Analysis of Audio Streaming Capability of Zigbee Networks

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Abstract. Although formerly conceived for industrial sensing and control over Wireless Sensor Networks, LR-WPANs are registering an increasing interest in experimenting multimedia applications, with particular emphasis on evaluating the streaming capability of Zigbee networks. Due to their limited throughput they are not expected to provide high QoS, nevertheless there are several application scenarios such as distributed surveillance, emergency and rescue where audio and video streaming over low cost Zigbee networks is highly desirable. In this paper we first investigate the feasibility of Zigbee-like networks for low-rate voice streaming applications. We analyze important streaming metrics such as throughput, packet loss and jitter in multi-hop topologies. We propose some improvements in the stack implementation and show the performance in order to determine the streaming capacity limits of LR-WPAN networks.

1 Introduction and Contribution

The past few years have seen an explosion of research studies on Wireless Sensor Networks (WSN) and in particular on low-rate wireless personal area network (LR-WPAN) conforming to the IEEE 802.15.4 standard. WSNs have been confirmed as an important embedded computing platform and in the next future it is expected that LR-WPANs will be used in a wide variety of embedded applications, including home automation, industrial sensing and control and environmental monitoring. In this paper we focus on Zigbee [1], which is one of the most promising standards for LR-WPAN. It relies on IEEE 802.15.4 and it is specifically designed to address the need of low cost, low power solutions and flexible network routing and management. Although most of the past and the current applications for WSN focus on simple sensing and reporting activities , there is a growing demand to make WSNs really ubiquitous and in particular there is the need to support multimedia streaming for audio, voice and low-rate video. A wireless sensor network is a collection of low-cost sensor nodes that can be deployed very quickly in the environment and can communicate with each other via radio interface. To allow large networks some nodes act usually as routers for multi-hop connections without relying on any pre-existing infrastructure. For all these features together with low-power consumption, WSNs are very attractive for many applications such as conferences, intra-building communications as well as surveillance and emergency scenarios (e.g. law enforcement, rescue activities ...) and voice streaming communication is an attractive feature for many of these network scenarios.

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Since so far much attention has been paid to low duty-cycle applications, to the best of our knowledge, streaming capabilities have not been yet extensively studied for multi-hop LR-WPAN. Moreover, if we consider that multimedia streaming is very different from data communication, assuring effective audio communication over LR-WPAN is an important new challenge in the sensor network arena. In particular our case study is an emergency scenario where a rescue team must go inside hostile and unknown environments (e.g. collapsed or blazing buildings, caves, long tunnels ...) and where classic long distance wireless communications might be hampered by the nature of the environment. The idea is therefore to disseminate small wireless sensor nodes while the rescuers advance and explore the environment in order to build dynamically a network for on-site data transmission and voice streaming between the place of the disaster and the base station. Of course the network will not be stable and long-term operating, but low-power characteristics of LR-WPAN are essential to guarantee the maximum life-time during the rescue activity, and the low cost of the Zigbee devices makes the loss or the destruction of some nodes affordable.

There are several limits for achieving an effective streaming capability over Zigbee-like networks. Wireless streaming is generally an expensive operation for the limited energy budget of the nodes and it is often infeasible replacing batteries of the devices. The energy reservoir problem seems will be solved by recent studies on hardware/software power harvesting techniques which attempt to realize perpetual powered systems [2,3]. Another well-known restriction in wireless networks is the high probability of transmission errors due mainly to multi-path fading or electromagnetic interference and its intrinsic location-dependency, which make wireless communication links to have fluctuating quality levels and time-varying characteristics. Finally time constraints are also very important, because audio/video streaming applications are delay-sensitive. Usually a late arriving packet is not useful to the end node, and it is better to drop such data rather than sending it several times.

The overall goal of this paper is to describe the audio streaming capability of a Zigbee-like network over the IEEE 802.15.4 framework. In particular we examine several metrics of multi-hop communication such as throughput, jitter, latency and packet loss, using different paths and routers to deliver the information. All the measurements are performed through the analysis of a real setup using Zigbee-enabled devices deployed in the environment.

The remainder of the paper is organized as follows. Related works are reviewed in the next section whereas an overview of the Zigbee protocol is presented in Section 3. We will also argue about the decision to adopt a free protocol stack for our work. Section 4 describes the experimental hardware we use, followed by Section 5 which illustrates the results during the analysis of the Zigbee network for streaming activity. The discussion on improvements and tuning of the protocol stack is reported in Section 6, finally Section 7 concludes the paper.

2 Related Work

Streaming over sensor networks makes power management, bandwidth, memory and energy supply very challenging especially in a multi-hop domain rather than in a direct link. Several studies have been already presented about performance analysis over IEEE 802.15.4 MAC framework. In [4] the author investigates the performance and feasibility of IEEE 802.15.4 for low bit-rate audio/video streaming applications. In particular he focuses on packet loss and latency in order to find a suitable operating rate value and he proposes a method for an adaptive streaming, based on a link quality indicator. The main weak point of the article is that the author presents only simulation results using a network simulator and does not perform real measurements with sensor nodes deployed in the environment. Formerly, [5,6,7] have presented overviews of the IEEE 802.15.4 standard showing simulations, experiments or combination of both.

Theoretical research about the real-time streaming capability in a generic multi-hop WSN is presented in [8]. The author defines the capacity of the network in order to estimate the amount of data sensor nodes can deliver real-time before packet deadlines.

Practical implementations of multimedia context transmission over WSNs are also available, but usually the information to deliver is just a still image and streaming issues are not tackled. An implementation of a sensor node that can deliver multimedia information is proposed in [9]. In [10,11] the authors present image transmission over WSN, but they mainly focuses on the point-to-point transmission and power management in order to minimize the overall compression-and-transmission energy consumption.

Finally a system for voice streaming over WSN is fully implemented in [12]. The authors do not exploit a Zigbee network but investigate a TDMA-based network scheduling to meet audio timing requirements. They provide 2-way voice communication with a 24 ms per-hop deterministic latency across 8 hops. The developed hardware has a dual-radio architecture for data communication and hardware-based global time synchronization.

None of the above references focuses on the streaming capabilities of a real Zigbee network, analyzing the performance of the stack protocol under different parameters or proposing changes and improvements to outperform the current characteristics. In this paper we fill this gap using a free Zigbee-like stack protocol.

3 Stack Overview

Zigbee and IEEE 802.15.4 wireless technology are specifically designed to provide cost-effective and flexible wireless networks, which supports low power consumption, interoperability, reliability for control and sensors acquisition with moderate data rates. The scalable capability is supported in particular by the IEEE 802.15.4 standard, which defines physical (PHY) specifications to operate into three ISM frequency bands (868 MHz, 915 MHz, 2.4 GHz) and can accommodate up to 27 channels with a maximum raw data rate of 250 Kbps for the 16 channels allocated in 2.4 GHz band. Clearly this might be a limiting factor if transferring larger amounts of data is required. Devices currently available on the market work with a transmission range between 10 and 70 m.

Medium access control (MAC) specifications are also provided by IEEE 802.15.4 standard. The network can operate in two configurations: beacon enabled and beaconless mode. Beacon mode defines synchronization and reliability of the transmission mechanism, whereas beaconless networks adopt a simple lightweight protocol based on CSMA-CA. Although using no beacons is generally preferred, this mode registers more power consuming communications because of the more collisions which make the node to wait for the retransmission of the frame.

Zigbee protocol relays on the underlying IEEE 802.15.4. It manages routing protocol in the network layer (NWK), security and name binding in the Application Support Sublayer (APS), and defines the Application Framework (AF) for user applications. In our work we focus on streaming applications which are built over (AF) and are forwarded using multi-hop path by the network layer. Zigbee specifies three types of nodes, for different activities: the Zigbee End Device (ZED) which provides information to deliver, the Router (ZR) and the Coordinator (ZC) which is unique in a network. The coordinator has to synchronize the network, maintaining the routing table, has to accept new nodes in the network and has to manage the disconnections. Usually it is also employed as data sink.

In streaming communications the maximum packet size should be transmitted, because with the increase of the data unit size the overhead of the headers is reduced. Unfortunately IEEE 802.15.4/Zigbee protocols do not define large payload in their specification. For instance, the maximum payload data unit defined by PHY layer is limited to 127 bytes. Since the MAC header requires a maximum of 23 bytes, and up to 17 additional bytes are reserved for NWK, APS, and AF layers, the actual user data unit size at application level is limited to 89-93 bytes (depending if long or short addressing is adopted during communication). In this situation an efficient fragmentation mechanism becomes essential for a streaming application, but again Zigbee does not specifies data fragmentation and reassembling protocols, and the implementation of fragmentation and flow control mechanisms at application layer is up to the end-user development.

We tried several commercial solutions ranging from Freescale [13] to Telegesis [14]. So far we have found out that a real integration and interoperability among these systems is not yet completely fulfilled, since some devices do not provide all the features of Zigbee 1.0. Since our work is not focused on interoperability and Zigbee profiles compliance, we decided to adopt for our tests a Zigbee-like protocol stack in order to evaluate the performances of the streaming capability. In particular it is provided source free allowing the developers to look deep into the code. This is an interesting feature because one goal of this work is to investigate and flush inefficiencies, optimize performances and point out hardware and software lacks. The stack is developed by MS State University [15], and although it is not certified as compliant by the Zigbee Alliance, it does use the NWK, APS, AF frame formats from the Zigbee standard implementing static trees and routing as specified in the standard, so it actually performs all the main features that are fundamental for streaming analysis of the Zigbee protocol [1,16].

4 Experimental Setup

As hardware platform, we exploit a solution provided by Texas Instrument [17], using the system on chip (SoC) CC2430. The system operates at 2.4 GHz band and offers a raw bit rate of 250 Kbps. The device integrates all operational functions such as radio transceiver, data processing unit, memory and user-application features on one single silicon die and this contributes greatly to performance, power consumption and cost. High performance and reliability at lower power consumption is achieved due to the close interaction of dedicated on-chip functions minimizing overhead. In particular MAC timing operations are handled more effectively by dedicated circuitries, and the system integrates a significant set of the IEEE 802.15.4 requirements (e.g. CSMA-CA, preamble generation, synchronization, CRC-16) to off-load the micro-controller.

As we already remarked, audio streaming is very different from data and control communication due to the inherent delay constraints. If data arrives too late, information is no more useful for playing audio and this leads to the consideration that it is better to drop it at the sender or somewhere in the path. Too late packets could happen for various reasons, for example, the necessity of the sensor node to react to external events in a timely manner. To evaluate the audio streaming performances over a Zigbee network, we off-load the SoC device from audio conversion and compression processing using external dedicated devices. Audio information is sampled at 8 KHz and data is coded using 8bit A-law conversion, moreover we can dynamically perform additional compression using an external ADPCM processor and select dynamically the desired audio rate ranging from 16 Kbps to 64 Kbps. Of course this first step of data processing results in an increase of power consumption due to additional devices but it helps to separate and identify the cost of the Zigbee stack in power consumption, computation effort and its reactivity, without any interference of other on-board activities. On the other hand the power consumption of the CODEC is comparable to the consumption that the SoC CC2430 registers when it performs additional A/D conversions and processing. For audio processing, we use the PCM codec TLV320AIC1107 from TI which consumes no more than 20 mW when it performs coding/decoding procedures, while the DS2165Q ADPCM processor chip from Maxim is exploited for ADPCM compression at the desired data rate and may require up to 60 mW. To guarantee a complete decoupling between signal processing and streaming procedures, we adopt double buffers architecture between CODEC modules and the micro-controller as depicted in Fig. 1. To verify the quality of the voice transmitted over Zigbee we implemented a simple full-duplex push-to-talk (PTT) application between nodes in the network. The FSM used in our system is illustrated in Fig. 2.

Depending on the RF environment and on the required output power consumption, Zigbee compliant wireless devices are expected to transmit in a range of 10-70 m. The evaluations presented in this paper are selected between several measurements

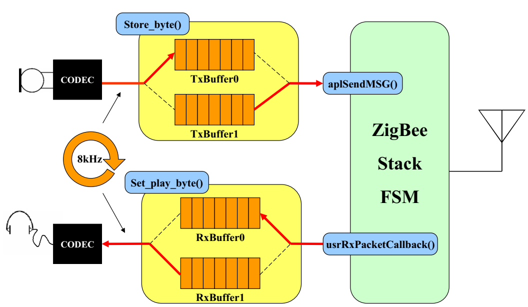

Fig. 1. Buffer architecture for audio streaming

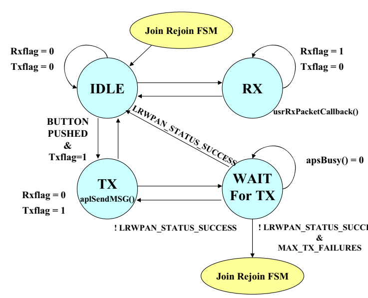

Fig. 2. FSM of the PTT application

performed using distances ranging from 5 m to 35 m between nodes in the network and exploiting different payload sizes at AF level (varying between 8 and 93 bytes). We used a beaconless network and direct transfer mode to send messages. In order to minimize the influence of other ISM transmissions over 2,4GHz, the WPAN is formed using the channel 15 because it is pretty unaffected by IEEE 802.11 networks in Europe (as well as channels 16, 21 and 22).

The voice stream is delivered at constant bit-rate (CBR) and its value can be selected at the startup. For the experiments presented in this paper we vary the bit rate ranging from 24 Kbps to 128 Kbps. In every measurement session at least 10000 packets were sent over the network for each experimental test conditions and the results were averaged over a minimum of 20 trials. It is worth to specify that measurements are taken indoor with no particular attention to serious obstacles for signal propagation in order to approximate a real scenario of streaming infrastructure for emergency rescue in hostile environments.

5 Performance Evaluation

In this section, the experimental results are reported. We begin with a baseline analysis of the timing performance of the used devices, then we discuss throughput measurements, followed by some considerations on the deployment of the network in the environment and the problem of the shared channel. Finally further measurements aim to investigate the latency, the inter-packet delay, the jitter, the packet loss and the power consumption of a streaming LR-WPAN.

5.1 Time Analysis

In addition to simulation results, another useful way to estimate the best-case streaming performance over Zigbee is to analyze the time the platform needs to deliver the messages and to receive acknowledgments from the destination node. In this way we can separate the contribution of the latency between software implementation and hardware components of the protocol stack. As remarked, the module CC2430 executes several MAC and PHY operations directly on dedicated built-in hardware to guarantee the maximum efficiency in terms of power consumption and execution time. We investigate the time necessary to deliver a message in a point-to-point configuration between nodes and an analogous measurement has been done for a router device. The results are depicted in Fig. 3. The time necessary for synchronization, preamble generation, accessing to the medium using CSMA protocol, sending and receiving is around 4,5 ms per link. Since routers forward incoming messages this hardware delay contribution is twofold. Crossing the stack from the upper layers requires less time because of the activities of NWK, APS, AF are generally simpler and our work does not consider any operations concerning security and cryptography of the messages. This time information is obtained measuring the interval between the activation of GPIO signals triggered in particular moments. We intercept for instance when the user calls the AF layer for message delivery, when the software part of the stack writes to the FIFO `TXFIFO` of the SoC letting the hardware to complete the delivery of the packet, and when the signal

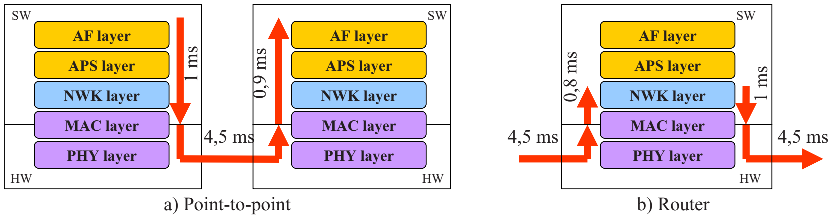

Fig. 3. Time to cross the HW and SW part of the protocol stack

`IM_TXDONE` of the MCU informs that the transmission has successful completed with acknowledgment.

Using a point-to-point configuration and exploiting the maximum data unit size of 93 bytes, the delivery of the message to destination takes 6,4 ms in the best case. We can use this information to find out a practical estimation of the best case data rate in a ZigBee network. Assuming that:

- We perform a single hop transmission;
- There is no overhead in the node activity;
- There are no lost packets.

we can compute that the highest data rate is expected to be

$$data\ rate = \frac{maximum\ payload}{time\ to\ deliver\ the\ message}$$

that is 116,25 Kbps. It means that the effective utilization of the channel for user information is limited to the 46,5% of the theoretical raw data rate (250 Kbps) claimed by the standard.

Under the same assumptions we could also compute the best forward rate of a Zigbee router in a network. In this case, if we consider a network with precompiled routing tables, the expected forward rate is

$$data\ rate = \frac{maximum\ payload}{time\ to\ cross\ the\ router}$$

that is 68,9 Kbps.

5.2 Throughput

In this work the throughput is defined as the amount of the data units (PDU) correctly arrived to the destination, divided by the length of the interval of the experiments (i.e. between the first and the last delivery $T_{end} - T_{start}$):

$$Throughput = \frac{\sum_i Length(PPDU_i)}{T_{end} - T_{start}} \tag{1}$$

The setup used for this experiments is a multi-hop string topology varying the number of the routers in the path as illustrated in Fig. 4.

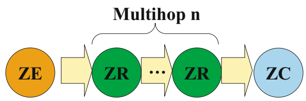

Fig. 4. Topology used for multihop tests

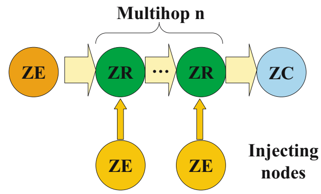

Fig. 5. Topology used for cross traffic tests

The throughput in a multi-hop path decreases quickly, as shown in Fig. 6. We performed measurements varying both the size of the user data unit (8, 20, 32, 46, 64, 93 bytes) and the number of hops using up to eight routers. In particular the plot shows that increasing the number of hops the throughput degrades faster for large sized payloads. This higher degradation is due to higher probability of collisions for large packets during CSMA, if there are several nodes of the WPAN in the same transmission area. No optimizations were implemented in the protocol stack during these experiments.

Considering that our scenario is intended the share the network infrastructure for other kind of data transmission (e.g. status of the rescuers, temperature in the blazing building ...), we performed experiments also with cross traffic. In particular, as illustrated in Fig. 5, other nodes, called injecting nodes, are joined to the same network and exchange data with the coordinator (ZC). We use different values of cross traffic, ranging the rate from 93 bps to the highest rate achievable by the node, using the maximum available payload and we change the position of the injecting nodes across the network. Figure 7 shows the degradation of the throughput when two hops divide the streaming sender from the receiver and only one injecting node was connected to the router. In the x-axis the different rates of the cross traffic packets are reported and considering that in the absence of cross traffic the measured throughput is 61,4 Kbps, the plot shows that only moderate injecting rate keep the degradation of the stream throughput reasonable low. Considerable reductions of over 40% are registered starting from 930 bps traffic rate.

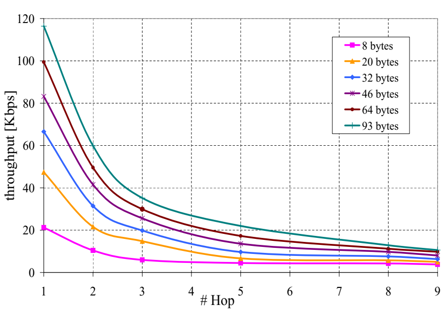

Fig. 6. Effective throughput varying the number of the hops

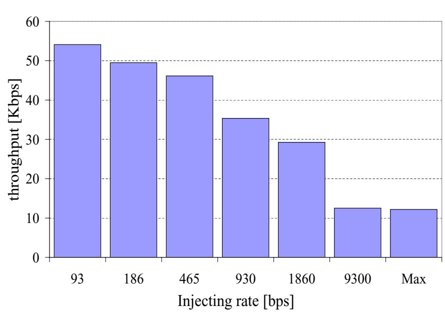

Fig. 7. Throughput in function of the cross traffic level

5.3 The Shared Channel Problem

The absence of a dynamic policy for channel switching between nodes belonging to the same PAN is one of the main shortcomings of the current Zigbee specifications. Once the coordinator has selected the channel for the personal area network, all the nodes will join the PAN will work and share the same channel. Of course nodes may join to different networks in the same time using different channels, but at the moment there are no specifications concerning frequency hopping within the same network. In other words, because the channel is fixed once the PAN is formed, if nodes are deployed too close to each other they have to share the same space and channel causing an uncoordinated access to it. To achieve high network utilization it is necessary maximize the number of nodes which can transmit concurrently and therefore exploiting spatial reuse becomes essential. As example we consider the deployments illustrated in Fig. 8. In a) a situation which does not take in account spatial reuse and can perform only one transmission is compared with the deployment b) where several nodes can communicate each other in the same time without interference, due to smart radio range coverage.

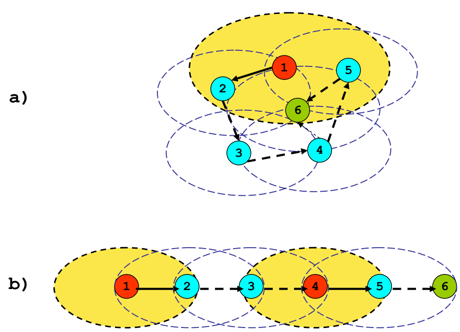

Fig. 8. Deployments: a) shared space and channel; b) network which exploits the spatial reuse

We propose algorithm 1 to enable this kind of smart coverage in a Zigbee network. The algorithm is solely designed for the initialization phase, when the PAN is already formed by the coordinator and sensor nodes are joining to the network. The main idea is to keep the transmission range of a sensor node as short as possible, avoiding the contention of the channel with other nodes of the PAN which are not directly linked to it. Obviously such a configuration may suffer of instable connections. but this problem can be solved with a dynamic tuning of the output power level using the Link Quality Indicator (LQI) of the connection as feedback information from the receiver. One of these method is presented in [4].

Using the proposed algorithm to build the network and maximizing the number of concurrent transmission, we repeat the measurements of throughput. The comparison between the two deployments is shown in Fig. 9. It confirms that adopting an intelligent distribution of the network may increase the throughput up to 30,5 Kbps. Since this kind of deployment defines a set of nodes which can transmit concurrently, this pattern can be repeated in the space with low effects on the throughput. In the figure we can see that

Algorithm 1. Spatial reuse algorithm

Require: Maintain an ordered set of increasing programmable RF output power level for the Zigbee node $\{Pout_i : 1 \leq i \leq NMAX\}$;
Output: Transmit power level *TPL*;
Data: Joined node *JOINED*;
initialize the Zigbee node;
n $\Leftarrow$ 1;
TPL $\Leftarrow Pout_n$;
JOINED $\Leftarrow$ FALSE;
JOINED $\Leftarrow$ join_to_network();
while (*JOINED* = FALSE) **do**
  if *TPL* = $Pout_{NMAX}$ **then**
    return Join procedure failed;
  end
  n $\Leftarrow$ n + 1;
  TPL $\Leftarrow Pout_n$;
  JOINED $\Leftarrow$ join_to_network();
end
return Join procedure successful;

it becomes almost independent from the number of nodes and the values we measured could perfectly sustain an audio streaming using a constant bit rate of 24 Kbps.

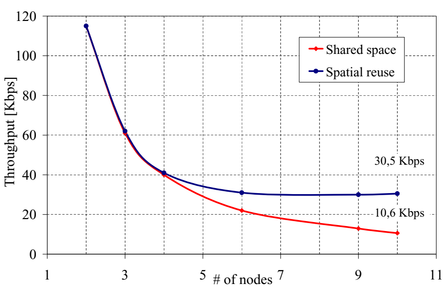

Fig. 9. Throughput of networks with different deployments

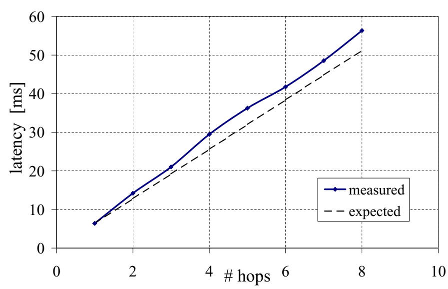

Fig. 10. Expected and measured latency varying the number of the hops

All the measurements presented in the following sections, are taken using this kind of setup which maximizes the number of concurrent transmissions.

5.4 Latency

The latency is also known as delay and it is usually defined as the amount of time required by a packet to travel from source to destination. Together, latency and throughput define the speed and capacity of a network. Real time and full-duplex streaming communications must consider this metrics very carefully in a network deployment. For example in full-duplex communication roundtrip delay of 300 ms is noticed by the final user and the human ear starts to detect delays of 250 ms. If such thresholds are exceeded the communication becomes annoying. Figure 10 depicts the measurements performed in order to characterize the end-to-end latency in our Zigbee testbed. Under ideal conditions and considering the timing analysis described in Section 5.1, we expected a linear increase of the delay with the number of the hops to the destination. The measured latencies validate this trend, registering also an additional overhead due to the not ideal environment. Our experiments is limited to 9 nodes and we covered a distance between sender and receiver of about 160 m, but considering that a transmission range of a Zigbee compliant device can arrive up to 70 m, these experiments confirm that a LR-WPAN could sustain voice transmission in a range of some hundreds of meters and fulfill the most common WSN multimedia application scenarios.

5.5 Inter-packet Delay

If voice streams are sent at constant bit rate, it is expected that also the receiver registers in average the same rate for arriving packets. Of course packets may be routed through different paths in the networks, take different time, and some of them may be lost during the travel to the destination, but in general it is possible to define an expected deadline for packet arrivals. For this reason we measure the average of inter-arrival time of the packets in Fig. 11. In streaming communication also the order of packet arriving is important, and in Fig. 12 we show the average of the interval time between consecutive packets.

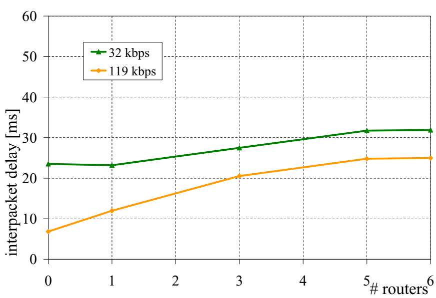

Fig. 11. Inter-packet delay varying the number of the hops

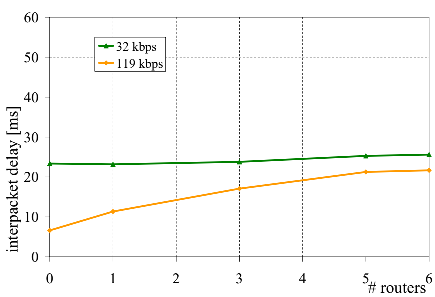

Fig. 12. Inter-packet delay between consecutive packets

The measurements were performed varying the number of the hops in the path and using different sending rate for voice transmission. Using an audio rate of 32 Kbps and exploiting the whole payload in the Zigbee messages, it is expected to receive a packet every 22,7 ms when no messages are lost. Higher values of delays are the consequence of packet loss, because missing messages at the receiver increase the inter-packet delay. To test the limits of the network, we perform the measurements also using a sender bit rate of 119 Kbps, that is the maximum rate we registered in a single hop connection (see Fig. 6). In this case the expected average inter-packet delay is only 6,2 ms, but the plot shows higher measured values in multi-hop configuration that is symptom of a high packet loss in the network.

5.6 Jitter

The jitter is a typical problem in connectionless networks and in particular in wireless infrastructure. It is closely connected to inter-packet delay since it is the measure of the variability over time of the latency across a network. Multimedia streaming has usually problems due to this effect, which affects the QoS. In a full-duplex voice service the jitter should be less than 100 ms. One of the solution to mitigate this effect is exploiting buffers between the network and the multimedia converters. A jitter buffer is basically a small queue where received messages are stored in order to give the information to the CODECs with a constant delay. Usually queue size may be dynamically modified and when it is tool small the packet loss increases. On the contrary a too large jitter buffer turns out in lower packet loss at the cost of a bigger delay experienced by final users. Figure 13 shows the maximum measured jitter varying the number of hops in the path and using different rates at the sender.

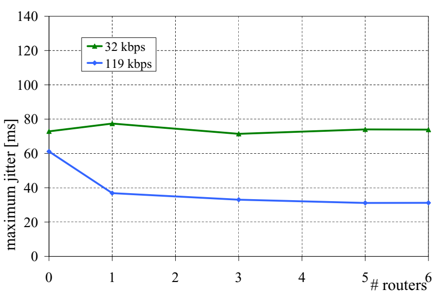

Fig. 13. Jitter varying the number of the hops

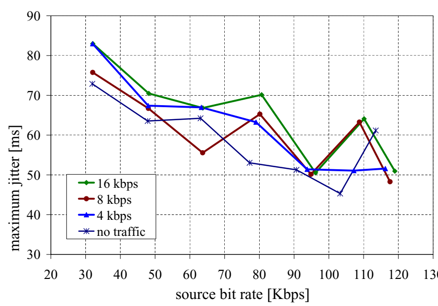

Fig. 14. Jitter varying the bit rate of the sender

Using the topology described in Fig. 5, we evaluated also the jitter under traffic condition with five routers between sender and receiver. The experiments depicted Fig. 14 show the maximum jitter measured with a variable source rate of the sender and a second node that injects data in the LR-WPAN with three different data rates. As expected, the jitter of the stream communication increases in average when cross traffic in the network grows. The evaluations are done over correctly consecutive arriving messages, therefore packet loss does not influence the measurement. All the experiments register a maximum jitter below the threshold of 100 ms for an acceptable QoS even in case of cross traffic configurations.

5.7 Packet Loss

Packet Loss can be due to several reasons, such as the congestion of the network, full buffers in some routers in the path and fails in the reception of packets (e.g. CRC fails, or channel interferences). Depending on the used audio compression level the loss of several consecutive packets may lead to a severe reduction of QoS. In fact if we use an audio rate of 64 Kbps, a single Zigbee packet contains 11 ms of audio stream, the information interval increases up to 46 ms adopting a more aggressive ADPCM compression of 16 Kbps. However voice is quite predictive and if the packet loss is isolated the voice can be heard in an optimal way. Moreover an emergency scenario accepts also low audio quality levels for the service, therefore even more lost packets are allowed if they do not occur in a burst way.

In this work we consider the packet loss as the number of the user messages that actually have never arrived to the destination at application level, divided by the total number of delivered packet:

$$Packet\ Loss = \frac{N_{sent\ packet} - N_{received\ packet}}{N_{sent\ packet}} \tag{2}$$

First of all in Fig. 15 we evaluated the dependence of the packet loss as a function of the distance using a direct link between two nodes. Measurements are taken in indoor environment with an uninterrupted burst of messages from the sender in order to emulate a critical scenario. For this reason the packet loss starts to grow quickly beyond 25 m.

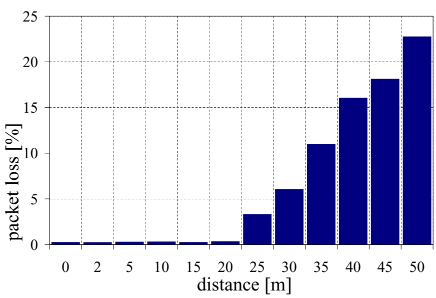

Fig. 15. Packet loss in function of the distance

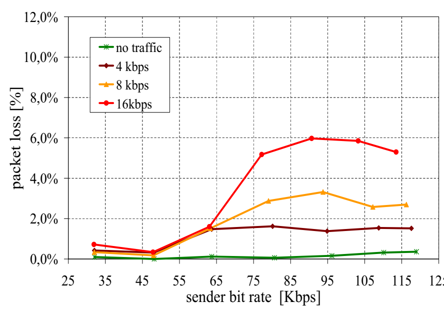

Fig. 16. Packet loss under traffic condition

Using a fairer setup with small distances between nodes and the cross traffic scenario already adopted for jitter experiments, the plot in Fig. 16 shows how the traffic from other nodes affects the packet loss. We remark that in a voice over WSN application, even if the audio stream requires most of the resources in the path, other kind of data (such as environmental data) may be delivered. This auxiliary information is characterize by low data rate and this is the reason because of the effect of the traffic begin to influence the performance with sender rate over 64 Kbps.

In a multi-hop scenario the packet loss increases with the number of hops. As depicted in Fig. 17, the maximum source rate (119 Kbps) at the sender results in a dramatic packet loss even with only one router, meanwhile with a controlled bit rate of 32 Kbps the effect of losses is mitigated, but it is still important if high QoS is required. The reason of such a high rate of missing packets is investigated in Section 6.

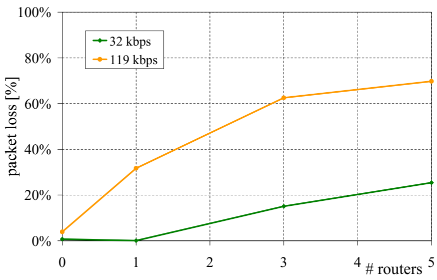

Fig. 17. Packet loss in multi-hop configuration

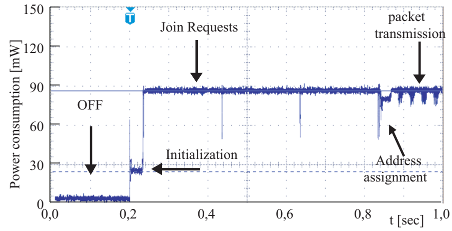

Fig. 18. Registered power consumption of the voice streaming node at the startup

5.8 Power Consumption

We analyzed the power consumption of the whole Zigbee transceivers, because the nature of the adopted SoC system does not allow to identify accurately the contribution of the micro-controller and the RF radio. If no low-power mode is adopted the Zigbee module operates always in receiving mode, after having joined a network. Figure 18 shows the power trace when the voice streaming module is switched on. It is possible to recognize, after a short initialization period of 36 ms, the voice node attempting to find a network and joining to the PAN. The join request lasts 200 ms, but the exploited hardware repeats the request three times as redundancy. Even in this phase the consumption is around 90 mW. We registered that the current used in receiving mode is around 28 mA, increasing to 30 mA when the transceiver switches to the TX mode. Since we supplied the sensor node with 3.3 V, the power consumed by the platform is 92,4 mW in RX, and 99 mW in TX. In our test we did not perform any power optimizations, neither had we modified the value of the RF transmitted power by setting internal CC2430 registers.

6 Refinements

In this section we investigate the adopted implementation for the Zigbee stack and we show how it is possible with small changes in the default parameters to improve the performance of the system. In particular we investigate the input and output buffer mechanism, varying the size of the queues in a Zigbee router device.

In Fig. 19 we analyze the causes of the packet loss in a multi-hop configuration. The main contribution to router losses is given by packet collisions at MAC level, whereas considering the messages actually received by a router, a small percentage is discarded by failure in the CRC verification. Finally a relevant amount of packets to forward are lost because the system buffers are not available to store them after a successful CRC.

The adopted stack implementation reserves memory space for 4 packets size both for the input and output queue by default. When these buffers are full, any further arriving packet is discarded. We tried to increase the buffer length and in particular we tested any combination using 4, 16, 32, 64, 128 packets as queue size. Results about the obtained

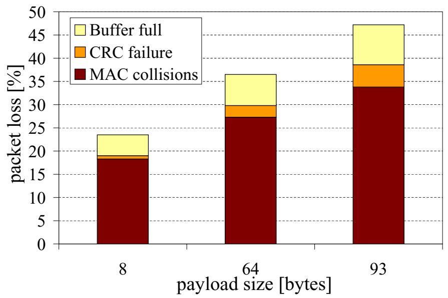

Fig. 19. Analysis of the causes of the packet loss

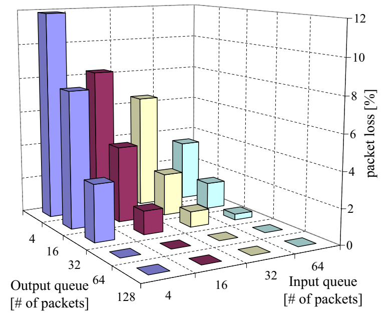

Fig. 20. How packet loss depends on the dimension of the input and output queues

performance are depicted in Fig. 20. In any measurements, we registered the number of arrived packets and correctly forwarded by the router, counting also the number of losses because of the full buffer. From the plot we deduce that the size of the input queue buffer does not really influence loss reduction (even the best size results to be 64) and it depends on the fact that the stack makes immediately a copy of the incoming message in the NWK space. Increasing the size of the NWK buffer, instead, helps to reduce the number of lost packets. With the adopted platform we found that a buffer size of 64 messages is enough to nullify the contribution of the stack to the packet loss.

7 Conclusion

This paper presented an accurate evaluation of streaming performance over LR-WPAN. All the experiments have been taken using a real Zigbee testbed with large LR-WPAN network deployed in indoor environment. We also discussed about methodology of network deployment in order to optimize the performance and we evaluated the optimal size for the input/output queues in a Zigbee router. Our investigation on metrics such as throughput, packet loss, jitter and power consumption demonstrates that it is possible to develop voice streaming applications over LR-WPANs network at the cost of an accurate deployment of the Zigbee network. Under these conditions the maximum throughput, which results almost unaffected by the number of hops, results to be around 30 Kbps and although it is not enough for high quality audio requirements, it suffices for the most common voice streaming applications. Main issues for an effective multimedia streaming over Zigbee are related not only to hardware improvements and smart deployments, but also to overcome drawbacks of the standard specifications such as an efficient low-level fragmentation mechanism and providing for larger data unit and for dynamic channel switching between nodes belonging to the same PAN.

Acknowledgements

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References

1. Zigbee Specification Version 1.0, p. 376, `www.zigbee.org`
2. Jiang, X., Polastre, J., Culler, D.E.: Perpetual environmentally powered sensor networks. In: IPSN 2005. Proceedings of the Fourth International Symposium on Information Processing in Sensor Networks (2005)
3. Moser, C., Brunelli, D., Thiele, L., Benini, L.: Real-time scheduling with regenerative energy. In: Real-Time Systems, 2006. 18th Euromicro Conference, pp. 261–270 (2006)
4. Deshpande, S.: Adaptive low-bitrate streaming over ieee 802.15.4 low rate wireless personal area networks (lr-wpan) based on link quality indication. In: IWCMC 2006. Proceeding of the 2006 international conference on Communications and mobile computing (2006)
5. Gang, L., Krishnamachari, B., Raghavendra, C.S.: Performance evaluation of the ieee 802.15.4 mac for low-rate low-power wireless networks. In: Performance, Computing, and Communications, 2004 IEEE International Conference, pp. 701–706 (2004)
6. Petrova, M., Riihijarvi, J., Mahonen, P., Labella, S.: Performance study of ieee 802.15.4 using measurements and simulations. In: IEEE WCNC 2006 (April 3–6 2006)
7. Lee, J.S.: An experiment on performance study of ieee 802.15.4 wireless networks. In: Emerging Technologies and Factory Automation, 2005. 10th IEEE Conference (2005)
8. Abdelzaher, T.F., Prabh, S., Kiran, R.: On real-time capacity limits of multihop wireless sensor networks. In: RTSS 2004. Proceedings of the 25th IEEE International Real-Time Systems Symposium, Washington, DC, USA, pp. 359–370. IEEE Computer Society Press, Los Alamitos (2004)
9. Huang, H.C., Huang, Y.M., Ding, J.W.: An implementation of battery-aware wireless sensor network using zigbee for multimedia service. In: Consumer Electronics, 2006. ICCE 2006. Digest of Technical Papers. International Conference, pp. 369–370 (January 7–11, 2006)
10. Yu, W., Sahinoglu, Z., Vetro, A.: Energy efficient jpeg 2000 image transmission over wireless sensor networks. In: GLOBECOM 2004. Global Telecommunications Conference (2004)
11. Pekhteryev, G., Sahinoglu, Z., Orlik, P., Bhatti, G.: Image transmission over ieee 802.15.4 and zigbee networks. In: ISCAS 2005. Circuits and Systems (2005)
12. Mangharam, R., Rowe, A., Rajkumar, R., Suzuki, R.: Voice over sensor networks. In: RTSS 2006. Proc. of the 27th IEEE International Real-Time Systems Symposium (2006)
13. Freescale Semiconductor, Inc., BeeKit Wireless Connectivity, `http://www.freescale.com`
14. Telegesis, Ltd., Telegesis ZigBee Technology, `http://www.telegesis.com`
15. Reese, R.B.: A Zigbee-subset/IEEE 802.15.4 Multi-platform Protocol Stack, `http://www.ece.msstate.edu/~reese/msstatePAN/`
16. IEEE Std. 802.15.4, Part 15.4: Wireless Medium Access Control (MAC) and Physical Layer (PHY) Specifications for Low-Rate Wireless Personal Area Networks (LR-WPANs).
17. Texas Instruments, Inc., TI ZigBee Solutions, `http://www.ti.com`