility (topology changes) and of possible missed transmissions (error-prone links), the information maintained at each node may temporarily diverge, giving rise to potential conflicts. The protocol addresses this problem by specifying a comprehensive conflict management scheme. Conflict detection and resolution is performed every time a node receives a MI message from one of its neighbors. A comparison against the node's local information is conducted for each of the reported fields included in the MI message. Verification is performed on fields such as the slot ownership information, the slot requests, the slot objections and slot preemption information. For each of the above, the protocol defines a number of conflict resolution rules. A description of these rules is beyond the scope of this paper.

THE PHYSICAL LAYER

As a part of an ongoing effort to develop high performance VHF/UHF military tactical communication waveforms, a specific type of Orthogonal Frequency Division Multiplexing (OFDM) modulation was investigated and developed. This variation of OFDM is designated as Pilot Equalized Coherent OFDM (PE-COFDM). It incorporates refined and adapted state-of-the-art techniques for its modulation and demodulation. Some of the advanced techniques include: robust multi-tone preamble for coarse and fine timing/frequency offset correction [7]; phase scrambling and automatic level control to minimize peak to average power ratio [8]; and equidistant pilot tones with spectral interpolation for accurate synchronous channel estimation and tracking [9][10].

Specific to PE-COFDM, a high coherent stability is obtained by means of its synchronous training pilot tone technology. In the same transmission, a waveform may contain symbols that use different numbers of sub-carriers leading to symbols with different periods. Symbols having as few as 16 carriers are used to minimize preamble overhead while symbols bearing 64 to 512 carriers are typically used to carry data more efficiently.

There is little published work regarding the exploitation of coherent OFDM in VHF/UHF narrow to mid-bandwidth (25-200 KHz) military tactical communication systems. In this work, PE-COFDM waveforms for VHF/UHF that support data rates ranging from 9600 bps to 2.0 mbps and in bandwidths of 25, 50, 100, 200 and 350 kHz have been designed and tested on air. Promising performance results have been obtained.

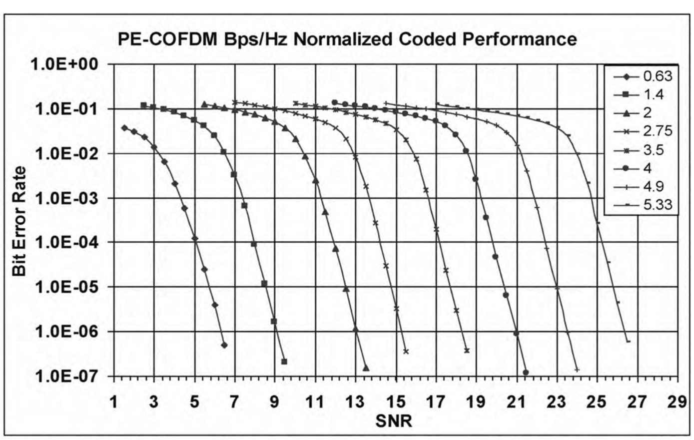

Figure 5: Coded SNR performance vs Spectral Efficiency

Figure 5 shows measured Signal-To-Noise (SNR) coded performance for a 100kHz waveform over an Added White Gaussian Noise (AWGN) channel. The waveform operates in bursts ≥ 8ms. The unit data symbol period is 2.5 ms and contains a coded block size of 200 to 1500 bits. A dual terminated serial concatenated turbo code rate (7,8), K=4 is used. The waveform includes a random interleaver limited to a depth equal to one symbol. The performance curves (left to right) in Figure 5 correspond to symbols mainly containing BPSK, QPSK, 8PSK, 16-QAM, 32QAM, 64QAM, 128QAM and 256QAM respectively.

A very flexible architecture to implement PE-COFDM waveforms was developed. Any number of symbol constructs can be designed. Every individual sub-carrier of every symbol can be assigned a specific constellation, scaling factor and scrambling phase. Each symbol can be set to carry sub-groups of data. For instance, a data symbol may carry auto-baud information along with a transmit power index and actual data payload. Symbols can be assigned specific error correcting codes, puncture masks and interleaver schemes. Symbols can be sequenced according to requirements.

Such flexibility was achieved through the design of a highly parameterized waveform definition model referred to as Software Defined Waveform (SDW). SDW essentially represents a complete waveform construct in the form of parameter lists that can be saved to a file. The SDW model is currently limited to OFDM based waveforms. A complete OFDM SDW set of primitives and processing tools was developed to enable an experienced waveform designer to define, compile and package new waveforms into files. The files are then stored in a repository and are loaded into the modulator/demodulator as required. Another important feature of the SDW is the ability to design/modify, compile and load a waveform very quickly. Once loaded, the waveform is ready for testing and characterization.

Although PE-COFDM can accommodate variable length point-to-point transmissions, it was mainly designed to satisfy modern network requirements. In particular, since the MATRIQS MAC scheme includes support for real-time data, it is necessary to pair it with a PHY layer that can meet the constraints associated with this type of data. When real-time data is carried, MATRIQS operates using short TDMA slot periods, resulting in repeated quick short transmissions bursts at the physical layer. Consequently, minimal processing delays and minimal waveform overhead are a prerequisite. The low processing complexity of PE-COFDM (characteristic of OFDM modulation) and the flexibility of its waveform design satisfy these requirements. This makes PE-COFDM particularly well suited for integration with the MATRIQS MAC protocol. In addition, the highly parametric and programmable attributes of the PE-OFDM and its SDW framework, combined with the ability of the MAC to operate at various bandwidths and data rates make it possible to readily investigate a multitude of radio and network configurations.

PROTOTYPE IMPLEMENTATION

In order to evaluate the technologies and validate the approach, a prototype of the system presented in this paper was developed. The complete RIU stack was implemented on a General Purpose Processor (GPP) platform equipped with Intel quad core technology. The analog input/output components are Commercial-Off-The-Shelf (COTS) PCI cards. The RF components (radio front-end) are VXI COTS modules and cover the VHF/UHF bands including a wide range of RF bandwidths.

For initial testing, a testbed composed of 5 tactical MANET nodes was assembled. Various tests were carried out in the lab over multiple network scenarios. A mix of RF attenuators and splitters was used to achieve the various topology configurations. The system was operated with PE-COFDM waveforms of 100 and 200 kHz bandwidths. GPS feeds were used to provide the systems with an accurate time source. The voice application used a MELPe 2.4 kbps codec.

The system was tested to verify the proper support of concurrent real-time services and data. While various data traffic (such as Web images, ftp and multicast SA) loaded the network, multiple voice sessions were dynamically established between the nodes. Both, unicast and multicast sessions were tested. The natural pre-emption of the NRT traffic by the RT flows as well as the adaptive NRT bandwidth on-demand capability were both successfully observed. Scenarios to verify the self-forming/self-configuring capability of the system were also conducted. This initial lab testing showed promising results. A complete performance characterization of the system together with on-air testing is planned for the near future.

CONCLUSION AND FUTURE WORK

The RIU presented in this paper was developed to provide advanced networking capabilities that address the new tactical communications needs. On one hand, the MATRIQS protocol permits the deployment of masterless, self-organizing multi-hop IP networks with support for real-time services. On the other hand, the PE-COFDM provides increased capacity by adapting state-of-the-art parallel waveform techniques to the narrow to mid-bandwidth VHF/UHF area. The two technologies operate within a flexible framework that conforms to SDR principles.

Although the work presented in this paper mainly focused on the two lower layers of the RIU, another key feature of the system resides with the integration of a complete protocol stack which provides a full abstraction capability to the network and application layers.

The next step is to undertake in-depth testing studies to evaluate the system in more dynamic environments and with an increased number of nodes. This will allow to assess the scalability of the system as well as to obtain performance results when exercised in more complex realistic scenarios.

It is also desired to carry out further investigation on extending the soft QoS support to include a DiffServ-like scheme. Provisions to support such a scheme are already part of the system (namely via the queuing strategy at the link layer and via the NRT bandwidth on-demand scheme at the MAC), but more work needs to be done to determine the possible impact of incorporating this capability on the MAC operations.

Finally, it is planned to make use of the cross-layering information exchange between the MAC and the PHY layers to increase the adaptiveness of the system to its environment. In particular, a dynamic rate adaptation capability would be implemented by making use of the modem's knowledge on link quality estimation.

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