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A Network Layer Protocol for UANs to Address Propagation Delay Induced Performance Limitations*

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***Abstract*- This paper provides a description of a novel network layer protocol for underwater acoustic networking (UAN) that provides a mechanism for network control and management enabling the implementation of responsive, self-configuring, adaptable, and scalable networks whose performance are predictable. The protocol draws from the demonstrated efficiencies of multi-protocol labeled switching, dynamic source routing, and multi-constraint based resource allocation schemes. The paper describes the expected benefits of establishing full duplex functionality between network nodes and presents some of the preliminary simulation findings regarding the viability of autonomously determining the network topology utilizing the full duplex node connections.**

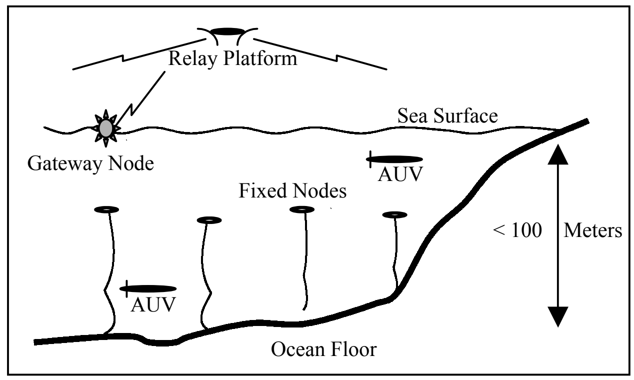

Fig. 1 Nominal Underwater Acoustic Network

I. INTRODUCTION

The effective implementation of underwater acoustic networks is inhibited by several key characteristics of the shallow water channel. Many of these issues are adequately addressed at the physical and data link layers of the network protocol stack leading to advances in acoustic telemetry modem technologies. These include selective frequency absorption, temporal spreading, Doppler induced frequency distortion, and multiple-propagation-path induced inter-symbol interference. However, the extreme signal propagation delays inherent in acoustic communications remain a significant problem to be addressed by the network layer protocol. While simple handshake based protocols (e.g., IEEE 802.11 [Kurose, 2000]), implementing collision avoidance multiple access techniques, provide a straightforward method of sharing the channel medium between multiple users, they tend to exacerbate the propagation delay problem. The unpredictable delays encountered during session establishment between neighboring nodes, and magnified as the session data packets travel across multiple hops, severely limits the ability of the network to support timing-critical, delay-sensitive applications. The effect of these delays is most keenly experienced during the setup stage of data communications. Until their effects are mitigated, the full potential of such applications as remote sensing, autonomous underwater vehicle and remotely operated vehicle command and control, positioning and navigation, and localization and tracking may not be fully realized. While improvements in signal recovery at the physical level, and forward error detection and correction at the data link layer, are providing improvements to the data reception efficiency, in terms of reducing the number of unrecoverable bit errors, the effective control of extreme signal propagation delays remains a major challenge in the design of a network layer protocol for UANs.

In pursuing this challenge, we are developing a protocol that explores a more pro-active method of autonomously establishing the network topology, managing network resource allocation, and controlling the flow of traffic through the network by a central network manager. Figure 1 depicts the target network environment consisting of a collection of tethered or otherwise fixed location network nodes. These nodes communicate with each other, as well as with transient autonomous underwater vehicles, across acoustic links. Both the fixed and mobile hosts are able to communicate to a remote command facility across a relatively high-speed relay platform, either airborne, satellite, or radio-based. The relay platform is interfaced to the acoustic network by a gateway node, a hybrid node containing both acoustic and radio modems and the buffer capacity to interconnect the two. The main component of the protocol described herein is a manager at the gateway node. The manager implements network management and routing agents

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which periodically probe the network for node status, extracting channel characteristics from the node responses, allowing the manager to determine a priori the most efficient delivery path between network nodes, actively avoiding network congestion, and providing a mechanism for guarantees of quality of service for high priority application traffic.

Simplex connections compound the effects of propagation delay on data transfer latency by requiring the exchange of several control packets to establish media access. The establishment of full duplex connections between each pair of nodes provides a means of assured access to the media without the exchange of access requests prior to each traffic exchange session. Professors Larraza and Smith, of the Naval Postgraduate School, are currently performing a series of experiments to demonstrate the viability of establishing this capability. Their initial results, although very preliminary, indicate that an acoustic node is able to send and receive on separate channels simultaneously, even though the receive signal level is several orders of magnitude less than the signal it is concurrently transmitting. [Smith, 2001]

One of the main objectives of this network layer protocol design is to take full advantage of this emergent capability. The full duplex capability, when implemented by an auto-configuring mechanism, may require a larger allocation of channels than if the network connections were pre-established. Since these channels share the available bandwidth, any channel not actively assigned as a unidirectional path between two nodes is not contributing to the data carrying capacity of the network. If those channels can be allocated on demand in such a way that they would not interfere with communications between neighboring nodes then the carrying capacity between nodes may be tailored to support the varying needs of the network nodes.

The rest of the paper is organized as follows. Section II discusses the fundamental aspect of network layer protocols – enabling effective traffic routing – and contrasts proactive and reactive routing schemes. Section III then describes an alternative solution that draws on the strengths of both while seeking to avoid their weaknesses. Section IV addresses the importance of full duplex connections to the pursuit of increasingly complex and higher throughput acoustic networks. Section V presents some early simulation results demonstrating the feasibility of autonomously establishing an acoustic network's topology by use of controlled flooding of control messages through the network. Section VI proposes a scheme for utilizing the unallocated channels to provide on-demand quality of service support. The paper concludes, in Section VII, with a brief description of planned future work.

Data Type	**Application**
Numeric Data	Sensor readings, position information, AUV speed, etc.
Text Data	AUV tasking commands, auto-configuration messages, etc.
Imagery	Low resolution or monochrome. JPEG images
Streaming Data	Video or audio

Fig 2. Anticipated data types of UAN applications

II. NETWORK PROTOCOL CONSIDERATIONS

It is reasonable to assume that an underwater acoustic network may be required to carry various data type, as depicted by Figure 2. Some UAN applications, such as remote sensor monitoring, generate short messages intermittently while others, like charting and mapping or video, require sustained throughput for an extended period of time. Most applications are sensitive to end-to-end message delay. Some are also sensitive to message delay variation, or jitter. To meet application requirements, the network layer protocol should provide at least three types of service characterized respectively by: low average message delay, predictable message delay, and sustained data throughput. An application may request one or more of these services. The network protocol must also use each channel's bandwidth efficiently and balance the load over all channels to avoid unnecessary congestion.

This efficient use of the network capacity is a key characteristic of a network protocol, the main functions of which are the efficient routing of traffic within the network and the provision of an interface between local network hosts and hosts on external networks. In order to route traffic the current network topology must be known, either to each individual host or centrally with the necessary routing information made available to the individual hosts as required. Therefore, it is important for a network layer protocol to have an efficient topology discovery and update method. Today's wired Internet primarily uses Open Shortest Path First (OSPF) or Routing Information Protocol (RIP) methods to establish the route information. [Kurose, 2000]

Current wireless routing methods (AODV, CBRP, DSR, etc.) have serious drawbacks with respect to their suitability to underwater acoustic networks. [Ramanathan, 1996, Broch, 1999, Perkins, 2000, Boukerche, 2000] Two key characteristics distinguish them – when and by whom routes are determined.

Proactive, or pre-computed routing, techniques seek to minimize route discovery induced message latency by establishing routes prior to the generation of traffic requiring them. Existing proactive routing schemes typically are based on fully distributed link state or distance vector algorithms, such as OSPF and RIP, or similar methods for establishing routes. These methods introduce significant overhead, as each node must determine the most appropriate route to every other node in the network. Changes in the topology can introduce a surge of overhead as the network attempts to cope with those changes. Many established routes, though, may never be required, as some node pairs may never need to communicate, and the overhead expended maintaining those routes is wasted. The totally distributed approach to routing has a more fundamental limitation. It may be difficult, in particular for networks with dynamic topologies, to provide quality of services in a most resource efficient manner when each node makes independent routing decisions. [Xie, 1998]

When a session is requested between two nodes in a *reactive* or ad hoc network, the source node normally floods a route determination message to its neighbors, and on through the network until a feasible route is found. This route information is returned to the source and traffic is then forwarded along the provided route. Dynamic Source Routing is one such method. The advantage of reactive routing is that routes will be determined only for nodes needing to communicate with each other. Routes for nodes that do not have traffic to exchange will not be computed. A critical drawback is that delay is introduced for the first packet of a session for which the required path is not already known. Further, if a route fails additional delay is incurred while an alternate path is found.

For air-propagating networks the delays introduced while determining routing information may not be significant. However, the relatively slow propagation speed of sound in water presents a significant barrier when messages must be exchanged to determine routing information before a packet may be sent. Requiring each node to maintain complete or at least partial network topology information can impact the router's performance.

A third possible routing method calls for static routing. As in proactive routing, the routes are predetermined, however with static routing, no mechanism is in place to automatically respond to host failures in the network. A network administrator must maintain all route information manually. Obviously, this mechanism is limited in its ability to support either highly mobile or autonomous networks. A new method is necessary which is able to capture the advantages of current network protocols and yet mitigate the adverse delay experienced in acoustic signaling. We are currently investigating the feasibility of incorporating full duplex connections in underwater acoustic networks and, as a part of that investigation, the adaptation of gateway centric network management procedures into a hybrid network protocol.

III. OVERVIEW OF THE PROPOSED PROTOCOL

The described network layer protocol manages the network topology through a central server, or gateway node. The risk normally attributed to using a central server is already assumed by the necessity of a communications gateway, as shown in Figure 1, to link the acoustic network to the external control infrastructure. The functionality added to the gateway, or master node, enables it to dynamically manage the topology of the network and establish paths through the network between any pair of network nodes, and support the allocation of bandwidth on demand to ensure guaranteed levels of service are satisfied.

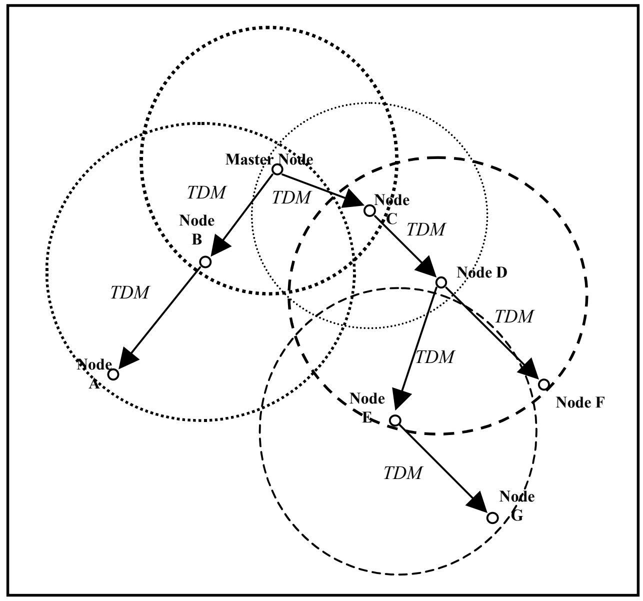

Fig 3. Topology Discovery Message (TDM) Propagation. (Circles represent the propagation radius about a given node.)

Central to the management of the network is the master node's ability to discover the topology of the nodes that comprise it. The topology discovery is accomplished by the transmission of a probe by the master node to its nearest neighbors, as depicted in Fig 3. The transmit level of the probe is set to a predetermined signal strength to constrain the range of the probe. Upon receipt of the probe, the neighbors append their identification to it and relay it to the next "ring of nodes," so that the probe propagates outward from the master like rings in a pond.

In addition to the names (integer IDs) of each node the probe has traversed from the master node outward, the probe includes the channel allocation for each of the known neighbors of the last node relaying it. This allows the most current recipient to select a channel from those not already allocated. In this way the contentions between neighboring nodes for channel allocations are minimized. However, some conflicts may occur as nodes receiving the topology discovery probe from the same source (their mutual parent node) may randomly select the same channel on which to respond. Should this happen the parent must send a conflict resolution message to all but one of the nodes in contention causing them to select another channel. The introduction of a small random queuing delay of response transmissions may mitigate the potential that two or more nodes respond simultaneously to a probe, should they have randomly chosen the same response channel.

To limit the number of times the topology probe is relayed, a node will only respond to a received probe if it is the first probe that node has received. This is verified by inspecting the list of traversed nodes contained in the probe for the recipient's ID. If the node had received the probe previously its ID would be in the list. This tracking of node IDs also prevents formation of routing cycles as the probe transits the network.

When a node relays the probe it sets a timer to control how long the node waits for a response. If the timer expires before a responding probe is detected the node resends the probe at an increased signal level, and resets the timer. In both cases, the duration of the timer is determined by the expected round-trip propagation time for a message sent at the transmitted strength. If the timer expires a second time without the node receiving a response then the node assumes that it is a leaf, or border, node and initiates the topology completion notice.

The topology completion notice is returned to the master node along the same route the topology discovery probe was propagated. As the notice transits each node in the route the node appends the information regarding its neighbors it has garnered from the discovery probes it has received. In this way, when the completion notice arrives at the master node it will contain the information necessary for the master node to manage traffic sessions across the network. This information may be used by the master node to establish path information connecting each pair of nodes in the network, allocate capacity across paths to meet quality of service constraints, establish a set of paths to expedite small, or high priority traffic, and support the movement of mobile nodes within the network.

The topology discovery and channel allocation process may be repeated, controlled internally by a timer or remotely by a network administrator, to enable the network to adapt to route failures induced either by node mobility or the fluidity of the underwater environment. A key design goal of the proposed protocol is that service received by normal data not be significantly degraded during a TDM cycle. Consequently, topology discovery may be repeated in relatively short intervals to adapt to route failures in a timely fashion, without the need for a "ping" type, traffic-intensive failure detection scheme.

IV. SIGNIFICANCE OF FULL DUPLEX CONNECTIVITY

While the International Standards Organization (ISO) model for networks differentiates the functions performed in support of communications between interconnected nodes into seven distinct layers, the implementation of those basic functions by specific protocol stacks often blurs those distinctions. This is no less the case in acoustic networks where the protocol stack must account for media access and resource allocation, flow control, and traffic routing.

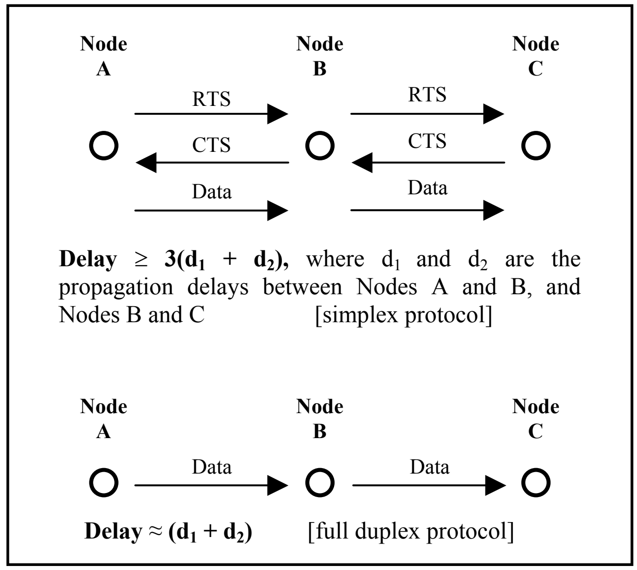

Fig 4. Message delay due to protocol

The degree of duplexing a connection supports has a strong impact on the mechanism used to control access to the network. Figure 4 illustrates the effect of simplex connections on average data delay vice that of a duplex connection. The net result of limiting the network to simplex connections is to dramatically increase the delay imposed on traffic while connections are coordinated. The determination of the connection also impacts the flow control mechanism that can be utilized by the network.

The handshake approach to media control, requiring the exchange of request-to-send and clear-to-send protocol packets to gain authorization to transmit data packets, in it's most basic form constitutes a Stop-and-Wait flow control mechanism, given that only one data frame is allowed per handshake exchange. If multiple frames are authorized for a single exchange, then the protocol implements the Sliding Window mechanism for flow control.

Bit errors are inevitable for any physical medium. The problem is particularly severe for underwater acoustic channels [Curtin, 1993, Sozer, 2000]. Therefore, a data link layer protocol is required for the proposed system to ensure reliable delivery of computer data between the underwater platforms. The most important performance metric for a link layer protocol is the achievable channel utilization. Consider a channel with a raw bit rate of 10 Kbps. If the link layer protocol could utilize the channel only 10% of the time, the effective data throughput over that channel would be just 1 Kbps!

The achievable utilization of a link layer protocol (denoted by U) depends on several system parameters. These are the raw bit rate of the channel (denoted by C), the data frame size in bits (denoted by F)[1], the one-way signal propagation latency (denoted by t_{prop}), and the uncorrectable frame error rate (P). Three link layer

[1] For ease of presentation, we assume a fixed frame size.

$$U \le \frac{1-P}{1+\frac{2t_{prop}}{(F/C)}}$$

Eq. 1 Channel Utilization

protocol candidates bear consideration for the proposed system: Forward Error Correction, Stop and Wait Automatic Repeat reQuest (ARQ), and Sliding Window.

The first protocol would add Forward Error Correction (FEC) code to each data frame so that the receiver can correct almost all bit errors in the frame. The transmitter sends out data frames continuously, and as rapidly as possible. The receiver may either process incoming frames at the same rate as they are received or buffer some of the frames for later processing. The main advantage of this protocol is that it works with a **unidirectional channel**. It has been demonstrated that Hamming codes perform well for an underwater acoustic channel [Reimers, 1995]. The principal disadvantage is that it is not 100% reliable, limiting its utility for transmitting critical textual or numerical data. Even for still-image data that may not require 100% reliability, a small number of uncorrectable bit errors may degrade the quality significantly when the image is compressed or should the errors cluster about the portion of the image of greatest interest to the user. FEC codes also incur communication overhead as they incorporate redundancy in the data. For this reason they have a negative impact on the attainable data throughput. For example, if each frame's FEC code were one quarter of the total frame size, then the actual data throughput would never exceed 75% of the channel capacity, even if the channel utilization approaches 100%. For a specific reliability target, such as limiting uncorrectable frame error rate below 1%, the required FEC code size increases as the bit error rate of the physical medium rises.[2] Since the bit error rate in our system is expected to be high, the communication overhead of FEC codes can be a major problem.

The second protocol of interest, Stop-and-Wait ARQ, does not explicitly require an FEC code, although one may be used to reduce the uncorrectable frame error rate (P). Error control is predominantly implemented by way of retransmissions. The receiver must acknowledge to the sender the receipt of each data frame that is free of errors. It may also send a negative acknowledgement for an incoming frame with uncorrectable errors. The sender cannot send a new data frame until the current frame is positively acknowledged. Upon receiving a negative acknowledgement or in the absence of positive acknowledgement within a specific time period, the sender will retransmit the current frame. This protocol would induce severe delay penalties on acoustic systems.

However, it has two major advantages. First, the protocol can achieve **100% reliability**. Second, it works with a **half-duplex (simplex) channel**. The main drawback, predominately due to the delay exacerbation, is that it may not be able to achieve high channel utilization in some system environments. Specifically, the achievable channel utilization of Stop-and-Wait ARQ, according to [Stallings, 2000], is bounded by Equation 1.

Assume that the channel capacity C is 20 Kbps and t_{prop} is 0.67 seconds.[3] As P approaches 0 the utilization is maximized for a given frame size (F). From the formula above, the frame size (F) would have to be larger than 134,000 bits in order to achieve channel utilization larger than 83%. Given the high bit error rate in our system, this frame size may be too large to assure a small P value. From the same formula, a large P value would clearly have a negative impact on U. This problem would be exacerbated if either C or t_{prop} increased.

The third protocol bearing consideration is the Sliding Window protocol, which improves on Stop-and-Wait ARQ and is the basis of the popular TCP protocol. Specifically, the sender may send multiple frames at a time as long as the total number of unacknowledged frames is not larger than a particular window size. The window size is determined either by the buffer space at the receiver or by the number of outstanding frames the receiver can track [Balakrishnan, 2001], as well as other system parameters. The receiver acknowledges the frames that it has received correctly. These acknowledgements will reduce the number of unacknowledged frames and prompt the sender to transmit new data frames. The sender will also retransmit a frame for which there is no acknowledgement within a given period of time. The main advantage of the Sliding Window protocol is that it can achieve **100% reliability** and **very high channel utilization**, specifically ($1-P$), without requiring a large data frame size [Stallings, 2000]. The principal disadvantage is that it can only work with a **full duplex channel**. In other words, there must be a back channel for the receiver to send acknowledgements to the sender while the sender is transmitting data frames.

Therefore, to mitigate the impact of the acoustic propagation delay and assure high reliability, full duplex functionality needs to be provided to support the use of sliding-window type link layer protocols in conjunction with an efficient routing protocol.

V. PRELIMINARY SIMULATION RESULTS

[2] The required FEC code size also jumps when the correlation of bit errors increases.

[3] Time it takes a sound wave to travel 1 kilometer underwater [Sozer, 2000].

The proposed topology discovery process was evaluated using a first-order simulation. The baseline distance between nodes was set to 1 kilometer and the transmit level set to allow propagation to just exceed that range. The baseline topology was modeled as a hexagonal honeycomb, with a node at each vertex. A set of nodes were positioned so that their closest neighbor was beyond the baseline distance, but within the extended range used to relay the probe after the first non-response time-out. The first-order simulation does not encompass mobile nodes, however, that function is planned for the second-order model. Initial results support the conclusion that a fully autonomous channel allocation process can be utilized to establish and manage a network topology. In addition, each node is allocated an exclusive channel for transmission of infrequent short messages.

For a 30-node network, arranged in 5 rows of 6 nodes, the time required to complete a discovery cycle was approximately 14 seconds, just slightly longer than the round trip time between the master node and the node most distant from it. Figure 5 summarizes the results for several simulation runs depicting networks of varying sizes. Preliminary analyses indicate that all channel contentions were resolved during that period.

The density of the node locations impacted the number of distinct channel allocations necessary to support the nodes, as evidenced when an outlying node retransmitted the discovery probe at the increased signal strength. The impact was to cause its signal to be acquired by nodes beyond its shortest neighbor distance. While the protocol was able to resolve channel allocation conflicts when this occurred, the effect was to increase the number of channels required to establish the full duplex connections beyond the nearest neighbor considerations to the number of potential neighbors within two baseline propagation distances. In can be seen from Figure 6 that channel assignments can lead to contentions either if neighboring nodes select the same channel or if nodes which have a common neighbor select the same channel, as shown in the leftmost frames of the figure. In either case the contention must be resolved so that each has its own dedicated channel. Re-assignments must be done to eliminate the adverse effect of contentions. The resolution must be coordinated so that only one of the two conflicting nodes changes its channel else both might attempt to change to a different channel and select identical channels resulting in another contention. This results in an increase in the number of channels necessary to establish the network connections, limiting the capacity of each individual

Number of Nodes in Test Topology	16	30	60
Topology Discovery Time (seconds)	15.4	14.1	18.1
Number of Channels Required	6	7	7

Fig 5. Preliminary Simulation Results for Hexagonal Grid

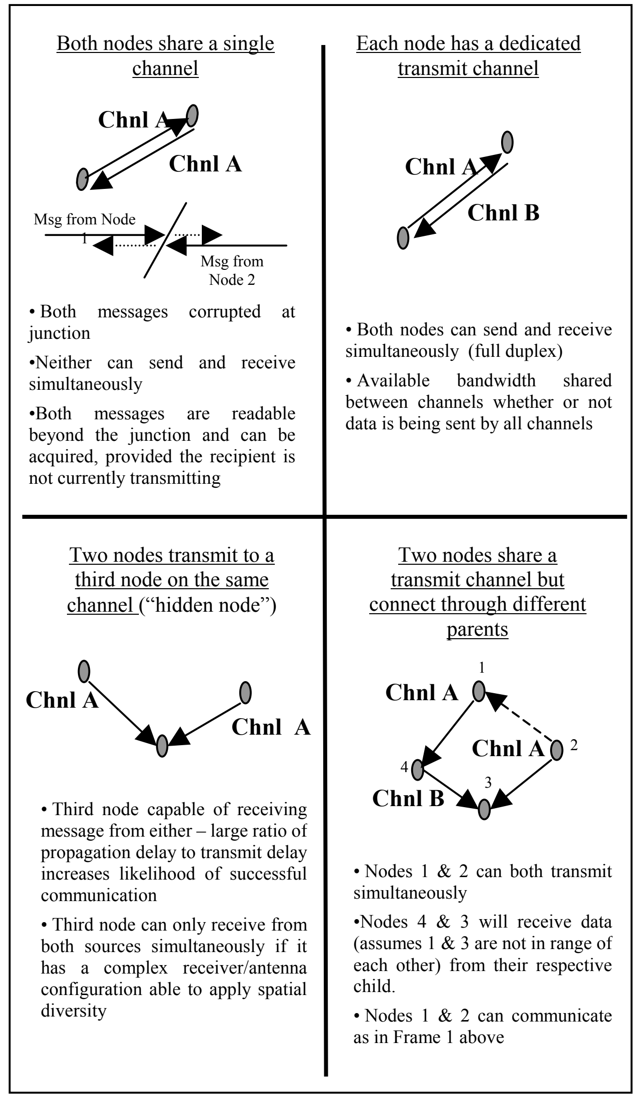

Fig. 6. Channel Allocation Contentions

channel. As the channel structure within the network is reported to the master node it is conceivable for the master node to allocate the unused capacity within local node clusters to support on-demand or guaranteed service levels. Not all contentions may require resolution, however. In the case where two nodes share the same channel but do not need to communicate with each other, and none of the nodes that one communicates with can hear the other's transmission, the conflict may be ignored without adversely impacting the network operation.

VI. ON-DEMAND BANDWIDTH ASSIGNMENT

The channel allocation process considers all nodes within reception range of each other as neighbors. As the allocation results in a dedicated transmit channel for each network participant, and each node knows the channel allocations for it's one- and two-hop neighbors, it is possible for nodes to reserve unallocated channel capacity to support increased data transfer or sustained throughput requirements for high demand applications. Such a scheme may be expanded to provide guarantees for quality of service for key or high priority users or applications. The assignment scheme must address the requirement to reconstruct or re-sequence the data packets received by a node from multiple channels. This requirement may be inhibited by the multiple access scheme incorporated.

The allocation may be performed centrally by the master node or by another designated node. Conversely, it may be accomplished using a decentralized method that allows each node to control the flow of traffic across its immediate neighborhood. The advantage of controlling the allocations centrally is the ability of the master node to redirect traffic or balance requests for resources across the entire network. The disadvantage is the delay encountered to request and establish the reservation across the network. The requests for additional bandwidth beyond that acquired by the single channel allocation during the discovery process are adaptations of the resource reservation protocols being generated in support of IPv6 development. [Feit, 1998] A key distinction is that the requestor need not be limited to the destination – the source may request the additional resources from the master node thereby reducing the control traffic required to set up the session.

To ensure no node conflicts with its siblings or the children of its neighbors, the number of channels required to allow complete configuration of the network is driven by the number of nodes within the most dense neighborhood of the network. This means that the less dense areas will have fewer of the channels actively assigned. These excess channels may be used to support service quality negotiations.

The proposed process of allocating the available channels using a decentralized approach is as follows:

- Node A requests bandwidth allocation from its parent, Node B
- Node B selects the required channels to support A's request from those not currently allocated and forwards those channel IDs to Node A
- Node B advertises the allocation to its neighbors to ensure the channels are not subsequently allocated by a neighbor to one of its children until Node A releases them
- Node A forwards his data until either the session is completed or he receives a message rescinding the allocation from Node B.

For a centralized scheme the requestor merely forwards its request to the master node across the same path as it received the most recent topology discovery message. In response to the request, the master node provides it a path consisting of a sequence of nodes and channel allocations through which to establish the session. Each node in the path forwards traffic received to the next node in the path. The master node must provide each node in the path with the channels which form its receive and transmit links for the path so that it can process the traffic accordingly.

The purpose of the bandwidth sharing is to recover the network capacity rendered idle by the number of channels necessary to support complete network configuration, but which are not required within a limited neighborhood. While the establishment of increased capacity sessions requires additional control traffic it should be noted that sessions within the capacity of the single channel allocation can be expedited without the session overhead and controlled by the sliding-window mechanism discussed earlier.

VII. CONCLUSIONS

The preliminary results of the full duplex experimentation are encouraging. Much remains to be done to increase the fidelity of the simulation and more work remains in gathering and analyzing the simulation results. The total number of nodes comprising the network places increased demand on the allocation of channels but not linearly. However, the average node density has a larger impact than the total number of nodes. The actual impact of node mobility on the protocol convergence also remains to be determined, requiring further study and analysis.

Currently, the ability to support on-demand channel allocation is purely conceptual. Additional work remains to develop an algorithm by which to implement that ability. Then the functionality must be integrated into the simulation and its utility and performance analyzed.

Finally, in-situ testing must also be performed to demonstrate the performance of the protocol under realistic conditions.

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