Quality-Aware Voice Convergecast in Mobile Low Power Wireless Networks

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Abstract. Many disasters have shown the critical need for reliable voice communication for mobile users. Low power ad hoc wireless networks have become a promising solution in emergency scenarios because of their low cost and portability. In order to increase communication amongst moving emergency personnel and disaster victims, we have developed a novel convergecast voice streaming system that guarantees robust voice quality in a low power mobile wireless network. The system integrates routing and mobility-aware admission control along with voice compression adjustment to ensure the quality of voice streams. The system is evaluated using Arduino Due micro-controllers with XBee 802.15.4 radios. Our results show that our system can adequately adapt to changing network and routing conditions to deliver sufficient voice quality by maintaining a certain number of concurrent voice streams. To the best of our knowledge, this work is the first complete system for quality-aware voice streaming in mobile lower power wireless networks.

Keywords: Voice streaming · Low power mobile wireless networks · Admission control

1 Introduction

Today, voice communication is still the primary method for exchanging information in disaster response. Voice communication has obvious benefits: it requires no special training and is hands-free. In the event that users are injured or otherwise incapacitated, available voice communication is critically important for their rescue. We envision that future buildings will be instrumented with a number of low power Zigbee sensor nodes (a.k.a. motes) due to their portability and low cost. These motes will monitor environmental conditions such as toxic gas levels and temperature. During emergency, if cellular service is down, people (including first responders and victims during emergency response) may communicate with

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their low power wireless devices such as smartphones, smart helmets, or other wearables using low-power Bluetooth interfaces. A set of Zigbee/Bluetooth gateway nodes will be deployed to enable communication between people's phones and infrastructure motes. Servers may be deployed near the entrance to enable communication from the low power wireless ad hoc networks to the outside world. This paper focuses on voice convergecast only and leave other communication modes (i.e., boradcast, multicast, unicast) as future work.

Several key issues must be addressed to support voice communication in mobile, low-power, wireless meshed networks (MLWMNs). First, voice streams have relatively high data-rate requirements, but MLWMNs have limited bandwidth (e.g., typical Zigbee nodes only support 250 Kbps). Second, audio streaming over multi-hop LWN often results in unsatisfactory voice quality because of notable loss over low-power wireless links. The problem is exacerbated in scenarios where wireless nodes (i.e., people) are mobile. Third, simultaneously maintaining quality of multiple voice streams over lossy wireless links is difficult.

Our Contributions. We developed a novel quality-aware convergecast mobile audio streaming system (QACM) that maintains the quality of mobile voice streams in an ad-hoc low power wireless network. Specifically,

- We designed an integrated mobility-aware admission control and routing algorithm to ensure the quality of streaming audio in a mobile ad-hoc low power wireless network. Our algorithm guarantees quality of voice streams by choosing routes that maximize the number of audio streams in the network; adjusting routes in reaction to node mobility; minimizing channel contention; and avoiding bottlenecks.
- We implemented our end-to-end system QACM on an Arduino based hardware platform.
- We evaluated the system in both stationary and mobile scenarios.

2 Related Work

We discuss two areas of previous research that are closely related to this work: audio streaming over low power wireless sensor networks and audio streaming in mobile ad-hoc networks (MANETs).

Voice Streaming in Wireless Sensor Networks. Previous work developed audio streaming systems such as QVS [8], ASM [7], FireFly [10], and RT-WMP [15] that are able to provide quality of service in harsh environments, but these systems only worked for stationary nodes. Users must be within the communication range of a radio. This not only limits users mobility, but renders a stationary system useless during an emergency since users are often out of reach of the radio. *As none of these systems are designed to account for node movement, they would perform poorly when nodes move around and network topology dynamically changes. This node movement is exactly the focus of our work.*

Voice Streaming in MANETs. Voice streaming in MANETs has been investigated in several previous studies. These systems [2,3,9,16] monitor the impact of node movement on interference, communication failure, and voice quality in a network. *Although these protocols account for mobility, they are designed for higher bandwidth networks such as 802.11 or require additional information like GPS. Furthermore, most have only been evaluated in a network simulator, and it is unclear of their performance when implemented on actual hardware and evaluated in a more realistic environment.*

3 System Overview and Preliminaries

The architecture of our quality-aware voice convergecast mobile system (QACM) is shown in Fig. 1. QACM is designed to produce satisfying audio in environments where audio streams originate from mobile nodes. In order to handle instability caused by node mobility, QACM makes several audio and routing adjustments. It monitors voice quality in real time and adjusts audio compression and data duplication levels to minimize bandwidth and maintain the guaranteed level of voice quality. It also monitors channel contention among transmitting nodes by controlling the admission or rejection of potential voice streams in order to preserve the necessary throughput at a sink for the already admitted streams. *What distinguishes QACM from existing work is its capability to function in a mobile network as a result of each node continuously adjusting its routing to the sink independently.* The rest of this section presents necessary background for the integrated system and the next section describes our major contribution, i.e., how to manage node mobility with admission control and routing decisions.

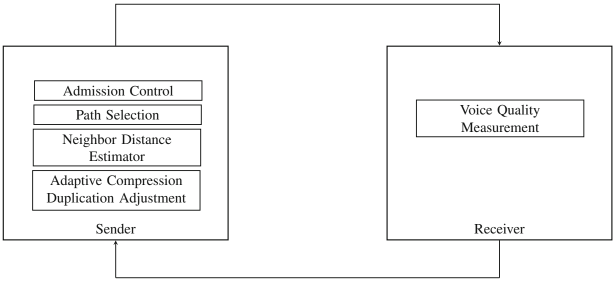


Fig. 1. System Architecture of QACM

In order to calculate voice quality in real time, we adopt the E-model [4] for voice quality measurement, where voice with an R-value above 50 is considered satisfactory. Voice quality is maintained through dynamic audio compression adaptation. We adjust audio compression and packet duplication by following the procedure described in [8]. We use ADPCM [14] to compress 16-bit raw audio into either 5-bit, 4-bit, 3-bit, and 2-bit to reduce bandwidth. We then duplicate a percentage of the packets depending on the amount of packet loss along the path of the audio stream.

Local Capacity Estimation. In order to guarantee high quality while admitting the maximum number of voice streams in the network, nodes must be aware of their saturation rate and contention domain. Contention Domain [8] of Node i refers to the set of nodes whose transmissions directly interfere with Node i's transmissions. Saturation Rate [8] is the maximum throughput observed at a node when all the senders are within the same contention domain. Thus when a node increases its data rate, it must be mindful not to cause the data rate for any contention domains to increase beyond its saturation rate. The local capacity of a node is the minimum local capacity of all nodes in its contention domain.

Although the interference range is larger than a node's communication range, we follow the same simplified assumption as stated in [8]. Only the nodes within a communication range of Node i are members of Node i's contention domain. Although it is well known that interference range exceeds communication range, we calculate saturation rate conservatively to account for the difference between interference and communication ranges.

Traditional Admission Control. For a Node v, we denote $N_p(j)_v$ as the set of nodes that lie from Node v through the first hop neighbor j to the sink. $\mathcal{P}_v$ is the set of all possible routes from Node v to a sink.

$$\mathcal{P}_v = \{N_p(1)_v, \ldots, N_p(j)_v\} \tag{1}$$

We represent all the nodes that are actively sending data in the network that will be affected by a stream s_v along the path $N_p(j)_v$ as $N_a(j)_v$. A route from Node v to a sink must satisfy the following two constraints to become a member of $\mathcal{P}_v$:

1. **Stream Quality Constraints.** Injection rate $\lambda_{in}(s_i)$ for any nodes in the network that forward voice streams $s_1 \ldots s_m$ must be greater than or equal to the threshold rate $\lambda_{th}(s_i)$ to maintain satisfactory voice quality. The threshold rate is derived from the E-model.
2. **Local Capacity Constraints.** For a Node $k \in N_a(j)_v$ the total data rate that flows through itself and the nodes in its contention domain must be no larger than its local capacity B_k.

If a path $N_p(j)_v$ satisfies both constraints, then the path can be added to the set $\mathcal{P}_v$.

4 Integrated Routing and Mobility-Aware Admission Control

In wireless networks comprised of stationary nodes, channel contention, environmental noise, and network congestion are common criteria for admission control. However, mobile nodes complicate admission control decisions. The changing topology of the network makes favorable admission control decisions at time t suddenly detrimental to voice quality at time $t+1$. Several previous works [2,3,9] have adapted traditional admission control to mobility, but often use high bandwidth wireless mediums like 802.11 or have only been evaluated through simulations. We have designed a novel mobility aware algorithm in which routing decisions complement admission control.

Admission control for a quality aware based voice streaming protocol determines if a network can guarantee robust audio throughput for the duration of the audio stream. Although several previous works have created admission control systems that provide good voice quality, mobility is not considered in their design [8].

Previous mobility aware admission control protocols are reactive [3]. These protocols wait until a current path is unusable before searching for an alternative causing unnecessary delay and disruption to an audio stream. Alternatively, our protocol proactively monitors paths to a sink node and automatically switches to superior paths without interruption. Furthermore, the routing decisions strengthen a future voice stream's admission candidacy by seeking paths for current voice streams that have the least impact on the network. Proactive routing algorithms often introduce high levels of overhead. Our system minimizes the maintenance by focusing only on local connections rather than maintaining global knowledge of the network.

4.1 Quality of Path

Path quality is an important metric to be used in our integrated admission control and routing protocol. We define $P_v(j)$ as the set of neighbors affected by a path between Node v via neighbor Node j to a sink. $N(v)$ defines all the neighbors of Node v. We define the cardinality of $P_v(j)$ as **Quality of Path** ($QoP(v)$). A node will use Quality of Path to choose the next hop neighbor on a path to a sink node. Unlike local capacity constraints, $QoP(v)$ accounts for nodes affected by the stream regardless if they themselves are streaming data. The purpose of this metric is to minimize the size of $N_a(j)_v$ and the local capacity constraint for a future stream. To calculate $QoP(v)$, a node takes the number of contention domains affected by its own transmission and unions this set over the set of contention domains along the entire path. The cardinality of the produced set is the $QoP(v)$:

$$P_v(j) = N(v) \cup P_j(i) \tag{2}$$

$$QoP(v) = \min_{j \in N(v)} (\|P_v(j)\|) \tag{3}$$

A path with a larger $P_v(j)$ will lower the local capacity for more neighborhoods and reduce the number of future data streams that can be admitted.

4.2 Integration of Quality of Path with Hop Count and Node-Degree

Selecting the best path between a sender and receiver has significant impact on the data loss and voice quality as discussed in [3,9]. Traditional wireless routing protocols such as Ad hoc On-Demand Distance Vector (AODV) and DSR

[6,11,12] use hop count as a metric to minimize the distance data travels, but these protocols do not account for link quality, utilization, and distance between nodes. Moreover, when two paths are available to a node with identical hop counts, path selection is made arbitrarily. Expected Transmission Count (ETX) [5,8] is an alternative metric that focuses on link quality rather than distance. However, ETX needs to know the number of transmissions and retransmissions along paths before making a routing decision. In mobile networks, historical transmission data is not available since routes change frequently. The delay incurred in collecting this data during a route switch would also reduce the quality of audio communication.

QACM employs a novel route selection protocol that combines the metrics of hop count and the number of neighbors a node has along the path. The goal is to avoid data bottlenecks by minimizing the degree of a node especially if that node is already generating a voice stream. Since QACM is designed for convergecast, it is preferable to avoid converging a large number of streams at a node that is multiple hops from the sink. Any hardware has a maximum data rate it can receive and forward without data loss. Minimizing the number of streams a node has to relay would reduce the likelihood of surpassing this maximum rate and resulting in data loss.

Every time a node moves into the broadcast range of another node and becomes a neighbor, that node must calculate the strengths and weaknesses of this new neighbor becoming its next hop. Contention domain, hop count, quality of path, and the presence of another stream being forwarded by this neighbor are all considered. The order in which these metrics are considered is based on that particular metric's impact on maximizing quality audio and the number of streams in a network. Our decision process is illustrated in Fig. 2.

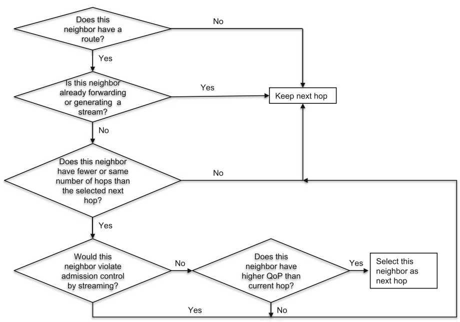


Fig. 2. Process for determining the next hop

4.3 Integrated Path Selection and Admission Control

Our protocol is based on AODV [12]. When a node is activated, it repeatedly broadcasts heartbeat messages to its neighborhood. Neighbors use the information in the heartbeat message to populate its neighborhood table.

When a node has received heartbeat messages from all its neighbors and has populated its neighborhood table, it determines which neighbors have a route to a sink node and marks the chosen neighbor as its next hop to a sink node. This updated information will then be reflected in the node's next heartbeat message.

Once a source node receives and obtains a route through a neighbor, the source node will begin the distributed admission control process. The admission control process regulates the number of voice streams in the network. Its goal is to guarantee that the voice quality of current streams does not fall below the stream quality constraints. Nodes that cannot adhere to the stream quality constraints would send a reject message (REJC) to the sender node. If a sender's injection rate is accepted by the affected nodes along the path, the sender can begin sending data.

The source node adjusts its compression and duplication settings to produce the best quality for the audio stream based on the data loss along the path to the sink. If the source node cannot set the compression and duplication to keep the audio quality above $\lambda_{th}(s_i)$, the audio transmission will be stopped. A full outline of our algorithm is described in Algorithm 1.

Algorithm 1. Integrated Routing and Mobility-aware Admission Control

```
loop
    broadcast heartbeat message
end loop
Event: Receive heartbeat message
    compute relative distance from RSSI and update neighborhood table
    compute QoP and update neighborhood table
    update next hop from the flowchart and change in relative neighbor distance
    update neighborhood capacity
    update local capacity
Event: Receive INIT
    call Admission Control Algorithm
Event: Receive PATH
    calculate packet loss from upstream neighbor
    update #PathPacketSent and #PathPacketReceived
    forward PATH packet
```

4.4 Path Adjustment

During audio streaming, intermediate nodes continuously look for lower cost paths to a sink for two reasons. One, a node may determine that a different path to the sink affects fewer contention domains, has fewer hops, or is not sharing a path with another stream. Two, a node may learn that its next hop neighbor is moving outside its contention domain. If either situation arises, a node will switch its next hop neighbor on a different path to a sink node. Although a different path may have fewer hops and be more stable, it must still adhere to both stream-quality and local capacity constraints. To test for local capacity constraints, the intermediate node attempting to switch will begin by broadcasting an initiation (INIT) message as before and continue to follow Algorithm 2.

Algorithm 2. Distributed Implementation of Admission Control

Input: An INIT message generated by source i is received by an intermediate node j
$\lambda_{inc} = 0$;
if It is the first INIT message originated from i **then**
  set a timer T_{ch};
end if
$\lambda_{inc} = \lambda_{inc} + \lambda_{in}(i)$;
if j is on the path from i to the sink **then**
  forward the INIT message;
end if
Event: Timer T_{ch} Fires
if $\lambda_{inc} > B_j$ **then**
  send REJC message to the source node;
end if

The purpose of the path adjustment is to minimize the number of link failures between nodes on a voice stream path and to avoid congested areas in the network. Link failures force sending nodes to pause voice streams and wait until a new path to the sink node is discovered. If a link failure is predicted, this audio stream pause can be avoided. Furthermore, to increase the overall number of voice streams in the network, it is vital to choose paths that interfere with the fewest number of nodes. This path selection process will keep local capacity for nodes high and increase the acceptance rate for future voice streams. If the new path can handle the injection rate, then the intermediate node will switch the route.

4.5 Neighborhood Maintenance

One of the biggest impediments to sustaining a route capable of delivering good audio quality are link breaks. A link break disrupts audio streaming because a new route must be established before streaming can continue. In our work, we use relative distance estimation to predict future link breaks. That is, nodes proactively start a route discovery operation and reroute streams to avoid a future broken link. For a Node i, it tracks the relative distance between itself and its neighbors through RSSI. We use the Path Loss Model [13] to estimate the distance between Node i and its neighbors.

If Node i receives consecutive messages from a downstream neighbor, Node i will search its routing table for neighbors who have paths to the sink with a closer distance. Determining a neighbor's relative distance is important to keep a routing table fresh; it also increases the lifetime of an audio stream despite uncertainty of link stability in the network.

Nodes monitor available paths to the sink. A node may have selected a next hop neighbor, but its next hop neighbor's movement will cause the link to break. If a neighbor moves out of range without a prediction mechanism, the upstream node would have to pause the audio stream and start a route discovery once its neighbor stops receiving voice stream data packets. Predicting link breaks will significantly reduce this delay. Suppose there is a voice stream with the path:

$$N_p(v_1)_{v_0} = \langle (v_0, v_1), \ldots, (v_{k-5}, 2), (8, 7), (9, 10) \rangle$$

Each Node v_i reports its RSSI through its periodic heartbeat messages. The packet loss across the whole path is then reported to the source Node v_0, which will adjust the compression and duplication settings for the stream.

This routing design makes a path completely dynamic. No single node has complete knowledge of its path to the sink. Each node with a route to the sink chooses its next best hop. Nodes try to minimize the number of contention domains affected and initiate a new route discovery before communication with a neighbor is lost.

5 Performance Evaluation

In practice, admission control is designed to reject streams that do not maintain quality audio. However, for our evaluation, in order to test whether our admission control is conservative, we explicitly turned off this feature and only record the admission control decisions while not notifying the sink. In other words, we allow a new stream to join even if it will potentially worsen the voice quality. We can determine the extent that streams would be able to recover. Due to page limitations, more detailed evaluation results can be found in [1].

5.1 Experimental Setup

Our hardware is comprised of a microcontroller, a 802.15.4 radio, and a modular circuit board or Shield to connect the radio to the board. Collision avoidance with CSMA is used as the MAC layer protocol. We chose Arduino Due microcontroller in combination with an XBee 802.15.4 radio to send both network layer and application layer packets.

We performed all evaluations in a nine-square-foot small home office that contained a bookcase, a desk with a computer, and a chair. Although XBee wireless radios are capable of transmitting data up to 100 ft indoor in theory, our preliminary testing showed that distances higher than 3 ft would increase packet loss to 50% when transmitting voice data. Therefore, we configured the

XBee radios to transmit at its lowest radio power of −10 dBm, so that a radio was considered out of range at a distance of 1.5 ft.

QVS [8] is a quality-aware voice convergecast system designed for stationary low power wireless networks and it is the most relevant to our work, so we will compare the performance of QACM with that of QVS. QVS evaluation used custom built hardware called SenEar that has a data rate of 500 kbps; to ensure fair comparison, we implemented QVS using our chosen hardware, the XBee S1 radio, which has only 250 kbps data rate.

5.2 Experimental Results

We begin our evaluation with one mobile stream and then increase the complexity through each additional test by adding more mobile streams, changing straight line movement to random movement, and increasing the number of sinks.

Scenario 1: One mobile node with voice stream. As shown in Figs. 3 and 4, We place the sink node and two relay nodes in a straight line 3 ft apart. We test two scenarios: (1) mobile Node 4 starts about 1.5 ft from Node 3 and moves towards sink; (2) mobile Node 5 is placed about 1.5 ft from the sink node and moves away from the sink. A dashed line indicates the path of the voice streaming data link at time instant t_n. A solid colored line indicates the direction of the moving node. In both scenarios, only the mobile node is sending voice data. The relay nodes only transmit data generated by the mobile nodes.

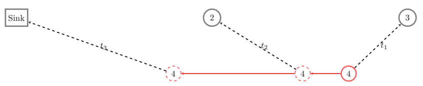


Fig. 3. Node placement (mobile scenario: one streaming node moves towards sink)

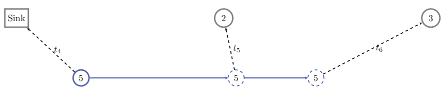
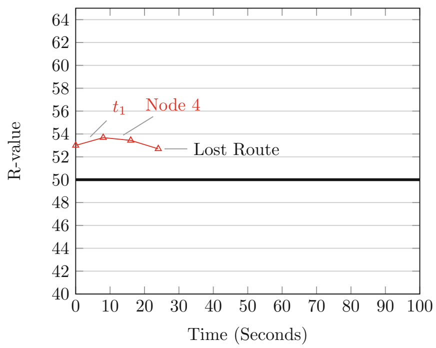


Fig. 4. Node placement (mobile scenario: one streaming node moves away from sink)

Figure 5 shows the voice quality of QVS in this scenario. At the start, Node 4 connects to Node 3 which then relays voice data to the sink. Node 4 then starts moving toward the sink. Once mobile Node 4 cannot reach Node 3 anymore, Node 4 ends the voice stream because it does not have a route. Since QVS is designed for stationary nodes, QVS does not adjust its next hop and the connection is lost. Since QVS cannot maintain a voice stream with even one mobile node, we do not further evaluate QVS with more complex scenarios.

Figure 6 shows the results for QACM. Each t_n is the time point when the mobile stream adjusts its next hop. Node 5 starts at a higher R-value and decreases slightly as it moves away from the sink. Moreover, Node 4's R-value marginally increases at it moves closer to the sink. Voice quality is strong throughout both scenarios, which shows QACM can easily support one mobile node sending voice data.



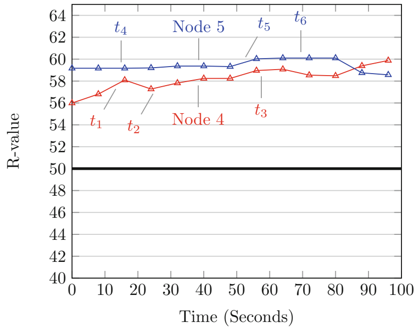


Fig. 5. Voice quality of QVS (one streaming node moves towards the sink)

Fig. 6. Voice quality of QACM in two separate mobile scenarios: streaming node moving towards the sink vs. streaming node moving away from the sink

Scenario 2: One mobile node and one stationary node with voice streams. The only difference in this scenario from the previous one is that we let a stationary node (i.e., Node 3) generate voice streams as well. In this setup, two streams are now competing for both bandwidth and hardware processing at the sink. Also, Node 3 is now both streaming voice data and could potentially be a relay for Node 4. As shown in Fig. 7, Node 4 is unable to establish a connection with Node 3. Until it moves into contact with Relay Node 2, Node 4 cannot start streaming. Node 4's voice quality never reaches an R-value of 50 and Node 3's R-value steadily declines from 51.43 to 44.65. In Fig. 8, Node 4's voice quality starts strong, but as it moves away from the sink, the voice quality decreases because the stream requires more hops to reach the sink. Node 3 starts at an R-value of 49.89, but steadily decreases as Node 4 moves away from the sink. We infer that once Node 4 switches to relay Node 2 at t_5, Node 2 is unable to receive, process, send both Node 3 and Node 4's voice data. From this scenario, it is clear that nodes are not capable of both relaying and streaming with our hardware.

Scenario 3: Two streaming nodes start moving in opposite directions. *Same start time.* We set up two streaming nodes moving in opposite directions. We first evaluate the case when two nodes start moving at the same time (Fig. 9). Figure 10 shows that streaming Node 5 starts near the sink with a voice quality above 50 and Node 4 starts near relay Node 3 with a voice quality slightly below 50. As Node 5 moves away from the sink, its R-value decreases and finishes slightly below 50 at 49.5. Node 4's voice quality initially decreases and then increases as it moves towards the sink and finishes with an R-value of 60.77. This decrease in voice quality for Node 5 and increase for Node 4 shows that an increasing in the number of hops results in lower voice quality.

Random start time. Figure 11 present the results for voice quality when two nodes have a different start time. Node 5 begins moving first, followed by Node 4 30 s

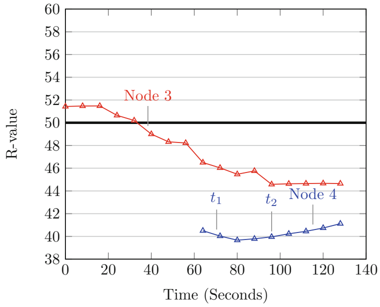


Fig. 7. Voice quality of QACM (mobile scenario: stationary Node 3 streams and also Node 4 streams while moving towards the sink)

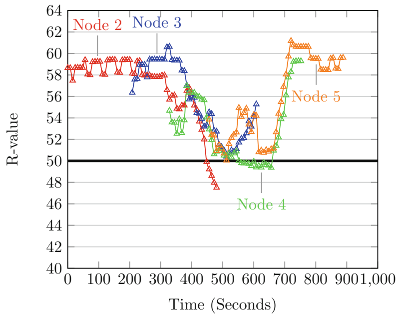


Fig. 8. Voice quality of QACM (mobile scenario: stationary Node 3 streams and also Node 4 streams while moving away from the sink)

later. Consistent with previous results, Node 5 begins with strong voice quality since it is only one hop from the sink. Voice quality then decreases as Node 5 moves toward the relay nodes and requires more hops. Node 4 starts with relay Node 3 as its next hop and begins with voice quality below 50. One obvious difference between this and the previous scenario is the number of next hop switches. When moving between Node 3 and Node 2, Node 4 switches back and forth between relay Nodes 2 and 3 14 times. Node 5 also switches more frequently than in previous experiments. Although QACM is designed to mitigate this issue, this ping-pong effect is still possible because of changes in signal strength. QACM prefers a next hop that has the minimum number of next hops to the sink. A node expects a heartbeat message at regular intervals. If a heartbeat message is missed, it is assumed that the sender is now out of range, and the mobile node assumes it must remove this node as a potential next hop resulting in a switch. It is clear that not receiving constant heartbeat messages from both relay Nodes 2 and 3 results in continuous next hop switching.

Random turn. We next evaluate the case when two streaming nodes start moving in opposite directions and then turn 180° at a random time, returning toward their starting location. Figure 12 show the results of this scenario. As described in previous evaluations, streams yield better voice quality when they are streaming directly to the sink or to a relay node that is not forwarding any other streams. Node 4 is streaming directly to the sink. Node 5 is sending its voice data across 3 hops, which results in slightly worse voice quality than Node 4's quality. When the mobile nodes start moving toward each other and must use relay Node 2 simultaneously as their next hop, Node 4's voice quality decreases. After Node 4 turns and begins moving back towards its starting point, its voice quality improves because it is reducing the number of hops to reach the sink. Node 5's voice quality also improves as it moves toward Node 3. Node 5's voice quality improves since it is the only stream using the relay nodes, but its improvement is not as dramatic as Node 4 since its moving away from the sink.

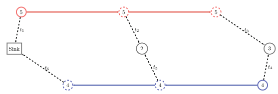


Fig. 9. Node placement: Nodes 4 and 5 start moving in opposite directions at the same time

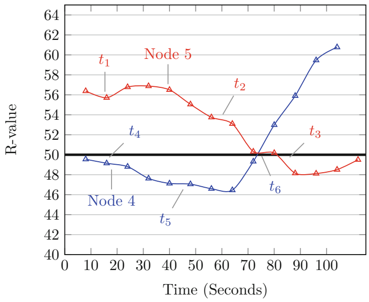


Fig. 10. Voice quality of QACM: Nodes 4 and 5 start moving in opposite directions at the same time

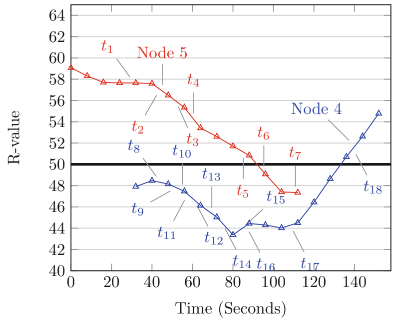


Fig. 11. Voice quality of QACM: nodes 4 and 5 start moving in opposite directions at random times

Scenario 4: Three streaming nodes move randomly with one sink. We next evaluate random movement of streaming nodes at one hop distance. The purpose of this scenario is to evaluate QACM's ability to maintain a quality connection despite continuous movement around the sink node, bandwidth competition, and sink processing power to handle all the streams. Results in Fig. 13 show that all three streams maintain R-value near 50.

Scenario 5: Streaming nodes move randomly with multiple sinks. Lastly, we evaluated one stream with multiple sinks at a one hop distance, since our previous results indicate that voice quality decreases with an increase in the number of hops. Thus, we would like to learn whether having sinks closer to the mobile streaming node can improve the situation; we, therefore, evaluate the case with multiple sinks. The purpose of this experiment was to determine QACM's ability to switch between sinks. Since there was only one stream, interference or bandwidth contention did not affect voice quality.

Figure 14 shows the R-value for one stream switching between multiple sinks. Each numbered node in this figure is a time point when the robot adjusts its direction. The colors of the curves match the colors of the sink nodes, representing the sink where the mobile node is currently sending voice data. The results show that QACM is capable of switching to different sink in order to preserve voice quality.

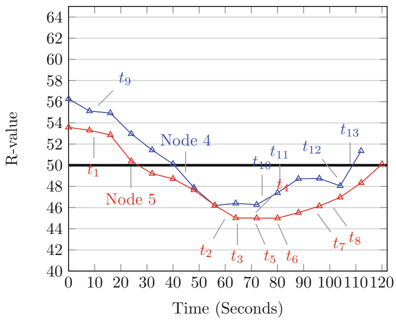


Fig. 12. Voice quality of QACM: streaming Nodes 4 and 5 move in opposite directions and turn back at random times

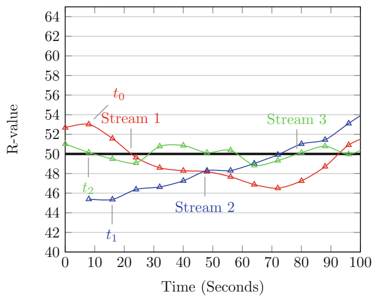


Fig. 13. Voice quality of QACM: three streaming nodes move around the sink

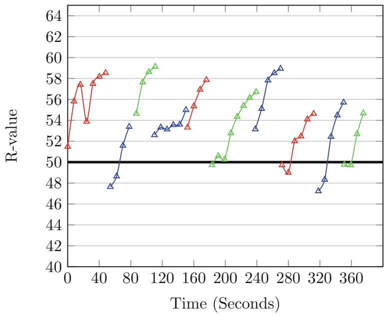


Fig. 14. Voice quality of QACM: one streaming node moves around multiple sinks

6 Concluding Remarks

In this work, we designed and developed a mobile low power voice streaming system (called QACM) that is built on a mobility-aware admission control mechanism integrated with path selection to maximize the total number of concurrent voice streams in the network with satisfactory voice quality. We implemented QACM on Arduino Due microcontrollers and evaluated the system in both stationary and mobile scenarios. Experimental results show that QACM can support up to three concurrent voice streams with quality assurance.

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