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A two-way radio communication across a multi-hop wireless sensor network based on a commercial IEEE 802.15.4 compliant platform

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Abstract

Wireless Sensor Networks, and especially IEEE 802.15.4, are originally defined for low cost applications, with low bit rates and power restrictions in mind. However, the ubiquity of the nodes and their easy connectivity also enable them to be used in supporting real time services, for instance, emergency scenarios, where TETRA is usually the employed audio technology. Focusing on voice transmission, we present a performance evaluation of audio streaming over a multi-hop Low-Rate Wireless Personal Area Network in order to provide bidirectional audio communication using a commercial IEEE 802.15.4 compliant platform. This paper includes an assessment of different software protocols and compression algorithms to support audio transmission on a CC2530 System-on-Chip WSN mote. The results establish the maximum number of hops of a bidirectional single-route network under real-time voice quality constraints.

Keywords: IEEE 802.15.4; Audio streaming; Multi-hop; Voice compression; Effective capacity

1. Introduction

Audio communications are essential to coordinate salvage and rescue tasks. They help in increasing the security inside caves, tunnels, mines or similar subterranean scenarios, being the radio coverage in such environments one of the biggest challenges to be solved by the emergency teams.

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The main contribution of this paper is to analyze the reliability of two-way radio communications across a commercial, IEEE 802.15.4 standard compliant, multi-hop wireless sensor network platform, the CC2530 System-on-Chip (SoC) from Texas Instruments (TI) [1]. In contrast to previous research studies, the audio transmission network is designed to improve the communication range of the current TETRA technology used by the emergency teams, while maintaining the voice quality.

TETRA (TErrestial Trunked Radio) is a TDMA (Time Division Multiple Access) radio standard which supports unicast (full-duplex and half-duplex) and multicast (semi-duplex) communications with toll quality. The typical setup time is below 300 ms for unicast transfers and up to 500 ms for multicast. One of the requirements of the system is to provide full-duplex, multicast communications.

To develop a multi-hop network with audio transmission capabilities, an IEEE 802.15.4 based solution is chosen. Low cost, low power and low data rate LR-WPAN (Low-Rate Wireless Personal Area Networks) sensor networks are usually employed for physical and environmental monitoring and reporting purposes. However, the easy installation, configuration and maintenance of such LR-WPAN make them also suitable for audio streaming applications in emergency scenarios [2].

This article investigates the feasibility of two-way communications across a multi-hop wireless sensor network based on a commercial IEEE 802.15.4 compliant platform.

2. Model and problem formulation

As hardware platform, the CC2530 System-on-Chip has been chosen. This Texas Instruments solution has an 8051 microcontroller and four power consumption modes which allow different energy saving levels. According to the IEEE 802.15.4 standard, the SoC operates at 2.4 GHz and offers a theoretical bit rate of 250 kbps.

This section is focused on the hardware and software constraints to implement an audio supporting platform.

2.1. Communication Protocol

To develop an audio communication solution, a chain topology is chosen. Due to the shared channel problem, the available bandwidth is divided among the overlapped radio links. In order to evaluate the available bandwidth, we have analyzed the net data rate of the available software stacks for the CC2530 SoC: TIMAC and SimpliciTI (table 1).

TIMAC is an IEEE 802.15.4 MAC software stack for TI SoC. This IEEE 802.15.4 protocol implementation does not support GTS (Guaranteed Time Slots) allocation. The channel access is based on a CSMA/CA algorithm. The delay associated to contention access is added to the latency related to the total number of intermediate nodes of the communication hop chain. Moreover, the time invested in channel access reduces the capacity of the protocol.

SimpliciTI is a low-power RF network protocol for TI platforms. The main advantages of SimpliciTI are easy-of-use and a shorter packet header length in comparison to TIMAC frames.

Table 1. Net data rate of the analyzed protocols

	TIMAC	SimpliciTI
Data length *(bytes)*	102	111
Average Data Rate *(kbps)*	129.40	122.92

Despite the advantages attributed to SimpliciTI protocol, the bandwidth measurement procedure works better in TIMAC. The reason is the virtual operative system layer included in the TIMAC stack, which provides event-based multitasking and reduces waiting periods.

Since the bit rate of a telephone quality voice signal after codification is 64 kbps, a simple TIMAC derived solution cannot support a multi-hop bidirectional audio communication so previous compression is required.

2.2. Voice compression

Due to the shared channel problem and the low available bandwidth in LR-WPAN, voice streaming applications require the use of compressing algorithms to support multi-hop transmissions. We propose the use of an external vocoder to release the microcontroller of carrying out audio processing and streaming procedures. An analysis of digital voice codecs is presented in table 2.

Table 2. Vocoder analysis

Vocoders	DS2164Q	CMX639	CMX7261	AMBE-3000	CMX618
Compression rate	32kbps 24kbps 16kbps	Int.: 32 kbps 24 kbps 16 kbps Ext.: 8 kbps	PCM CVSD G.729A	9,6 kbps to 2 kbps	2050 bps 2400 bps 2750 bps 3600 bps (+FEC)
Voice quality	G.726	CVSD	Variable	Toll	~ Toll
Delay	< 375 µs	< 10 ms	< 15 ms	56 ms	20 – 80 ms
CODEC	TLV320AIC1107 TLV320AIC1106	Integrated	Integrated	PCM3500	Integrated
Audio interface	I/O pin	I/O pin	C-BUS SPI	SPI McBSP UART Parallel	C-BUS
Pins and package	28 pin PLCC	24 pin TSSOP; 16 pin SOIC; 22 pin PDIP	64 VQFN	100 pin TQFP	48 LQFP 48 VQFN

The vocoder complexity grows proportionally to the compression rate; therefore the selection is done according to the requirements of the multi-hop communication. The next section presents the multi-hop capability of an audio streaming application based on an IEEE 802.15.4 network depending on the used compression algorithm.

3. Multi-hop communication

A real-time audio communication solution imposes a maximum end-to-end delay of 250 ms [3]. Factors affecting this value are the number of hops and the computational complexity of the used compressing algorithm.

Figure 1 shows the maximum number of hops supported by the network according to the delay constraint when the maximum payload size of IEEE 802.15.4 data message is used to transmit audio. The delay can be reduced for higher compressive vocoders by sending smaller data packets, but then the efficiency of the communication protocol is also affected.

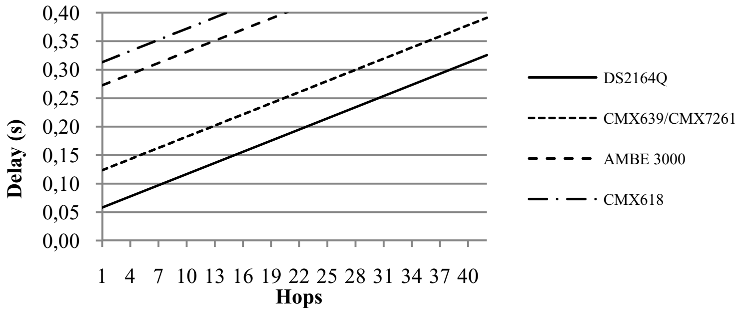

Figure 1. Maximum hops supported by the network according to the delay constraint.

In addition, the number of virtual audio links needed to establish a two-way communication depends on the nodes which have an overlapped radio range. So, the compression data rate should be selected on the basis of the required speech links and underlying protocol stack efficiency. Table 3 shows the available maximum uncorrelated links depending on the compression rate when assuming a 50% efficiency protocol and the maximum acceptable transmission period.

Table 3. Maximum number of uncorrelated links.

	DS2164Q	CMX639	CMX7261	AMBE-3000	CMX618
Transmission period *(ms)*	51.00	102.00	102.00	204.00	226.67
Transmission rate per link*(kbps)*	16.0	8.0	8.0	4.0	3.6
Uncorrelated links available	7	15	15	31	34

In the ideal case, a minimum of 6 uncorrelated links are required to avoid the radio range overlap.

4. Conclusions

In this paper, the most important aspects to maximize the network coverage of a 2-way radio communication WSN have been identified. As seen, an efficient communication protocol to reduce the idle listening periods, as well as a good vocoder (CMX639), are essential in order to deploy audio transmissions capable of transporting telephone quality voice communications.

Acknowledgements

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