Adaptive Reliable Transport Protocol for Cognitive Radio Mobile Ad Hoc Networks

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Abstract— A novel transmission control protocol, ARTP for multi-channel multi-radio multi-hop CRAHNs is presented in this paper. The channel availability is uncertainty in CRAHNs. An adaptive coding scheme in ARTP is used according to wireless environment, which can provide a better performance for multi-hop CRAHNs. ARTP can dynamically choose the number of packets involved in network coding operation. The ARTP picks the broadcast channel carefully and adapts itself according to the current network characteristics in order to guarantee successful broadcast operation without interfering with the PU communications. Performance metrics such as success rate, average broadcast delay are taken into account for comparison. Simulation results portrays that the proposed ARTP outperforms in terms of the chosen performance metrics.

Key words: Explicit Congestion Notification, Modular Clock, Cognitive Radio, multi-channel, multi-radio, multi-hop

I. Introduction

Cognitive radio is a variety of wireless communication in which a transceiver can cleverly detect which communication channels are in use and which are not and right away move into vacant channels while avoid occupied ones. This optimizes the use of available Radio-Frequency range while minimize interference to other users. In its basic form, CR is a hybrid technology concerning Software Defined Radio as applied to spread spectrum communications probable functions of cognitive radio include the capability of a transceiver to resolve its geographic location, identify and authorize its user encrypt or decrypt signals, sense nearby wireless devices in operation and change output power and modulation characteristics. Cognitive radio is the enabling technology for supporting DSA: the policy that addresses the spectrum scarcity problem that is encounter in many countries. Thus CR is regarded as one of the most hopeful technologies for future wireless communications. To make radios and wireless network truly cognitive, however, is by no way a simple task, and it requires collaborative effort from different research communities, with communications theory, networking engineering, signal processing, game theory, software-hardware design, and reconfigurable transmitter and radio-frequency design. In this paper, we provide a logical overview on Cognitive Radio networking and communications by looking at the key functions of the Physical, Medium Access Control, and networking layers involved in a Cognitive Radio design and how these layers are crossly related. In exacting, for the PHY layer, we will address signal processing technique for spectrum sensing, supportive spectrum sensing, and transceiver design for cognitive spectrum access: for the medium access control layer, we analysis sensing scheduling scheme, sensing-access exchange design, spectrum-aware access MAC, and Cognitive Radio MAC protocols. In the network layer, Cognitive Radio Network tomography, spectrum-aware routing, and QoS control will be addressed. Emerging CRNs that are actively developed by various consistency committee and spectrum-sharing economics will also be review. Finally we point out some open questions and challenges that are related to the Cognitive Radio design.

II. Literature review

Currently, only two papers investigate the broadcast issue in CR ad hoc networks [1], [2]. However, both papers adopt impractical assumptions, which make them inadequate to be used in practical scenarios. In [1] and [2], the global network topology and the available channel information on all SUs are assumed to be known. Additionally, in [2], a common signaling channel for the whole network is employed, which is also not practical. Although some studies on single-hop channel rendezvous for CR ad hoc networks can be used for finding a common channel between two nodes [3]–[7], they suffer various limitations, which make them incapable to be used in broadcast scenarios. In [3], a channel-hopping mechanism based on quorum systems is proposed for guaranteed rendezvous. However, this mechanism requires the knowledge on the common available channels between two nodes, which is often obtained via broadcasts. In [4], two channel-hopping schemes, namely, the Generated Orthogonal Sequence (GOS) algorithm and the Modular Clock (MC) algorithm, are proposed. However, GOS only works in a scenario where two users have the same available channel sets, while MC cannot guarantee rendezvous under some special circumstances. In addition, in [5], the proposed channel-hopping scheme cannot guarantee rendezvous in a finite time. In [6], a jump–stay based channel-hopping algorithm is proposed for guaranteed rendezvous. However, the expected rendezvous time for the asymmetric model (i.e., different users have different available channels) exponentially increases when the total number of channels increases. Thus, it is unsuitable for broadcast scenarios in which short broadcast delay is usually required. Other channel-hopping algorithms explained in [7] require tight time synchronization, which is also not feasible before the control information is exchanged.

III. Proposed work

A novel transmission control protocol, ARTP for multi-channel, multi-radio, multi-hop CRAHNs is presented. Different from conventional protocols that make use of all the packets in the congestion window for network coding. In ARTP, coding window in the TCP layer is added. Thus, the [8] TCP layer maintains two windows: TCP window and coding window. The sender module receives original packets from the TCP source and delivers them to the coding window. After receiving a certain number of original packets, the coding window generates a linear combination of the packets in the coding window and delivers them to the IP layer.

The receiver module also maintains a buffer of linear combinations of packets that have not been decided yet. Upon receiving a coding packet, the receiver module first retrieves the coding coefficient from the packet's header and appends it to the basis matrix which stores each received packet's coding coefficient. Then calculates the rank of the matrix as follows: if the rank increase by 1, then the receiver [8] module constructs an ACK packet and sends it to the source, otherwise, the receiver will discard this packet and return to wait for another incoming packet. If the value of rank received by the sender is smaller than generation size, it means that some coded packets are lost or the corresponding ACKs are lost. In the case of corresponding ACKs are lost, the retransmission is unnecessary. To prevent this, ARTP scheme is adopted. The encoding algorithm is shown below.

Algorithm 1 Encoding algorithm.

k : Generation size, the number of packets in each generation. The sequence is from Start to End.

diff_recvACK_new: the difference in the number of the received ACK for this period of time.

diff_recvACK_old: the difference in the number of the received ACK in the previous period.

R: redundancy factor.

rank: the number of packets the receiver has seen in current generation.

1) $Start \leftarrow 0, End \leftarrow k, NUM \leftarrow 0, NUM \leftarrow NUM + R + 1$
2) while $\lfloor NUM \rfloor > 0$ do
3) Generate an encoded packet, which is a random linear combination of the original data packets from the current generation.
4) $NUM \leftarrow \lfloor NUM \rfloor - 1$
5) end while
6) if rank $< End - Start$ then
7) if $diff\_recvACK\_new - diff\_recvACK\_old \geq 0$ then
8) Generate an encode packet, which is a random linear combination of the original data packets from the current generation to the next generation.
9) end if
10) if $diff\_recvACK\_new - diff\_recvACK\_old < 0$ then
11) Retransmit $((End - Start) - rank) \times (1 + R)$ coded packets which are random linear combination of the original data packets from the current generation.
12) end if
13) end if
14) if $rank = End - Start$ then
15) $Start \leftarrow End, End \leftarrow End + k$
16) end if

In Algorithm 1, the changing ratio of ACK is used for the condition of coding scheme in ARTP. Hence, the number of the received ACK over a certain time T_{cal} needs to be calculated. In CRAHNs, PUs has the highest priority to use channels. SUs will give up the channels for PUs when PUs arrives. There is a new loss service interruption for SUs due to the arrivals of PUs. The more activities PUs has, the less time the SUs use to transmit data. As a result, the number of service interruptions and loss probability become larger; TCP throughput of SUs is also lower. In our scheme, we exploit an Explicit Congestion Notification (ECN) to notify the source node that a service interruption occurs, which is generated by the affected node. It distinguishes the interruption loss from traditional loss caused by congestion. To improve TCP throughput, the TCP window will not decrease over a service interruption. In sensing state, the sensing node does not know the status of successor/predecessor nodes. The existing wireless TCP protocol does not know the lower layer's behavior, thus, it still sends data packets. As a result, a lot of data packets have to be stored at the node preceding the sensing node.

The ARTP picks the broadcast channel carefully and adapts itself according to the current network characteristics in order to guarantee successful broadcast operation without interfering with the PU communications. The key for achieving an efficient broadcast that enables the coexistence of both PU and CR transmissions within specified vicinity is to apply strict control over the channel selection. In addition, a CR node with no data to transmit uses the same criteria to select the tuning channel that offers best connectivity for data reception. Moreover, it is likely that CR users in the transmitter's vicinity share the same channel availability; hence, adopting the same classification by all CR nodes in the network allows the nodes within a close geographic area to choose with high probability the same channel. Once a CR receives a packet, it undergoes the same procedure again to convey the data to its CR neighbors. The proposed protocol ensures perfect protection for the PU communication and guarantees a high packet broadcast ratio.

IV. SIMULATION SETTINGS

Number of SUs N	16
Number of PUs K	40
Number of Channels M	20
Side length of the simulation area L	10(unit length)
Radius of the sensing range r_s	2(unit length)
Radius of the transmission range r_c	2(unit length)
Number of selected channels n	1
The normalized PU arrival rate λ_p	0.5
The PU Packet length L_p	10 (times slots)
The probability of a successful transmission σ	1

Table 1:

V. RESULTS AND DISCUSSIONS

A. Number of Secondary Users Vs Success Rate:

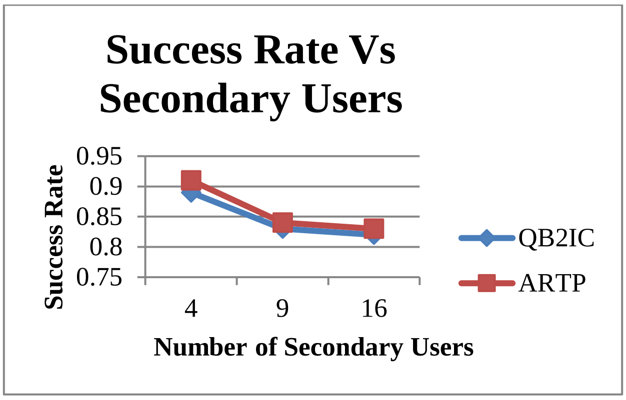

Fig. 5.1: Number of Secondary Users Vs Success Rate

Fig.5.1. Portrays the success rate performance subject to increasing the number of secondary users of the ARTP compared with QB2IC. It is evident that ARTP attains better success rate than that of QB2IC protocols. The simulation result values are shown in Table.5.1.

	Protocols	
No. of SUs	QB2IC	ARTP
4	0.89	0.91
9	0.83	0.84
16	0.82	0.83

Table 5.1: Number of Secondary Users Vs Success Rate

B. Number Of Secondary Users Vs Average Broadcast Delay:

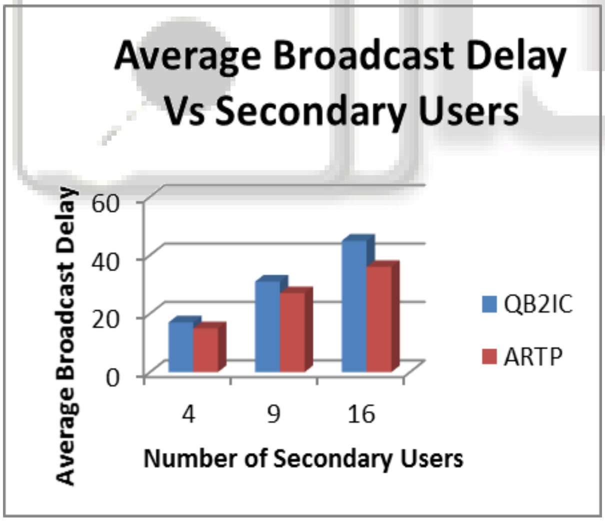

Fig. 5.2: Number of Secondary Users Vs Average Broadcast Delay

Fig.5.2. Projects the average broadcast delay performance subject to increasing the number of secondary users of the ARTP compared with QB2IC. It is obvious that ARTP attains less broadcast delay than that of QB2IC protocols. The simulation result values are shown in Table.5.2.

	Protocols	
No. of SUs	QB2IC	ARTP
4	17	15
9	31	27
16	45	36

Table 5.2: Number of Secondary Users Vs Average Broadcast Delay

C. Number of Primary Users Vs Success Rate:

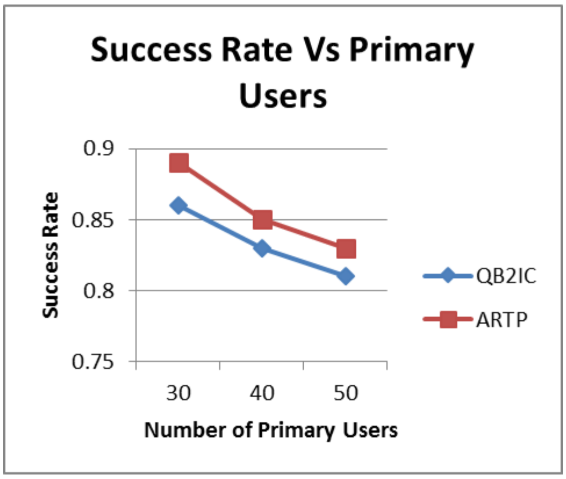

Fig. 5.3: Number of Primary Users Vs Success Rate

Fig.5.3. Projects the success rate performance subject to increasing the number of primary users of the ARTP compared with QB2IC. It is certain that ARTP attains increased success rate than that of QB2IC protocols. The simulation result values are shown in Table.5.3.

	Protocols	
No. of SUs	QB2IC	ARTP
30	0.86	0.89
40	0.83	0.85
50	0.81	0.83

Table 5.3: Number of Primary Users Vs Success Rate

D. Number of Primary Users Vs Average Broadcast Delay:

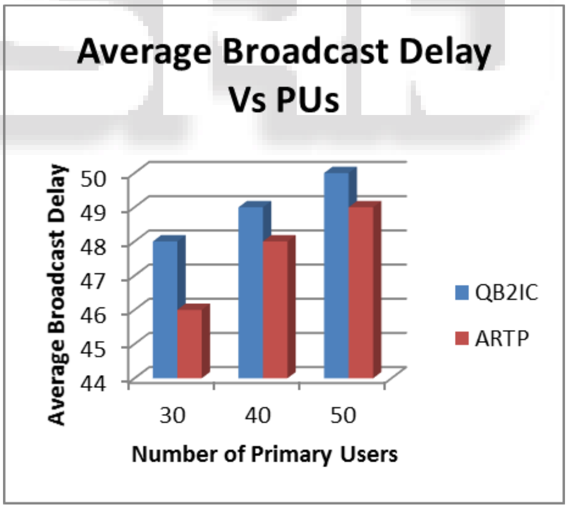

Fig. 5.4. Number of Primary Users Vs Average Broadcast Delay

Fig.5.4. showcases the average broadcast delay performance subject to increasing the number of primary users of the ARTP compared with QB2IC. It is evident that ARTP attains less broadcast delay than that of QB2IC protocols. The simulation result values are shown in Table.5.4.

	Protocols	
No. of SUs	QB2IC	ARTP
30	48	46
40	49	48
50	50	49

Table 5.4: Number of Primary Users Vs Average Broadcast Delay

E. Number of Channels Vs Success Rate:

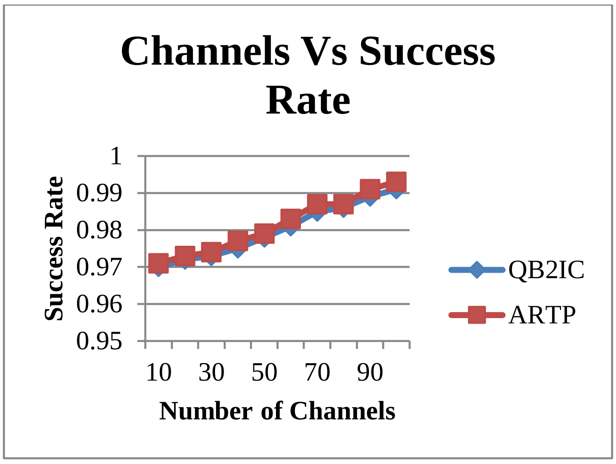

Fig. 5.5: Number of Channels Vs Success Rate

Fig.5.5. envisage the success rate performance subject to increasing the number of channels of the ARTP compared with QB2IC. It is certain that ARTP attains increased success rate than that of QB2IC protocols. The simulation result values are shown in Table.5.5.

	Protocols	
No. of SUs	QB2IC	ARTP
10	0.97	0.971
20	0.972	0.973
30	0.973	0.974
40	0.975	0.977
50	0.978	0.979
60	0.981	0.983
70	0.985	0.987
80	0.986	0.987
90	0.989	0.991
100	0.991	0.993

Table 5.5: Number of Channels Vs Success Rate

F. Number of Channels Vs Average Broadcast Delay:

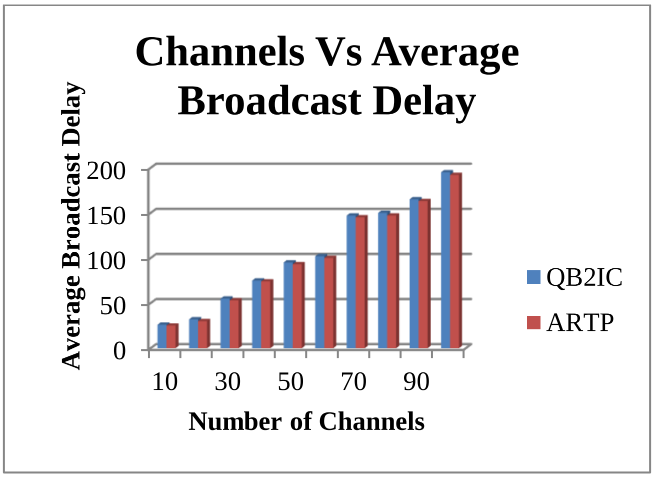

Fig. 5.6: Number of Channels Vs Average Broadcast Delay

Fig.5.6. projects the average broadcast delay performance subject to increasing the number of channels to the ARTP compared with QB2IC. It is obvious that ARTP attains less broadcast delay than that of QB2IC protocols. The simulation result values are shown in Table.5.6.

	Protocols	
No. of SUs	QB2IC	ARTP
10	26	25
20	32	30
30	55	53
40	75	74
50	95	93
60	102	100
70	147	145
80	150	147
90	165	163
100	195	192

Table 5.6: Number of Channels Vs Average Broadcast Delay

G. Number of Unsynchronized Time Slots Vs Success Rate:

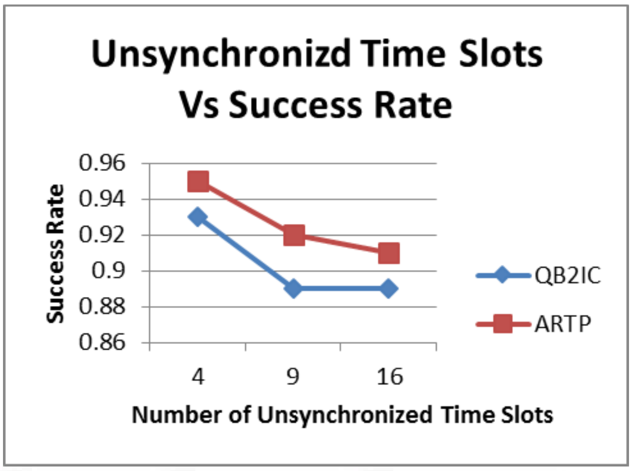

Fig. 5.7: Number of Unsynchronized Time Slots Vs Success Rate

Fig.5.7. envisages the success rate performance subject to increasing the unsynchronized time slots to the ARTP compared with QB2IC. It is certain that CARTP attains increased success rate than that of QB2IC protocols. The simulation result values are shown in Table.5.7.

	Protocols	
No. of SUs	QB2IC	ARTP
4	0.93	0.95
9	0.89	0.92
16	0.89	0.91

Table 5.7: Number of Unsynchronized Time Slots Vs Success Rate

H. Number of Unsynchronized Time Slots Vs Average Broadcast Delay:

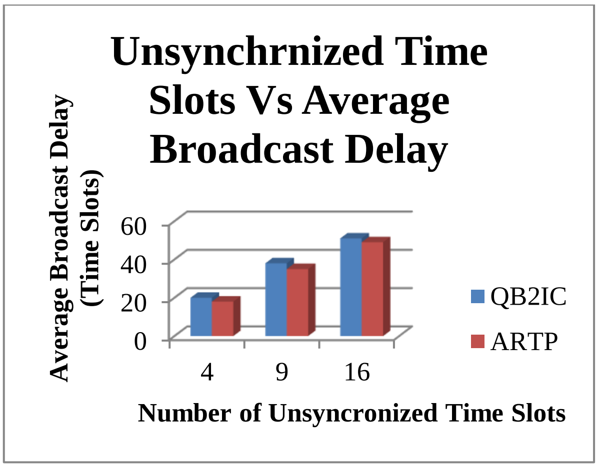

Fig. 5.8: Number of Unsynchronized Time Slots Vs Average Broadcast Delay

Fig.5.8. projects the average broadcast delay performance subject to increasing the number of unsynchronized time slots to the ARTP compared with QB2IC. It is obvious that ARTP attains less broadcast delay than that of QB2IC protocols. The simulation result values are shown in Table .5.8.

	Protocols	
No. of SUs	QB2IC	ARTP
4	20	18
9	38	35
16	51	49

Table 5.8: Number of Unsynchronized Time Slots Vs Average Broadcast Delay

VI. Conclusion and Future Work

This paper focuses on providing adaptive delay tolerant routing protocol for heterogeneous cognitive radio ad hoc networks. Simulations are carried out using cognitive radio cognitive network (CRCN) simulator. The performance metrics such as throughput, packet delivery ratio and delay are taken into account based on pause time. Simulation results prove that the proposed routing protocol ARTP has better performance in terms of increased throughput, better packet delivery ratio, decreased packet drop and reduced delay. The protocol can be further extended by incorporating security mechanism in the near future.

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