A Reliable and Scalable Broadcast Protocol for Wireless Multi-hop Networks using Subcarrier-level Tone-signals

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***Abstract*—In this paper, we propose a scalable broadcast protocol, named Subcarrier-level Tone-signal based Broadcast (ST-BCAST), that disseminates a packet over OFDM-based wireless multi-hop networks in an efficient and reliable manner. Exploiting collision-resilient tone-signals and receiver-triggered forwarding decision/cancellation, ST-BCAST achieves both high packet delivery ratio and low communication overhead without using any topological information, thereby providing scalability to the network size. Under a mild assumption, ST-BCAST satisfies two sufficient conditions for reliable broadcasting: first-hop delivery condition and successful relay condition. We verify the feasibility of tone-signal generation and detection through experiments using Universal Software Radio Peripheral (USRP) devices, and show through NS-3 simulations that ST-BCAST significantly outperforms the state-of-the-art broadcast schemes in terms of packet delivery ratio and communication overhead.**

I. Introduction

Network-wide packet broadcast is a communication service that aims to disseminate a packet from a source node to all the other nodes in the network. It has been widely used in wireless multi-hop networks for various purposes. For example, on-demand routing protocols such as AODV [1] utilize broadcast service to find a route from a source node to a destination node. The broadcast service is also used in vehicular ad-hoc networks for emergency message dissemination [2], and in wireless sensor networks for code update [3] and network-wide information dissemination [4].

A typical way of packet broadcast in wireless multi-hop networks is flooding: a node that receives a broadcast packet forwards it after some random backoff time [7]. The advantage of this method is its simplicity and robustness of the operation. However, it can cause numerous unnecessary retransmissions especially in dense networks, resulting in high degree of collisions and waste of resources [12]. In addition, it does not guarantee the packet delivery due to the lack of acknowledgment mechanism. Hence, the reliability and efficiency of the simple flooding scheme highly depend on network topology.

A number of protocols have been proposed for reliable and efficient network-wide packet broadcast. For reliable broadcast, most protocols require feedback messages from medium access control (MAC) or network layers to guarantee the delivery of broadcast packet to neighboring nodes [**?**], [8]–[11]. However, they are not scalable with node density since communication overhead for feedback message exchange increases with the degree of node connectivity. In addition, reacting to per-neighbor feedback can incur severe retransmission overhead since a node may have multiple neighbors with weak link condition. In this case, finding a different packet forwarder can improve the performance.

Regarding the efficiency, previous work can be grouped into three categories: counter-based schemes, probability-based schemes, and concurrent transmission based schemes. Under counter-based schemes [12], [16], [25], when a node receives a broadcast packet, it schedules its transmission after a random backoff time, and in the meantime, overhears neighbors' transmission for the packet. If the number of overheard transmissions is beyond a threshold, the node cancels its transmission, which helps to remove unnecessary transmission of the packet. However, the broadcast may terminate incompletely since overhearing multiple packet transmissions does not imply that all its neighboring nodes have received the packet successfully. Under probability-based schemes [16], [18], when a node receives a broadcast packet, it makes a forwarding decision with probability p. The value of p can be either the same for all nodes [12], [16] or different according to topological information. Unfortunately, the probability-based schemes do not guarantee complete delivery of the packet either, since a forwarder who is essential for reliable broadcast rejects to forward the received packet with probability $1-p$. Finally, concurrent transmission based schemes can improve the efficiency of broadcast by reducing the time for broadcast completion [19], [21], [22], but their reliability highly depends on the tightness of transmission time synchronization as well as the spatial distribution of nodes.

In this paper, we propose a scalable broadcast protocol that achieves both high reliability and efficiency without using any topological information in Orthogonal Frequency Division

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Multiplexing (OFDM) based wireless multi-hop networks[1]. The main contributions of this paper are in designing a receiver-triggered forwarding decision and a forwarding cancellation mechanism which are enabled by *subcarrier-level tone-signals*, leading to high reliability and efficiency. In the proposed scheme, a broadcast packet receiver notifies its reception by transmitting a Ready-To-Forward (RTF) signal, which is a 'subcarrier-level tone-signal'. Then, any of its neighbor nodes that have not received the packet yet requests packet forwarding by transmitting a Forward-Request (FR) signal, which is also a 'subcarrier-level tone-signal'. Since a receiving node can detect tone-signals simply by energy detection even when multiple nodes transmit them together, we can greatly reduce the communication overhead for control signal exchange. We check the feasibility of tone-signal generation and detection capability through experiments using USRP devices [28].

In order to verify reliable packet delivery of our Subcarrier-level Tone-signal based Broadcast protocol (ST-BCAST), we show that ST-BCAST satisfies two sufficient conditions, i.e., first-hop delivery condition and successful relay condition, for the completion of a packet broadcast under a mild assumption. We also show that ST-BCAST outperforms conventional broadcast schemes in terms of packet delivery ratio and communication overhead through extensive NS-3 simulations. While other comparable broadcast schemes have communication overhead that increases *linearly* with node density, ST-BCAST maintains a constant overhead beyond a certain level of node density and achieves scalability.

The rest of the paper is organized as follows. In Section II, we briefly describe the system model. In Section III, we explain ST-BCAST in detail. In Section IV, we analytically show the reliability of ST-BCAST. Section V demonstrates the feasibility of subcarrier-level tone-signal transmission and detection through experiments using USRP devices. In Section VI, we provide simulation results in comparison with other state-of-the-art broadcast schemes. Discussing related work in Section VII, we conclude our paper in Section VIII.

II. System model

We consider a wireless multi-hop network with a set V of wireless nodes. Each node has a half-duplex OFDM transceiver [20], [24]. The channel bandwidth is W, and the size of the Fast Fourier Transform (FFT) window (i.e., the number of subcarriers) is N_{FFT}. Among N_{FFT} subcarriers, we use N_S ($< N_{FFT}$) subcarriers for transmission of data symbols or tone-signals. We assume that all nodes transmit with the same power and data rate.

Each node detects transmission of a packet or a tone-signal by measuring the energy on the channel spectrum. Specifically, we assume that subcarrier-level energy detection is possible: if node $v \in V$ transmits a tone-signal on a subcarrier j, then the nodes located within the transmission range of node v detect the signal transmission and identify subcarrier j. This subcarrier-level tone-signal transmission and detection capability has been shown to be feasible [8], [14], [15].

For an interference model, we use the additive interference model that reflects the packet reception behavior in realistic wireless links [13]. Suppose that node $v \in V$ transmits a frame to node $w \in V$. Then, the SINR perceived by node w is

$$SINR_w = \frac{P_{vw}}{N + \sum\limits_{k \in V \setminus v} P_{kw}}, \tag{1}$$

where P_{vw} is the received power of node v's transmission at node w, N is the thermal noise floor, and P_{kw} is the interference experienced at node w due to the transmission of node k. A receiver can receive a frame with probability that is determined by the perceived SINR.

For wireless channel access, every node operates Carrier Sense Multiple Access / Collision Avoidance (CSMA/CA) mechanism [23]. In CSMA/CA, if a node detects that the channel is idle for a predefined long inter-frame space ($LIFS$), then it transmits a frame immediately. Otherwise, it attempts to transmit it after some random backoff time.

Finally, the broadcast source generates packet P(i) with sequence number i, which is used as the unique identifier of the packet in conjunction with the source address. In Medium Access Control (MAC) layer, each node encapsulates packet P(i) with a broadcast MAC frame header for forwarding. All its neighbors are assumed to be an intended receiver.

III. ST-BCAST

In this section, we describe our scheme, starting with important components.

A. Control signals

In ST-BCAST, two control signals are used for enabling the receiver-triggered forwarding and cancellation: Ready-To-Forward (RTF) signal is used by a node to advertise the possession of a broadcast packet; Forward-Request (FR) signal is used by a node to request the forwarding of it. Both signals take the form of a *tone-signal* on a specific subcarrier. Suppose that a node wants to advertise the possession of a broadcast packet P(i). Then, it transmits a tone-signal on data subcarrier j $(= i \ mod \ (N_S/2))$ which is the RTF signal for P(i), denoted as RTF(i). Similarly, a node that requests the forwarding of P(i) transmits a tone signal on data subcarrier j' $(= N_S/2 + (i \ mod \ (N_S/2)))$, which is the FR signal for P(i), denoted as FR(i). A node can identify the type of a control signal and the subcarrier number over which the tone-signal has been transmitted.

The benefit of using subcarrier-level tone-signals as a control signal is that the receiver can detect overlapped transmissions of a control signal correctly by measuring the energy level on each data subcarrier. Since the energy detection does not require any decoding procedure, the receiver can detect a control signal (i.e., a tone-signal) with high probability even when multiple nodes transmit it simultaneously in an

[1]Smart phone based ad-hoc disaster recovery network [5], [6], where emergency messages are frequently broadcasted, can be a good example application.

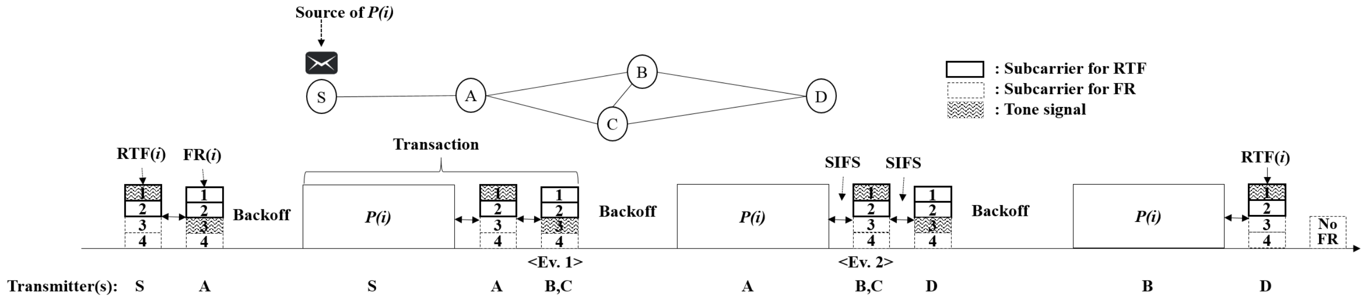


Fig. 1. An example of protocol operation when node S wants to flood broadcast packet P(i). In this example, the number of available subcarriers is 4.

asynchronous manner. We exploit this benefit of subcarrier-level tone-signals to reduce communication overhead for the forwarding transaction.

Note that the number of simultaneous broadcasts in the network, which maintains one-on-one relationship between the sequence number and its corresponding subcarrier, is limited to $N_S/2$. Hence, if multiple source nodes generate different broadcast packets *at the same time*, some of them may assign the same subcarrier number for their packets, resulting in a confusion. We address this problem in Subsection III-E.

B. Forwarding state lists

Each node maintains two lists for forwarding state information.

1) Forwarding Packet List (FPL): A broadcast source puts a broadcast packet into its *FPL* indexed with its sequence number. As well, if a node receives broadcast packet P(i) after transmitting FR(i), it puts the packet into its *FPL* to forward the packet when necessary. The broadcast packet will be held in *FPL* until the timer for that packet expires after *FPL_timeout*.

2) Requesting Packet List (RPL): This list contains the subcarrier numbers (after modulo operation) of the broadcast packets that the node is currently waiting for reception. The packet sequence number i is not known yet. When a node detects RTF(i) (i.e., on subcarrier j) but it does not have the subcarrier number j in its *RPL*, it adds j to its *RPL* and transmits FR(i) (i.e., on subcarrier j'). The subcarrier number j will be removed from *RPL* if the node receives the broadcast packet with sequence number i or the timer for subcarrier number j expires after *RPL_timeout* .

Note that each node maintains the state information independently. Since it does not share the information with its neighbors, there is no additional communication overhead except for using the two control signals RTF(i) and FR(i).

C. Receiver-triggered forwarding decision and cancellation

Suppose that source node s disseminates broadcast packet P(i). Then, ST-BCAST works as follows.

1) Source node s adds packet $P(i)$ to its *FPL*. After that, node s transmits RTF(i) as it already has the packet.
2) If a node detects RTF(i) but the subcarrier number j is not in its *RPL*, it transmits FR(i) after a predefined short interval (short inter-frame space, $SIFS$) and adds j to its *RPL*. If the node detects RTF(i) from another RTF(i) sender again, it transmits FR(i) after $SIFS$ and resets the timer for the subcarrier number j since it may receive the packet P(i) from a new RTF(i) sender.
3) If a node detects FR(i) and has packet P(i) in its *FPL*, it forwards the packet after a random backoff time $T_{backoff} \in [0, T_{max}]$. While the node is waiting for its transmission, if it detects RTF(i) before the backoff timer expires, it cancels the scheduled transmission for the packet.
4) If a node receives packet P(i) which corresponds to subcarrier number j in its *RPL*, it transmits RTF(i) after $SIFS$, and removes j from its *RPL*. Then, it adds packet P(i) to its *FPL*.
5) If a node cannot receive any broadcast packet with sequence number i after transmitting FR(i) within *FR_tx_interval*, it retransmits FR(i). If the number of retransmissions reaches a predefined number, it removes the subcarrier number j from its *RPL* and gives up the packet.

From Steps 2 and 3, the node that has broadcast packet P(i) forwards the packet *only if* there is an explicit request for the packet, i.e., on the reception of FR(i). This receiver-triggered forwarding decision prevents unnecessary transmissions of the packet. Furthermore, the forwarding cancellation mechanism in Step 3, which makes FR(i) receivers cancel their transmission schedule of packet P(i) after detecting RTF(i), reduces the number of redundant transmissions further since overheard RTF(i) often implies the reception of packet P(i) at the FR(i) sender.

What happens if a node cancels the transmission when some of its neighbors have not received P(i) yet? These neighbors will keep retransmitting FR(i) according to Step 5, prompting their neighbors to reschedule the transmission of packet P(i). Therefore, the forwarding cancellation after receiving RTF(i) does not harm reliable broadcasting.

In ST-BCAST, multiple neighbor nodes can transmit a control signal for the same packet concurrently. According to Step 2, nodes that receive RTF(i) will transmit FR(i) simultaneously. Similarly, according to Step 4, nodes that receive broadcast packet P(i) will transmit RTF(i) concurrently if they receive the packet at the same time. However, these overlapped transmissions of the control signals can be identified by receivers correctly thanks to the benefit of designing control signals as simple subcarrier-level tone-signals. Some unavoidable difference in their transmission timing can be ignorable by setting the length of $SIFS$ accordingly.

We highlight that the transaction of P(i)-RTF(i)-FR(i) can be completed without being interrupted by data transmissions of other nodes. Note that a node can transmit a data frame only when it detects the channel idle for $LIFS$, which is substantially longer than $SIFS$. However, a problem may occur in the presence of a hidden terminal since its data transmission can collide with P(i), RTF, or RF signals. The problem can be resolved partly by tuning the carrier sensing threshold as in [26]. The detailed setting of this parameter is beyond the scope of the paper. And we assume that the hidden terminal problem is negligible.

D. Operation example

Fig.1 depicts an example operation. In this example, the source has packet P(i) to broadcast, and N_S is 4. At the beginning, the source node S transmits a tone-signal of RTF(i) to advertise the presence of P(i) on subcarrier 1, assuming that $i \ mod \ (N_S/2) = 1$. Then, node A detects RTF(i) on subcarrier 1 and responds with FR(i) on subcarrier 3 ($= N_S/2 + i \ mod \ (N_S/2)$). Node S detects FR(i) and decides to transmit P(i) after a random backoff time.

Suppose that node A has received P(i) successfully. Then, node A responds with RTF(i) after $SIFS$. Nodes B and C receive RTF(i), and thus transmit FR(i) simultaneously (see <Ev.1> in Fig.1 where 'Ev' represents an event). Though the two signals overlap, node A can figure out FR(i) since it observes an energy peak only on subcarrier 3. Now, node A decides to forward P(i) after some backoff time. Then, nodes B and C respond with RTF(i) at the same time after receiving P(i) (<Ev.2> in Fig.1). In this case, node D recognizes RTF(i) by detecting an energy peak only on subcarrier 1, and responds with FR(i) making nodes B and C schedule the transmission of P(i).

Finally, suppose that node B picks up a smaller backoff time than node C. Then, node B transmits P(i), which is followed by RTF(i) from node D, and node C will cancel its transmission according to Step 3. After that, the broadcast of P(i) ends since no node transmits FR(i).

E. Mode switching mechanism for multiple source case

Using *subcarrier-level tone-signal* for the control signals makes the system scalable by allowing control signal transmissions to overlap. However, the limited number of data subcarriers can cause a problem when multiple sources generate different broadcast packets *at the same time*, since some of them may use the same subcarrier number for their packets. The broadcasts with the same subcarrier number j can cause a confusion at intermediate nodes, resulting in incomplete broadcast.

To alleviate this problem, we propose a simple mode switching mechanism. Suppose that two different source nodes v and w happen to broadcast their packets, denoted as $P_v(i)$ and $P_w(i')$, at the same time. If $i \ mod \ (N_S/2)$ is equal to $i' \ mod \ (N_S/2)$, both sources will assign a same subcarrier number j to their packets, which will be broadcasted using the same control signals, RTF(i) and FR(i). Therefore, if an intermediate node x detects RTF(i), it will transmit FR(i), and delete j from its *RPL* if it receives either $P_v(i)$ or $P_w(i')$.

Let us consider that node x has received $P_v(i)$ first. Then, it is possible for node x to receive $P_w(i')$ from one of its neighbors even though node x has not transmitted FR(i') for $P_w(i')$[2]. In this case, node x checks at its upper layer whether it has already received the broadcast packet with the sequence number i' from source w. If not, node x sets a bit-flag 'treat-as-flooding' in its MAC header and broadcasts $P_w(i')$ as the conventional flooding, i.e., forwards the packet after some random backoff time *without any tone-signals*. Any node that receives a broadcast packet with the 'treat-as-flooding' flag performs the same operation as node x. In other words, $P_w(i')$ will be broadcasted following the conventional flooding procedure when the duplicate subcarrier number has been detected.

F. Energy detection threshold of RTF and FR

Let R_{RTF} and R_{FR} denote the transmission range of RTF(i) and FR(i), respectively. Since a node detects a tone-signal by energy measurement, R_{RTF} and R_{FR} can be different from that of a broadcast packet, R_{DATA}. The difference between the transmission ranges can cause undesirable forwarding behavior. Suppose that $R_{RTF} = R_{FR} > R_{DATA}$ and consider two nodes v and w with distance r ($R_{DATA} < r < R_{RTF}$). When node v transmits RTF(i), node w will respond with FR(i) to receive P(i). Since $r > R_{DATA}$, the probability of successful transmission of P(i) can be very small, and thus multiple retransmissions of FR(i) and P(i) end in vain. We address this problem by setting the detection threshold of FR(i) such that R_{FR} and R_{DATA} are the same.

G. Duration of RTF and FR

When a node senses a transmission on the channel, it performs FFT using the received time-domain samples to extract spectral components. If the transmitted signal is a tone-signal, the node can identify the subcarrier number over which the tone-signal has been transmitted, by calculating the magnitude of the extracted spectral components. However, for the successful detection of a tone-signal, all the time-domain samples fed into FFT block (i.e., N_{FFT} samples in total) should contain tone-signal information [8]. Therefore, the duration of RTF and FR should be greater than N_{FFT}/W that denotes the duration of an OFDM symbol without cyclic prefix [24]. We conjecture that setting the duration of control signals as a multiple of N_{FFT}/W would be sufficient for robust detection.

[2]A neighbor that transmits $P_w(i')$ will have another neighbor that has transmitted FR(i') for $P_w(i')$.

IV. Reliability of ST-BCAST

In this section, we analyze the reliability of ST-BCAST. First, we introduce two conditions for reliable broadcast and show that the two conditions are sufficient for guaranteeing the complete delivery of a broadcast packet to all nodes in the network. Then, we show that ST-BCAST satisfies the two conditions assuming that RTF signals are detected without errors.

Notations and assumptions used in the analysis are as follows. We consider a wireless multi-hop network that is represented by a graph $G(V, E)$ where V is the set of nodes and E is the set of edges. The graph G is a connected graph, i.e., any node $v, w \in V$ can communicate with each other through a direct link or multi-hop routes. We assume that a source node s begins to broadcast packet P(i). We represent the set of l-hop neighbors of node $v \in V$ as $N_l(v)$.

First-hop delivery condition. *Suppose that the source node $s \in V$ transmits P(i). Then, all nodes in $N_1(s)$ receive P(i).*

Successful relay condition. *Consider node $v \in V$ that has not received P(i) yet. If any node $w \in N_1(v)$ receives P(i), then node v receives P(i) eventually.*

The two conditions are sufficient for the completion of the broadcast for P(i).

Proposition 1. *Suppose node s begins to broadcast P(i). If the first-hop delivery condition and the successful relay condition are true, then all nodes in $V\backslash s$ receive P(i).*

Proof. Let the maximum hop distance between node s and another node $w \in V\backslash s$ be L. We denote the entire set V as

$$V = \{s, N_1(s), N_2(s), \ldots, N_L(s)\}. \tag{2}$$

We use the mathematical induction to show that every node in $N_l(v)$ where $l \in \{1, 2, \ldots, L\}$ receives P(i). First, all nodes in $N_1(s)$ receive P(i) since the first-hop delivery condition is true. Suppose that all nodes in $N_k(s)$ receive P(i) where $1 \leq k \leq L-1, k \in \mathbb{N}$. Note that each node $x \in N_{k+1}(s)$ has at least one connected neighboring node in $N_k(s)$. Since we assume that all nodes in $N_k(s)$ have received P(i), all nodes in $N_{k+1}(s)$ receive P(i) by the successful relay condition. Therefore, the first-hop delivery condition and the successful relay condition are sufficient conditions for delivering P(i) to all the nodes in $V\backslash s$. □

Proposition 1 implies that any broadcast protocol that satisfies the first-hop delivery condition and the successful relay condition guarantees the reliable delivery of P(i) over the network. ST-BCAST satisfies the two conditions if the transmission of RTF(i) is correctly detected by neighboring nodes.

Claim 1. *Let node s be the source of broadcast packet P(i). If each node detects the transmission of RTF(i) correctly (and FR(i) with non-zero probability), then ST-BCAST satisfies the first-hop delivery condition and the successful relay condition.*

Proof. First, we prove that ST-BCAST satisfies the first-hop delivery condition. When node s transmits RTF(i), all nodes in $N_1(s)$ detect RTF(i) by the assumption. Thus, they add the subcarrier number $j(= i \ mod \ (N_S/2))$ to their *RPL*s and transmit FR(i). If node s detects FR(i), then it transmits P(i) after some backoff time. If node s fails to detect FR(i), nodes in $N_1(s)$ will transmit FR(i) again, making node s forward P(i).

Suppose that node s transmits P(i) after detecting $FR(i)$ from its neighbors in $N_1(s)$. If all nodes in $N_1(s)$ receive P(i), then they stop transmitting FR(i) and the first-hop delivery condition holds. Otherwise, some nodes in $N_1(s)$ that have not received P(i) transmit FR(i) periodically until they receive P(i). Since node s schedules transmission of P(i) whenever it detects FR(i), all nodes in $N_1(s)$ can receive P(i) eventually.

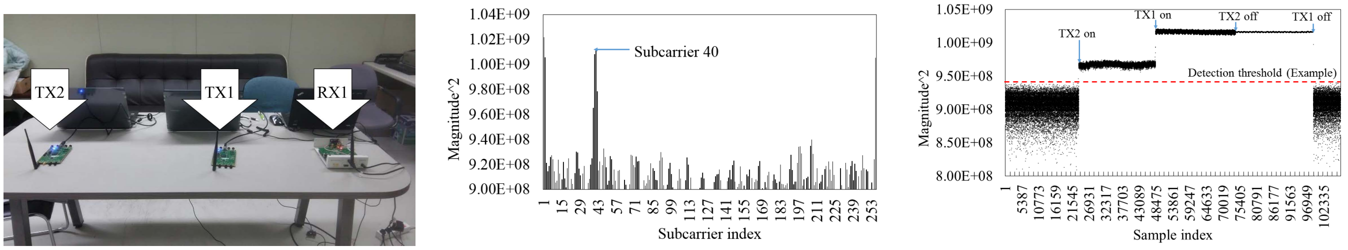
We now prove that ST-BCAST satisfies the successful relay condition. Consider node $v \in V\backslash s$ that has not received P(i). If node $w \in N_1(v)$ receives P(i) for the first time, it transmits RTF(i). Then we can consider node w as a new source node of P(i) that has transmitted RTF(i). Since node v belongs to $N_1(w)$, it receives P(i) by the first-hop delivery condition. Therefore, ST-BCAST satisfies the successful relay condition, which completes the proof. □

Previous feedback-based broadcast protocols that use the information of neighboring nodes can also satisfy the two conditions. However, they often consume a significant amount of communication resources to check per-neighbor feedback messages. For counter-based and probability-based protocols, the two conditions are not satisfied even when there is neither a channel error nor a collision, since a node that is essential for network connectivity may not forward the received packet.

In practical wireless multi-hop networks, Claim 1 may not be always true since RTF(i) can be lost for some reasons such as channel distortion. However, detecting RTF(i) will be more robust to channel fading and collisions than receiving P(i) directly, thanks to the energy detection technique. Additionally, there can be multiple chances for a node to receive RTF(i) if its neighboring nodes receive P(i) at different times, increasing the probability of recognizing P(i). Once a node recognizes the presence of P(i) by detecting RTF(i), it can transmit FR(i) for several times until it receives P(i).

V. Feasibility of tone signal detection

In this section, we confirm the feasibility of tone-signal generation and detection through experiments using USRP devices and GNU Radio software package [28]. We set up a simple testbed that consists of three hosts as shown in Fig. 2(a). One host acts as a tone-signal detector (RX1), and the other two hosts act as tone-signal transmitters (TX1, TX2). We consider an OFDM communication channel with 1 MHz baseband and 256 subcarriers. Thus, the frequency spacing between two adjacent subcarriers is 3906.25 Hz. The tone-signal detector (RX1) calculates the squared magnitude of frequency-domain samples, obtained from FFT with the time-domain complex samples.



(a) USRP testbed

(b) The squared magnitude of frequency-domain samples calculated at RX1.

(c) The change of the squared magnitude of frequency-domain samples on the subcarrier 40.

Fig. 2. Experimental results of tone-signal generation and detection.

Fig. 2(b) shows the squared magnitude of frequency-domain samples at RX1 when TX1 transmits a tone-signal on subcarrier 40. We observe that there is a peak around the subcarrier 40 as we expected. Subcarriers 0 and 255 also show peaks, but this is due to DC bias [8]. However, the adjacent subcarriers near the subcarrier 40 also show large values. This spectral leakage is known to be inevitable in practice since the time-domain samples of the observed signal is likely to contain any discontinuity at the end of the measurement time [15], [24]. Therefore, mapping a sequence number into a group of subcarriers will improve the accuracy of RTF and FR detection.

Next, we evaluate the impact of overlapped tone-signals on detection capability. Fig. 2(c) shows the changes of the squared magnitude of frequency-domain samples across time on subcarrier 40. Note that TX1 is closer to RX1 than TX2. First, only noise exists when both transmitters are inactive. When TX2 begins to transmit a tone-signal on subcarrier 40, values on that subcarrier increase suddenly. Since the values are greater than the detection threshold (the dotted red line in Fig. 2c), RX1 detects the tone-signal on subcarrier 40. After that, TX1 starts to transmit a tone-signal on the same subcarrier. We can see that the values increase further and do not decrease even when TX2 stops the transmission. Therefore, we claim that it is possible for a node to detect the overlapped transmissions of a tone-signal by measuring the energy of spectral components, and the received signal strength is dominated by the transmission from a node that is closer to the receiver.

However, the overlapping feature of the control signals may cause a side effect, since multiple simultaneous transmissions of a control signal can increase their transmission range. As we noted, the mismatch of transmission ranges of RTF and FR signals can lead to undesirable forwarding behavior. To this end, the dynamic control of transmission power is an interesting open problem.

VI. Simulation results

In this section, we evaluate the performance of ST-BCAST through simulations. We implemented ST-BCAST on NS-3 network simulator, and compared it with other broadcast schemes that operate without topological information: unconditional flooding, GOSSIP1(p), and GOSSIP3(p, k, m) [16]. The simulation results show that ST-BCAST outperforms the others in both reliability and efficiency.

TABLE I
Simulation parameters used for ST-BCAST

Parameter	Value
Energy detection threshold for CCA	-96.0 (dBm)
RTF detection threshold (*RTF_thres*)	-95.5 (dBm)
FR detection threshold (*FR_thres*)	-94.0 (dBm)
Forwarding packet list timeout (*FPL_timeout*)	2 (sec)
Requesting packet list timeout (*RPL_timeout*)	0.5 (sec)
FR transmission interval (*FR_tx_interval*)	0.2 (sec)
Duration of RTF (T_{RTF})	16 ($u$sec)
Duration of FR (T_{FR})	16 ($u$sec)
Duration of $SIFS$	9 ($u$sec)
Duration of $LIFS$	34 ($u$sec)
Number of data subcarriers (N_S)	48

A. Implementation

We implement ST-BCAST by modifying WiFi modules provided in NS-3.21 package [29]. We add RTF and FR signals to the set of MAC-layer control frames. A node can detect these signals if the received power is higher than RTF detection threshold (*RTF_thres*) and FR detection threshold (*FR_thres*), respectively. We set the energy detection threshold, *RTF_thres*, and *FR_thres* as -96 dBm, -95.5 dBm, -94.0 dBm, respectively. Setting *FR_thres* slightly higher than *RTF_thres* prevents nodes that have weak links with FR transmitters from forwarding packets. The simulation parameters for ST-BCAST are listed in Table. I.

B. Broadcast schemes in comparison

We compared ST-BCAST with the other three well-known broadcast schemes: flooding, GOSSIP1(p), and GOSSIP3(p, k, m). In the flooding scheme, if a node receives a broadcast packet, it forwards the packet after a random backoff time. In GOSSIP1(p), if a node receives the packet, then it forwards the packet after a random backoff time with probability p. GOSSIP3(p, k, m) extends GOSSIP1(p) in order to improve the coverage of broadcast. In GOSSIP3(p, k, m), the nodes within the first k-hop from the source node forward the packet with probability 1 if they receive the packet. The other nodes forward the packet with probability p or listen to the channel for timeout period T_{WAIT}. If they cannot overhear the transmission of the packet more than m times within

T_{WAIT}, they forward the received packet with probability 1. Note that all the schemes including ours do not require any topological information. We set the parameters of p, k, m to $0.7, 4, 1$, respectively, referring to [16].

C. Performance metrics

We use three performance metrics to evaluate the considered protocols.

1) Packet delivery ratio (PDR): The PDR is the ratio of the number of nodes that received the broadcast packet to the total number of nodes. In simulation runs, the achievable PDR can be less than 1 since some nodes can be isolated (i.e., disconnected) from the broadcast source after random deployment.

2) Communication overhead (CO): We define CO as the total sum of transmission times used for packet forwarding and signaling. Let T_P denote the transmission time for a broadcast packet, N_P denote the total number of transmissions for a broadcast packet over the network, and N_{FR} denote the total number of FR retransmissions for a packet. The overhead of all the comparable broadcast schemes can be calculated as $T_P N_P$. On the other hand, ST-BCAST consumes additional times for RTF and FR transmissions, which are denoted as T_{RTF} and T_{FR}, respectively. Thus, the CO of ST-BCAST is obtained by calculating $(T_P + SIFS + T_{RTF} + SIFS + T_{FR})N_P + T_{FR}N_{FR}$.

3) Packet delivery time (PDT): The PDT is the time spent until a packet is received by a node. Specifically, let $t_s^{P(i)}$ denote the time when source node s begins to broadcast packet P(i). If node v receives P(i) at time $t_v^{P(i)}$, then the PDT of node v for P(i) is $t_v^{P(i)} - t_s^{P(i)}$.

D. Simulation environments

We use the standard 802.11 DCF for the MAC protocol [23]. Each node transmits with the same rate (6 Mbps) and power (16.0206 dBm). The tone signal is also transmitted with the same power. The size of a broadcast packet is fixed to 64 bytes. In order to model practical wireless links, we adopt a propagation loss model that considers both large-scale path loss (log-distance path loss) and small-scale short-term fading effect (Rayleigh fading) [27]. Under these models, the transmission ranges of tone-signal and data symbols are about 100 m. Finally, we adopt the additive interference model.

We consider two simulation scenarios: 1) single broadcast source and 2) multiple broadcast sources. In the first scenario, a randomly chosen source node broadcasts 5 packets at a rate of 1 packet/second. In the second scenario, multiple source nodes broadcast 5 packets *simultaneously* at the same rate. The simulation stops when there is no node in the network with a transmission schedule of the last broadcast packet. In each simulation, nodes are randomly distributed within a 1000×1000 m^2 area. We set the backoff window size, T_{max}, to 100 ms[3]. We run simulations on 30 randomly-chosen topologies. The simulation results are averaged to get a bar or point. Error bars in the graphs represent the standard error.

[3]We may need to determine T_{max} accounting for the node density. In our simulation scenarios, we use 100 ms, which is sufficiently large and keeps the collision rate low at an acceptable level.

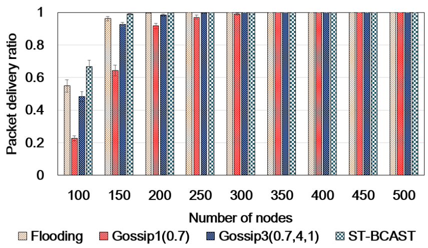
E. Results of single source scenario

Fig. 3(a) shows average PDR according to different node densities. In networks with high node density (i.e., $|V| > 250$), all the schemes achieve the PDR close to 1 since a majority of nodes participate in packet forwarding and redundant forwarding trials compensate for packet losses due to collisions or channel errors. However, as the network becomes sparser, ST-BCAST shows better PDR than the other broadcast schemes. As the degree of connectivity becomes smaller, the number of nodes that are essential for reliable packet delivery increases and losses in packet forwarding are likely to impair the PDR. In GOSSIP1(0.7) and GOSSIP3($0.7, 4, 1$), a node rejects to forward a received packet with probability 0.3. Hence, additional PDR drops are observed compared to the flooding scheme. In contrast, ST-BCAST suffers less packet losses, in part, because the node that has not received packet P(i) transmits FR(i) repeatedly once it detects RTF(i).

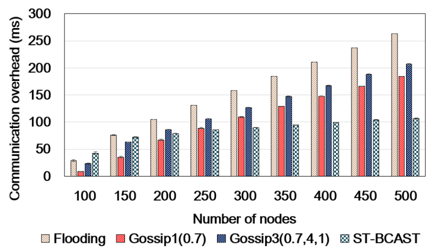
The advantage of ST-BCAST is more notable in the efficiency perspective. Fig. 3(b) shows CO according to different node densities. We can see that when the number of nodes is more than 200, ST-BCAST achieve the smallest CO among the compared schemes. As the node density increases, ST-BCAST successfully manages CO at a lower level and show scalable operations. In sparse networks (i.e., $|V| = 100$), ST-BCAST shows higher CO than others since most of forwarding attempts are essential to complete packet delivery. While GOSSIP1(0.7) and GOSSIP3($0.7, 4, 1$) have less CO than the flooding scheme, they suffer from low PDR.

We emphasize that under all the other broadcast schemes, CO increases *linearly* with respect to the node density. In contrast, ST-BCAST achieves a bounded CO regardless of the node density. In ST-BCAST, the number of nodes that decide to forward packet P(i) after detecting FR(i) increases as the node density increases, but most of them cancel their transmission schedules after overhearing RTF(i). In addition, as shown in Fig. 3(c), the number of periodic FR transmissions also decreases as the node density increases. As nodes have more chances of detecting RTF(i), the retransmission schedules of FR(i) are more likely to be delayed and often canceled after receiving P(i).

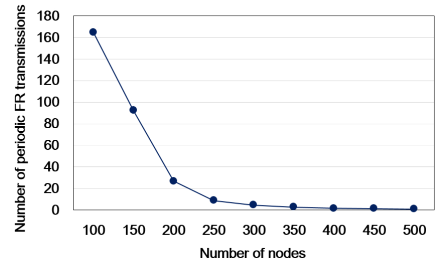
Fig. 3(d) shows average PDT over all the nodes in the network. We can see that every scheme shows decreasing PDT as the node density increases. In dense networks, the broadcast packet tends to be delivered over the shortest path, and packet forwardings tend to occur earlier due to the increasing number of forwarders with random backoff. However, in sparse networks (e.g., 100 nodes), ST-BCAST experiences larger PDT than other schemes. The reasons are two folds. First, the communication time for RTF/FR handshake is accumulated as the packet propagates. Second, ST-BCAST achieves higher PDR than the other schemes, implying that more nodes which are distant from the source node receive packets. Meanwhile,



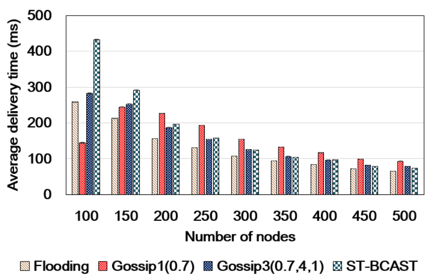
(a) Single source - PDR vs Node density



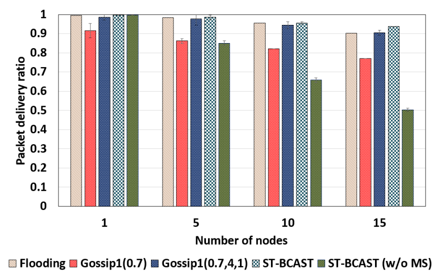
(b) Single source - CO vs Node density



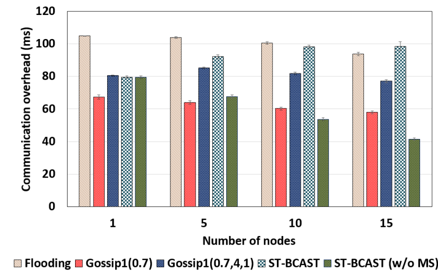
(c) Single source - Number of additional FR transmissions



(d) Single source - PDT vs Node density



(e) Multiple sources - PDR vs Number of broadcast sources



(f) Multiple sources - CO vs Number of broadcast sources

Fig. 3. Simulation results. In the scenario of multiple broadcast sources, the total number of nodes in the network is 200.

GOSSIP1(0.7) shows low PDT in the network of 100 nodes due to its low PDR.

F. Results of multiple sources scenario

Fig. 3(e) shows PDR of each scheme when the total number of nodes is 200 and the number of broadcast sources is 1, 5, 10, and 15, respectively. We set the number of data subcarriers to 48. First, we can observe that ST-BCAST always shows comparable PDR with the flooding scheme even when multiple nodes begin to broadcast *simultaneously* thanks to the proposed mode switching mechanism. Without the mode switching mechanism (ST-BCAST (w/o MS) in the graph), the PDR drops significantly as the number of concurrent broadcast sources increases. On the other hand, the mode switching operation increases CO as shown in Fig. 3(f). As the number of concurrent broadcast sources increases, the sources are likely to pick up a same subcarrier number for their broadcast packets, and ST-BCAST works as the conventional flooding.

VII. Related Work

The subcarrier-level tone-signal transmission and detection capability has been adopted in various works to improve the throughput of wireless LAN. In Back2F [15], the authors propose a frequency domain backoff mechanism where random backoff is realized by selectively transmitting a tone-signal on a subcarrier. In FICA [14], a fine-grained channel access scheme that enables subchannel-based concurrent transmissions has been proposed for improving the throughput of wireless LAN. ST-BCAST shares the benefit of subcarrier-level tone-signaling with Back2F and FICA, but ST-BCAST focuses on the reliability and scalability of broadcast, rather than throughput performance.

In SMACK [8], the authors introduce a subcarrier-level acknowledgment technique for providing reliable broadcast transmission in wireless networks. In this scheme, when a node broadcasts a packet, the receivers respond with a tone-signal on the subcarrier that is associated with the packet transmitter. Then, the packet transmitter checks the presence of a tone-signal (i.e., an acknowledgment) on each subcarrier and retransmits the packet if necessary. In broadcast, the arrival times of acknowledgment tone-signals can be used to select the farthest node as the next packet forwarder, which can prevent redundant retransmissions. However, SMACK is not scalable in dense networks since every node must keep the list of its neighbors, and negotiate the acknowledgment subcarrier with all of its neighbors. In addition, nodes using SMACK should maintain 2-hop neighbor information since the next packet forwarder is determined by the current packet forwarder.

The receiver-triggered packet forwarding mechanism of ST-BCAST is motivated by the SPIN protocol which is an application-level approach for disseminating information in wireless sensor networks [4]. In SPIN, when a node obtains new data, it sends an advertisement message to its neighbors. Upon receiving the advertisement message, a neighboring node responds with a request message if it has not received the advertised data. The advertisement message sender transmits the data if it receives a request message for the advertised data.

ReMHoC [9] also takes a similar approach for reliable multicast service in mobile ad-hoc networks. In this scheme, if a node detects the loss of a multicast packet through sequence number matching, it *multicasts* a request message that includes the sequence number for the missing data packet. If a multicast member that has the copy of the missing packet receives the request message, it responds by *multicasting* the cached copy. The SPIN-like approach has also been adopted for disseminating emergency massages in vehicular ad-hoc networks [2]. In this scheme, the emergency message sender selects one of its neighbors as a proxy for transmitting an acknowledgment frame. When this proxy responds with an acknowledgment frame, nodes that overhear this frame transmit a request frame as 1-hop broadcast if they have not received the emergency message with the sequence number included in the acknowledgment frame.

The mentioned schemes have drawbacks for scalable broadcast. First, they require topological information (i.e., the list of neighboring nodes) for the operation. Second, they do not support concurrent transmissions of signaling messages, resulting in excessive communication overhead. In contrast, ST-BCAST requires neither topological information nor association procedures. Furthermore, it provides scalability since the communication overhead for control signals does not increase with the node density thanks to the reliable detection of overlapped tone-signals.

VIII. Conclusion

In this paper, we have proposed a scalable broadcast protocol that achieves high packet delivery ratio with small communication overhead. In ST-BCAST, a node schedules the transmission of a broadcast packet only if it detects an explicit request for the packet from its neighbors. The node delays packet forwarding for a random backoff time and cancel the forwarding if it speculates that one of its neighbors has received the packet already by overhearing an RTF signal in the meantime. False forwarding cancellations can be recovered by repetitive requests from the neighbors. We verified the reliability of ST-BCAST by showing that it satisfies two sufficient conditions for the complete delivery of a broadcast packet. In addition, we verified the feasibility of subcarrier-level tone-signaling through simple experiments using USRP devices. Through NS-3 simulations, we showed that ST-BCAST outperforms existing broadcast schemes in terms of the reliability and efficiency. There remain many interesting open problems, e.g., setting carrier sensing threshold to alleviating hidden terminal problems, and designing dynamic transmission power control for the control signal.

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