



the sounds of smart environment



WP1 Test bed qualification for acoustic

C. Pham and P. Cousin (EGM)

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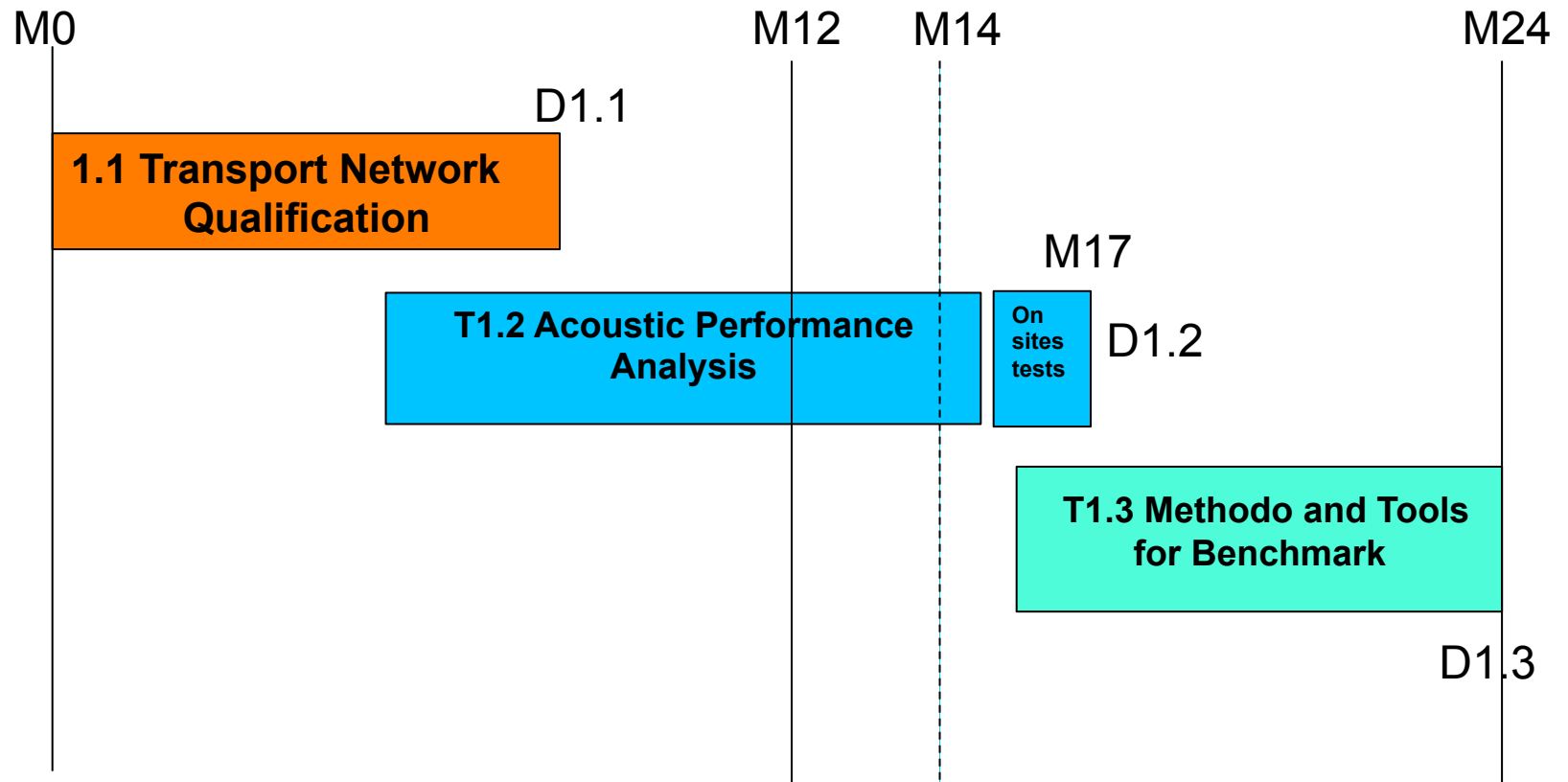
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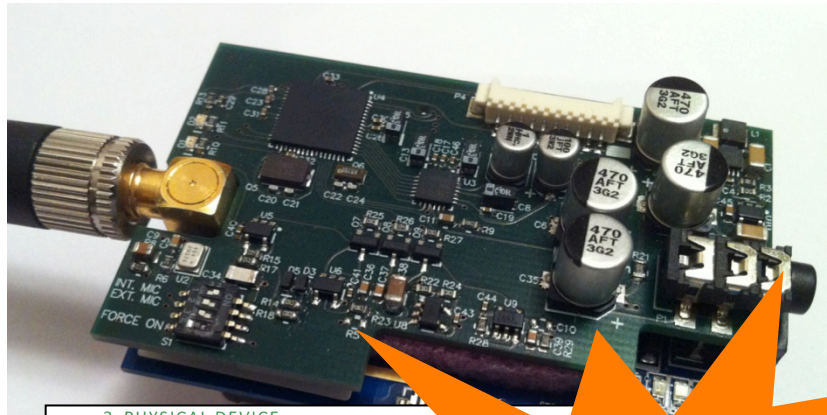
Recall WP1 objectives

- Ensure **qualification of various test beds to be able to deliver audio** data and will bring feedback to researchers on potential limitation.
- **Methodology and tools for measurements and benchmarking will be developed for use by acoustic sensors** in order to benchmark test beds and ensure reproducibility of experiments.

WP1 tasks



Deliver 2 strong assets



3 PHYSICAL DEVICE

3.1 BOARD TOP VIEW

Figure 3: Audio Board Top View

3.1.1 SELECTOR S1

- 1 – (S1A): Selects the embedded microphone as the input of the CODEC [U3]
- 2 – (S1B): Selects the embedded microphone as the input of the CODEC [U3]
- 3 – (S1C): *not defined*

Prove that simple IoT can be used for audio related apps

INDOOR EXPERIMENTS // SURVEY // **BENCHMARK** // M...

BENCHMARK PROCEDURE

BENCHMARKING TOOLS

CONCEPT

APPLICATIONS

Provide tools to check the conditions for using audio

BENCHMARKING PROCEDURE FOR AUDIO TRAFFIC MONITORING

In the context of the EAR-IT project, this section describes the development of a device test-bed based on IEEE 802.15.4 low-resource networks.

WHY DOING A BENCHMARKING PROCEDURE

The EAR-IT project as working in various scenarios (e.g. in city center and with building in Geneva) has demonstrated that nice applications can be developed using audio (e.g. traffic monitoring, security, energy efficiency, etc).

Also using advanced audio codec (i.e. speex, codec2) we have demonstrated that even constrained network using 802.15 wireless network can be used for audio applications as audio streaming (the most constrained case) can be performed with only 2kbps bandwidth which is often available on these networks.

The project has now defined the minimum conditions for any test bed to be capable of

3 scientific papers from WP1

Streaming the Sound of Smart Cities: Experimentations on the SmartSantander test-bed

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Abstract—Smart Cities have emerged as an efficient infrastructure to contribute to so-called global sensing or situation-awareness applications. One example of large scale deployment of sensors in the city is the SmartSantander test-bed. Most of the deployment so far propose traditional scalar physical measures such as temperature or humidity for a number of environment-related applications. The EAR-IT project moves a step further and proposes large-scale “real-life” experimentations of intelligent acoustics for supporting high societal value applications. One scenario that will be demonstrated is an on-demand acoustic data streaming feature for surveillance systems and management of emergencies. In this paper, we will present experimentations on streaming encoded acoustic data on low-resources devices. We will highlight the main sources of delays assuming no flow control nor congestion control to determine the best case performance level and will demonstrate that streaming acoustic data can be realized in a multi-hop manner on the SmartSantander infrastructure.

Index Terms—Smart Cities, Sensor networks, Audio streaming, surveillance

I. INTRODUCTION

In the last few years, the research efforts in the field of Wireless Sensor Networks (WSN) have shown high potentials for surveillance applications and have paved the way to nowadays so-called ubiquitous/global sensing and smart cities paradigms that extends WSN to a more generic Internet-of-Thing (IoT) concepts. A number of leading projects on global sensing and smart cities have been launched recently and the SmartSantander infrastructure [1] is probably one of the most important one in term of deployment scale and in number of hosted applications test-beds and project. One of the hosted project is the EAR-IT project [2] which focuses on large-scale “real-life” experimentations of intelligent acoustics for supporting high societal value applications and delivering new innovative range of services and applications mainly targeting to smart-buildings and smart-cities. One scenario that will be demonstrated in an on-demand acoustic data streaming feature for surveillance systems and management of emergencies. Figure 1 depicts the EAR-IT context with a 2-tier architecture of sensing nodes. The first tier consists of a limited number of powerful Acoustic Processing Units (APU) with advanced analysis capabilities to accurately detect events of interest. The second tier is composed of a large number of low-cost, low-power sensing devices, noted IoT nodes in the figure, that can be used in a complementary way to capture, on an on-demand

basis, acoustic data that will be streamed to the central control system using other IoT nodes as relay nodes.



Fig. 1. EAR-IT context on-demand audio data streaming

Although the acoustic capture system on the numerous IoT nodes are not as efficient and powerful than the one on the APU, the advantage of IoT nodes is their density that provides a large-scale coverage of the city. Therefore, in an on-demand applications, a human operator could request acoustic data from a set of IoT nodes to improve its understanding of the emergency. Note that the central control system depicted in figure 1 is actually a gateway node that manages a number of APU and IoT nodes. Many gateways are deployed across the test-bed and a gateway is connected to the Internet with a large bandwidth network technology: WiFi, wired Ethernet or 3G depending on what is available. We will then consider that the difficult part is to stream acoustic data from an IoT to its corresponding gateway, and once the data has reached the gateway, powerful and traditional streaming tool/software/protocol could be used to transfer the acoustic data to the final destination that would be typically under the supervision of a human operator.

There have been studies on multimedia sensors but few of them really consider timing on realistic hardware constraints for sending/receiving flows of packets [3], [4], [5], [6]. In this paper, we will present experimentations on streaming encoded acoustic data on low-resources devices. We will highlight

Benchmarking low-resource device test-beds for real-time acoustic data

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Abstract. The EAR-IT project relies on 2 test-beds to demonstrate the use of acoustic data in smart environments: the smart city SmartSantander test-bed and the smart building HoNet test-bed. In this paper, we take a benchmarking approach to qualify the various EAR-IT test-bed based on WSN and IoT nodes with IEEE 802.15.4 radio technology. We will highlight the main performance bottlenecks when it comes to support transmission of acoustic data. We will also consider audio quality and energy aspects as part of our benchmark methodology in order to provide both performance and usability indicators. Experimentations of multi-hop acoustic data transmission on the SmartSantander test-bed will be presented and we will demonstrate that streaming acoustic data can be realized in a multi-hop manner low-resource device infrastructures.

Key words: Smart Cities, Internet of Thing, Audio streaming

I Introduction

There is a growing interest in multimedia contents for surveillance applications in order to collect richer informations from the physical environment. Capturing, processing and transmitting multimedia information with small and low-resource device infrastructures such as Wireless Sensor Networks (WSN) or so-called Internet-of-Things (IoT) is quite challenging but the outcome is worth the effort and the range of surveillance applications that can be addressed will significantly increase. The EAR-IT project [1] is one of these original projects which focuses on large-scale “real-life” experimentations of intelligent acoustics for supporting high societal value applications and delivering new innovative range of services and applications mainly targeting to smart-buildings and smart-cities. One scenario that can be demonstrated is an on-demand acoustic data streaming feature for surveillance systems and management of emergencies. Other applications such as traffic density monitoring or ambulance tracking are also envisioned and are also requiring timely multi-hop communications between low-resource nodes. The EAR-IT project relies on 2 test-beds to demonstrate the use of acoustic data in smart environments: the smart city SmartSantander test-bed and the smart building HoNet test-bed.

Real-time on-demand multi-hop audio streaming with low-resource sensor nodes

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Abstract—Smart Cities have emerged as an efficient infrastructure to contribute to so-called global sensing or situation-awareness applications. One example of large scale deployment of sensors in the city is the SmartSantander test-bed. Most of the deployment so far propose traditional scalar physical measures such as temperature or humidity for a number of environment-related applications. The EAR-IT project moves a step further and proposes large-scale “real-life” experimentations of intelligent acoustics for supporting high societal value applications. One scenario that will be demonstrated is an on-demand acoustic data streaming feature for surveillance systems and management of emergencies. In this paper, we will present experimentations on streaming encoded acoustic data on low-resources devices. We will highlight the main sources of delays assuming no flow control nor congestion control to determine the best case performance level and will demonstrate that streaming acoustic data can be realized in a multi-hop manner on the SmartSantander infrastructure.

Index Terms—Smart Cities; Sensor networks; Internet of Thing; Audio streaming; Surveillance

I. INTRODUCTION

In the last few years, there is a growing interest in multimedia contents, such as images and acoustics, for surveillance applications in order to collect richer informations from the physical environment. Capturing, processing and transmitting multimedia information with small and low-resource infrastructures such as wireless sensor networks (WSN) is quite challenging but the outcome is worth the effort and the range of surveillance applications that can be addressed with WSN will significantly increase. For instance, so-called ubiquitous/global sensing and smart cities infrastructures could benefit from such multimedia support to go beyond the traditional scalar data approach (luminosity, humidity, temperature to name a few) thus enabling new forms of interactions and decision-making. The SmartSantander infrastructure [1] is probably one of the most important test platform in terms of deployment scale and in number of hosted applications test-beds and projects. One of the hosted project is the EAR-IT project [2] which focuses on large-scale “real-life” experimentations of intelligent acoustics for supporting high societal value applications and delivering new innovative range of services and applications mainly targeting to smart-buildings and smart-cities. One scenario that will be demonstrated is an on-demand acoustic data streaming feature for surveillance systems and management of emergencies. Figure 1 depicts the EAR-IT context with a 2-tier architecture of sensing nodes.

The first tier consists of a limited number of powerful Acoustic Processing Units (APU) with advanced analysis capabilities to accurately detect events of interest. The second tier is composed of a large number of low-cost, low-power sensing devices, noted IoT nodes in the figure, that can be used in a complementary way to capture, on an on-demand basis, acoustic data that will be streamed to the central control system using other IoT nodes as relay nodes. Delay can be an important factor as the on-demand scenario is typically intended for a human operator requesting acoustic data on well-identified parts of the city.



Fig. 1. EAR-IT context on-demand audio data streaming

Although the acoustic capture system on the numerous IoT nodes are not as efficient and powerful than the one on the APU, the advantage of IoT nodes is their density that provides a large-scale coverage of the city. Therefore a human operator could request acoustic data from a set of IoT nodes to improve its understanding of the emergency. Note that the central control system depicted in figure 1 is actually a gateway node that manages a number of APU and IoT nodes. Many gateways are deployed across the test-bed and a gateway is connected to the Internet with a large bandwidth network technology: WiFi, wired Ethernet or 3G depending on what is available. We will then consider that the difficult part is to stream acoustic data from an IoT to its corresponding gateway, and once the data has reached the gateway, powerful and traditional streaming tool/software/protocol could be used to transfer the acoustic data to the final destination.

Streaming the sound of smart cities: experimentation on the smartSantander test bed

things August 13

Benchmarking low-resource device test-beds for real-time acoustic data

Tridentcom May 2014

Real-time on demand multi-hop audio streaming with low-resources sensor

LCN Sept 2014

SHOUTING FOR HELP

How to use the IoT to detect people shouting



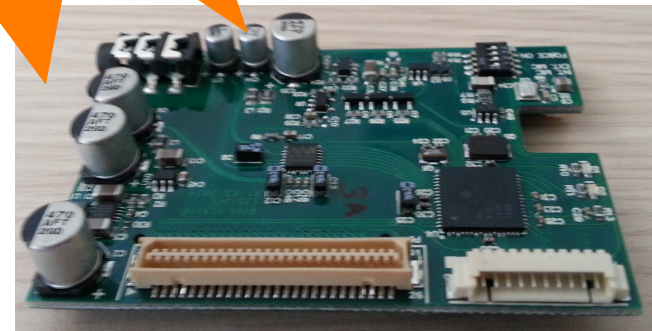
Demonstrate use of full chain on using simple IoT audio capabilities (audio streaming)

SHOUTING DETECTION

The Acoustic Processing Unit

GENERATED ACTIONS

Audio Recording & Streaming

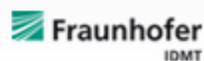




the sounds of smart environment



WP1.2 Minimum requirements for use of acoustic sensors



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SmartSensingStars

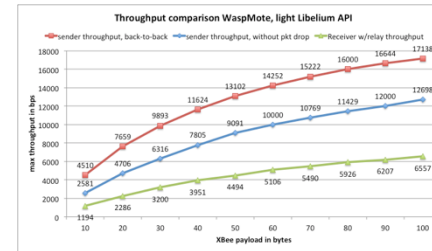
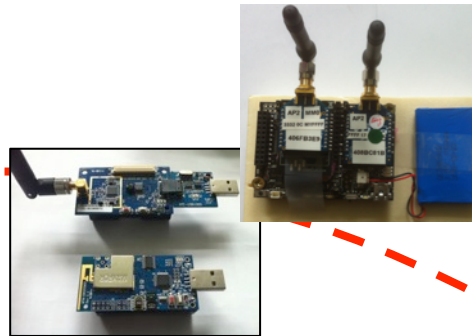
Acoustic data in low-resource IoT networks



PLAY/STORE RECEIVED AUDIO DATA

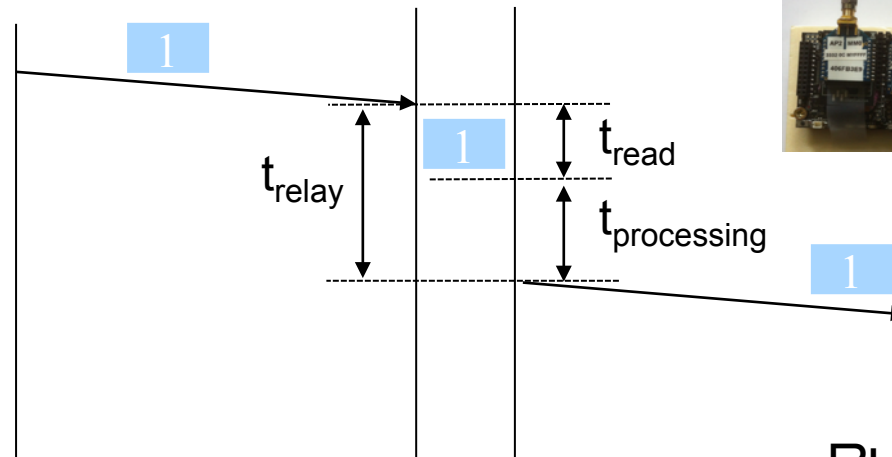
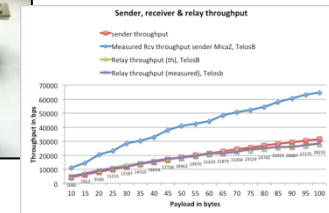
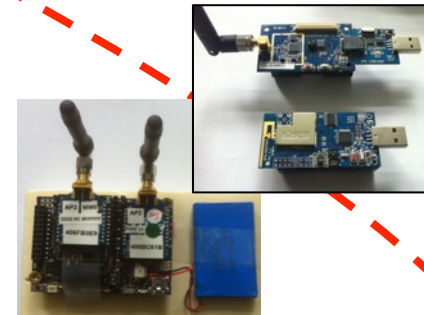


Multi-hop audio constraints

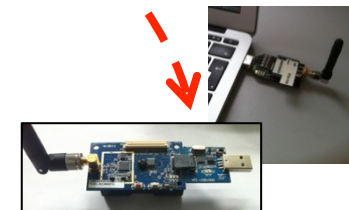


RELAY

RELAY



PLAY/STORE RECEIVED AUDIO DATA



Objectives of WP1.2

- Presents for some selected performance indicators the minimum requirements for use of acoustic sensors on the various EAR-IT testbeds
- Santander's SmartSantander test-bed
- Geneva's Hobnet test-bed

- Need to define and determine performance indicators
 - IoT node performance indicators
 - Network performance indicators
- Quality and usability indicators are also necessary
 - Audio quality indicators
 - Energy indicators

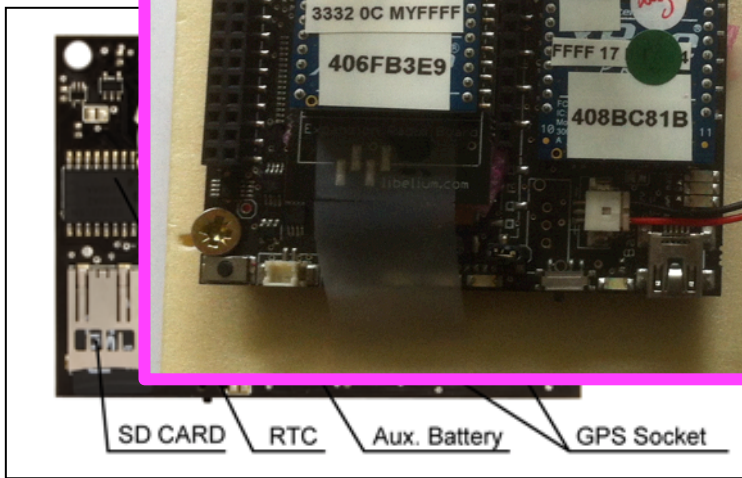
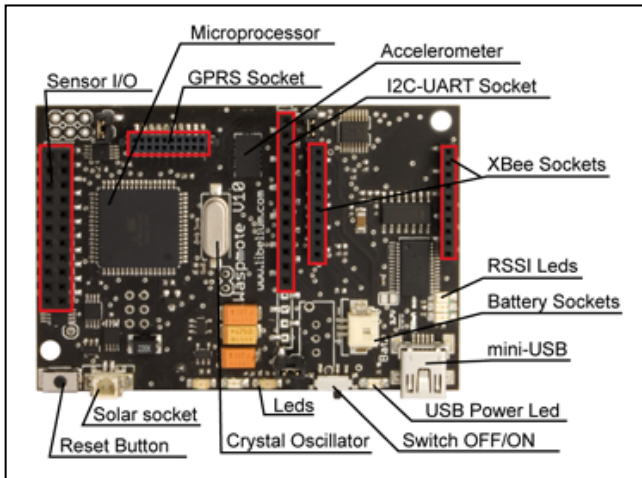
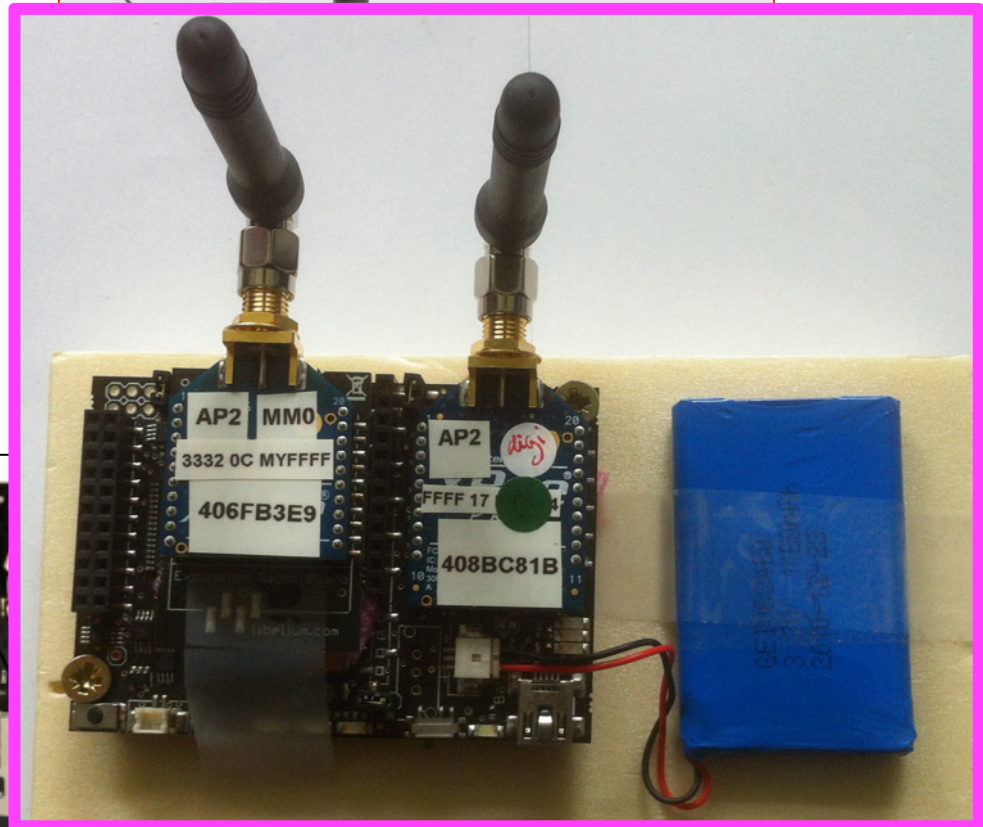
Indicators

- NETWORK performance indicators
 - At audio source (sending time)
 - At relay nodes (relaying time, buffer)
- AUDIO quality indicators
 - Supported packet loss rate
- ENERGY consumption indicators
 - Node lifetime

- Presentation of the available hardware
- Presentation of the implemented acoustic solutions
- Presentation of minimum requirements
- Conclusions

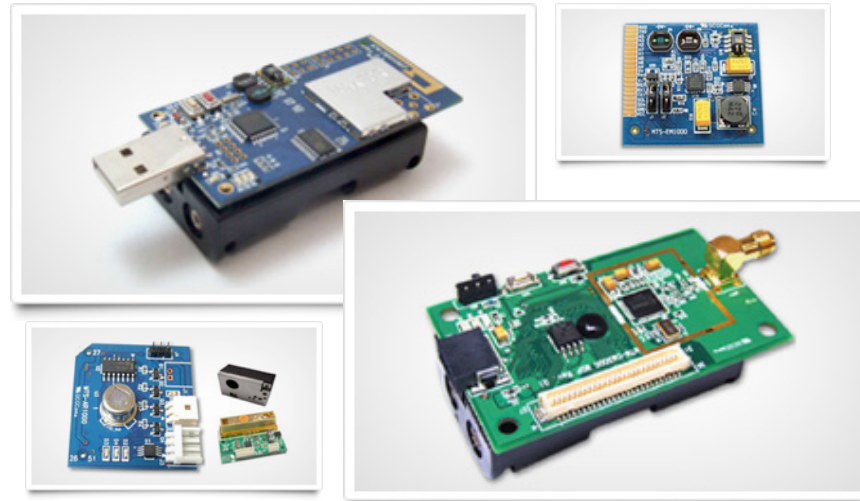


ATmega1281 microcontroller
8Mhz, 4K RAM & 2G SD card.
2.4GHz IEEE 802.15.4 XBee
Libelium API v031

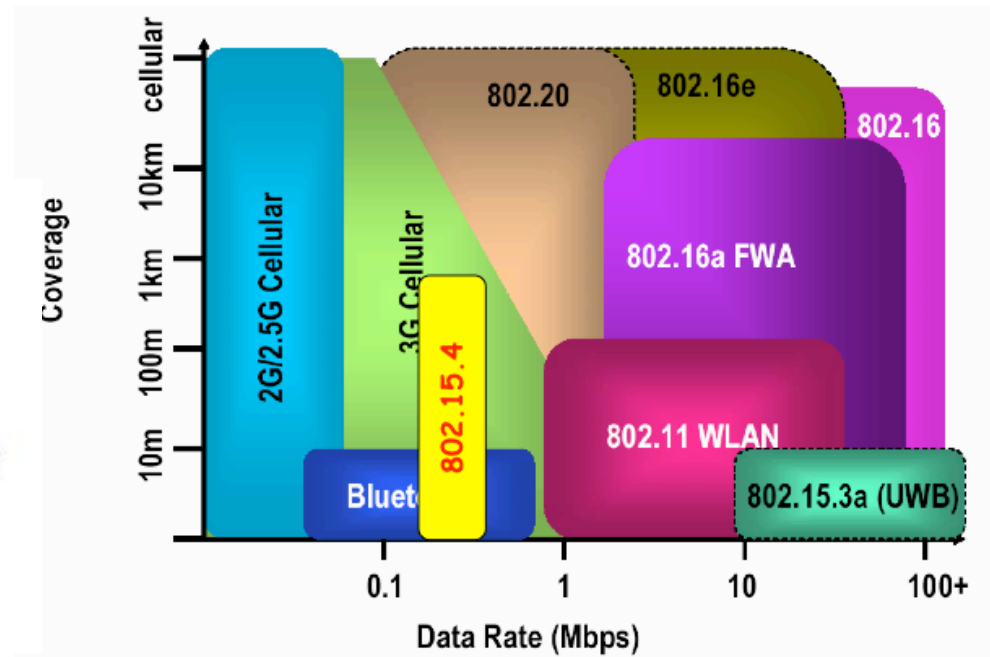




MSP430F1611 microcontroller
8Mhz, 48K flash, 10K RAM
2.4GHz IEEE 802.15.4 CC2420
Programmed under TinyOS
Similar to TelosB



- Low-power radio in the 2.4GHz band offering 250kbps throughput at physical layer
- Power transmission from 1mW to 100mW for range from 100m to about 1km is LOS
- CSMA/CA (beacon & non beacon)

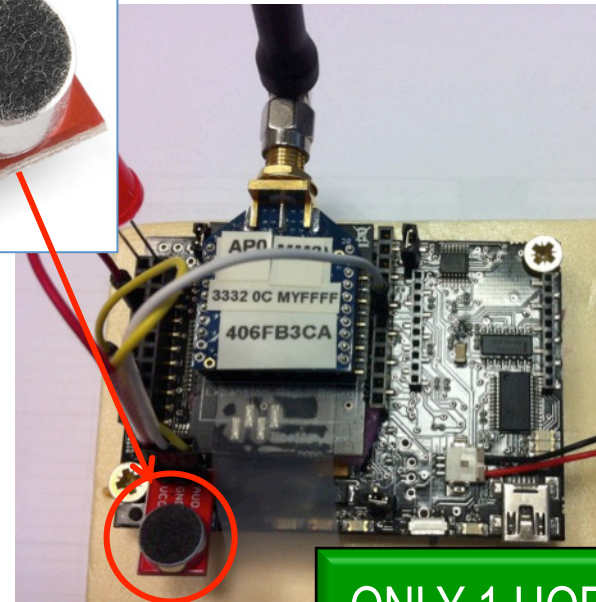
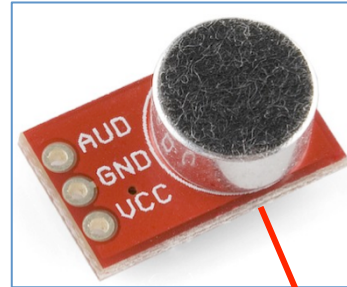


Adding audio capabilities to IoT

- Raw audio, non compressed
 - simple to implement
 - small processing overhead
 - ...but bandwidth consuming
- Compressed audio
 - high processing overhead
 - need additional hardware
 - ...but low bit-rate codecs are available

Raw audio on WaspMote

- Electret mic with amplifier
- 8-bit 4Khz sampling gives 32000bps
- 8Khz sampling gives 64000bps
- Host microcontroller responsible for sampling & data processing
- XBee in AP0 mode (transparent mode)

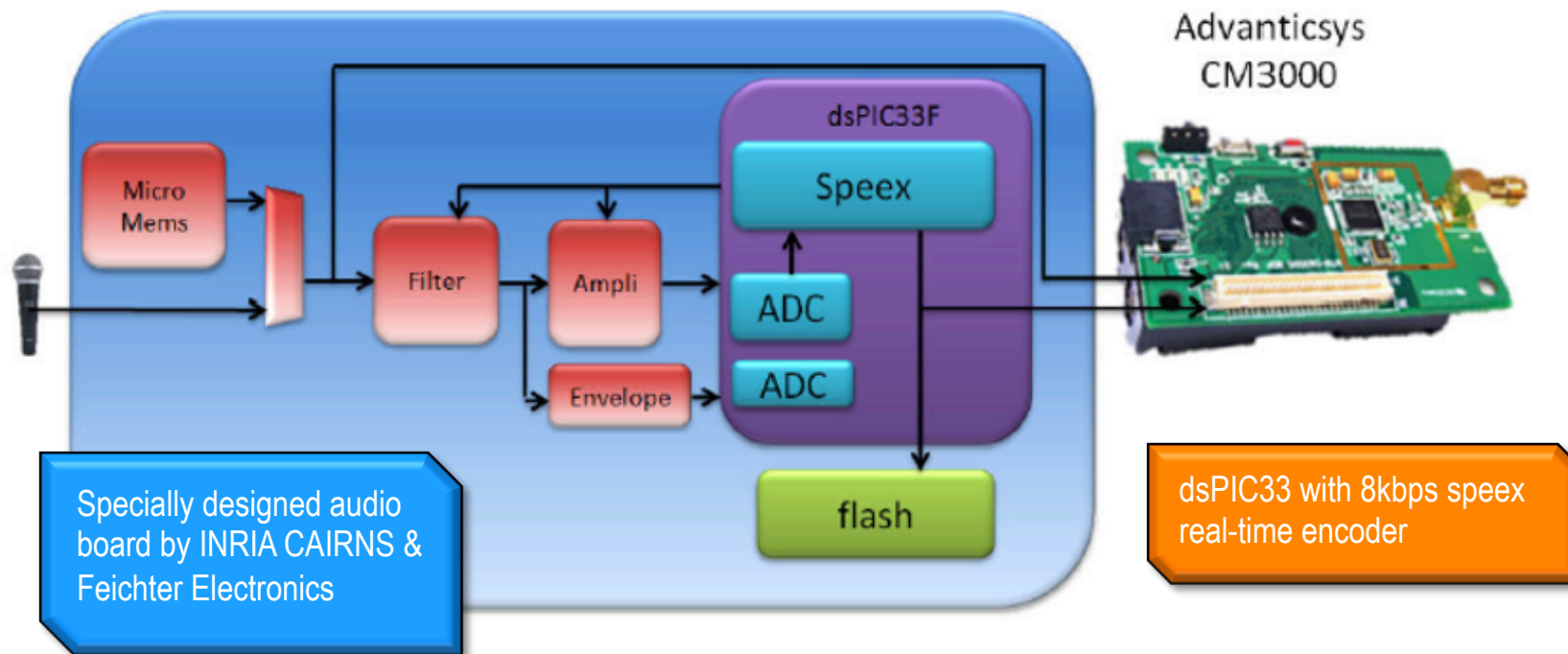


ONLY 1 HOP!

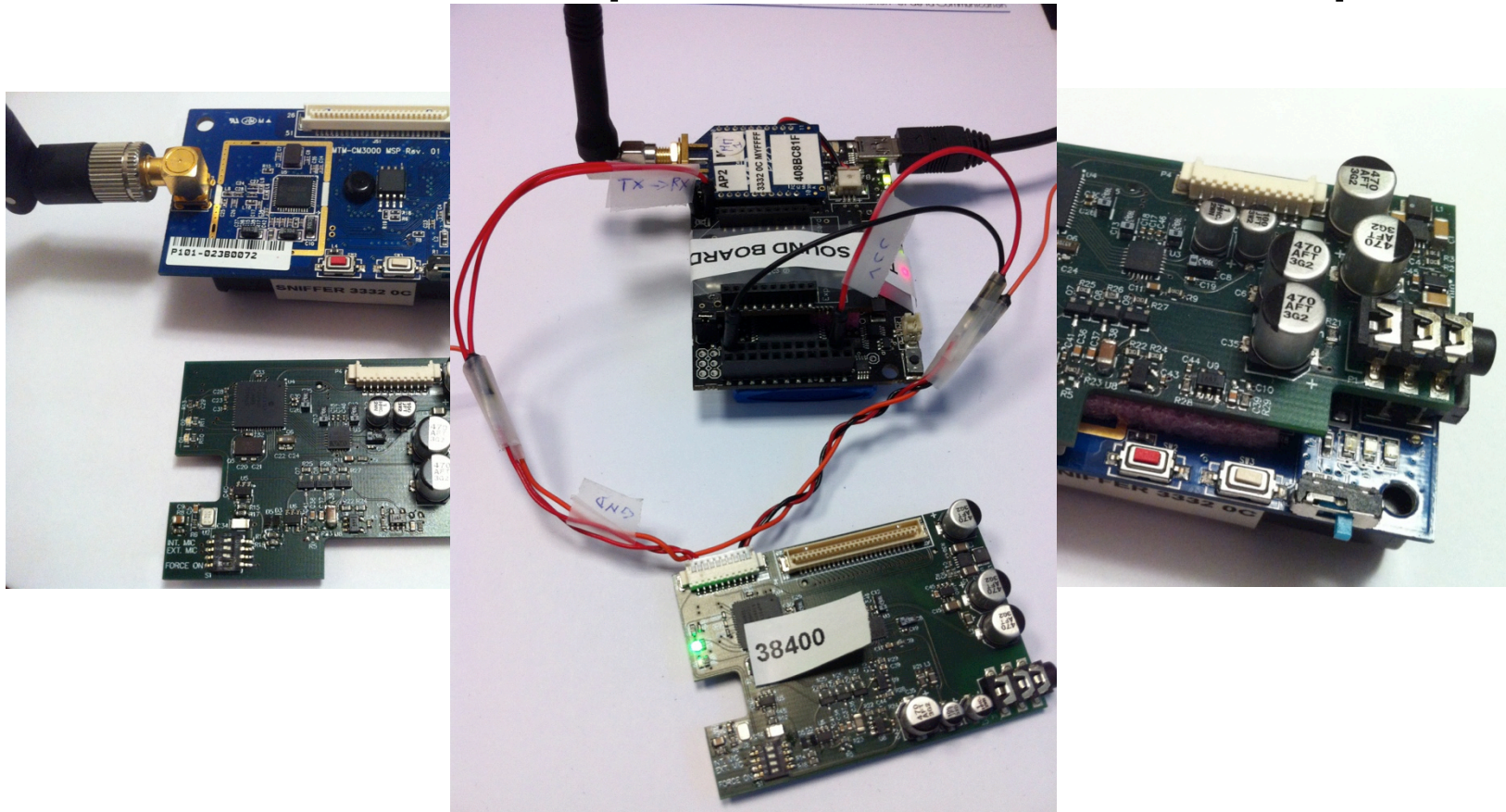


Compressed audio

- Develop an audio board to sample and encode audio in real-time



- Encode with Speex codec at 8kbps



Audio board software

- The audio board is fully reprogrammable (dsPIC33)
- The audio source can be controlled wirelessly with text-based message
 - "@/A" for aggregation mode
 - "@/D" to set destination address
 - "@/C" to start/stop audio capture

A1/2/3/4 aggregate audio frames

D0013A2004086D828 set the 64-bit dest. mac addr

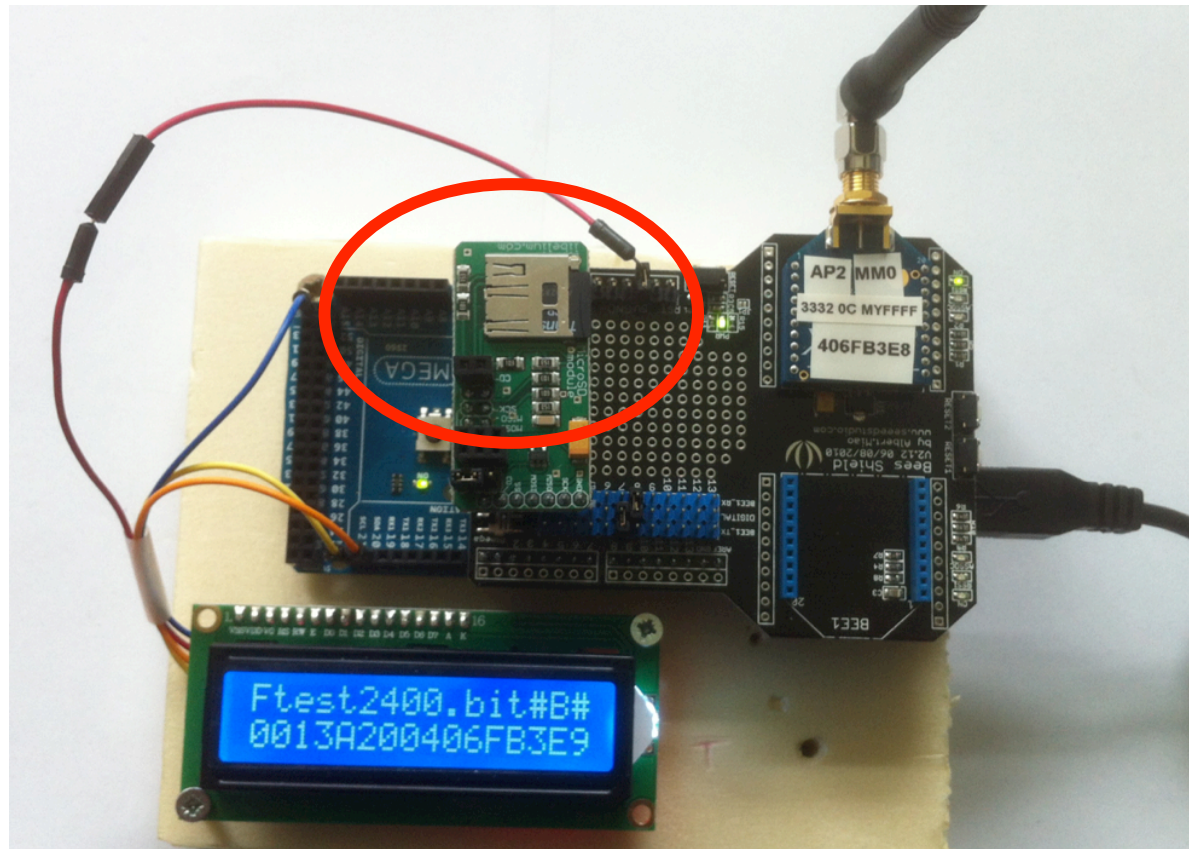
D0080 set the 16-bit dest. mac addr

C0/1 power off/on the audio board

- Test other codecs such as Codec2

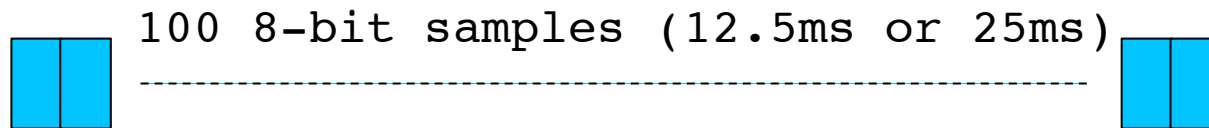
Fully configurable:

File to send
 Size of packet chunk
 Inter-packet delay
 Image/Binary mode
 Destination node
 Clock synchronization

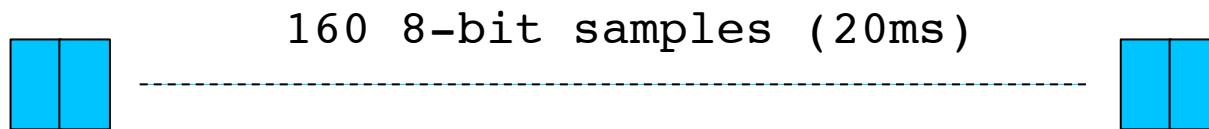


Minimum requirements at sender

- Raw audio at 4kHz and 8kHz



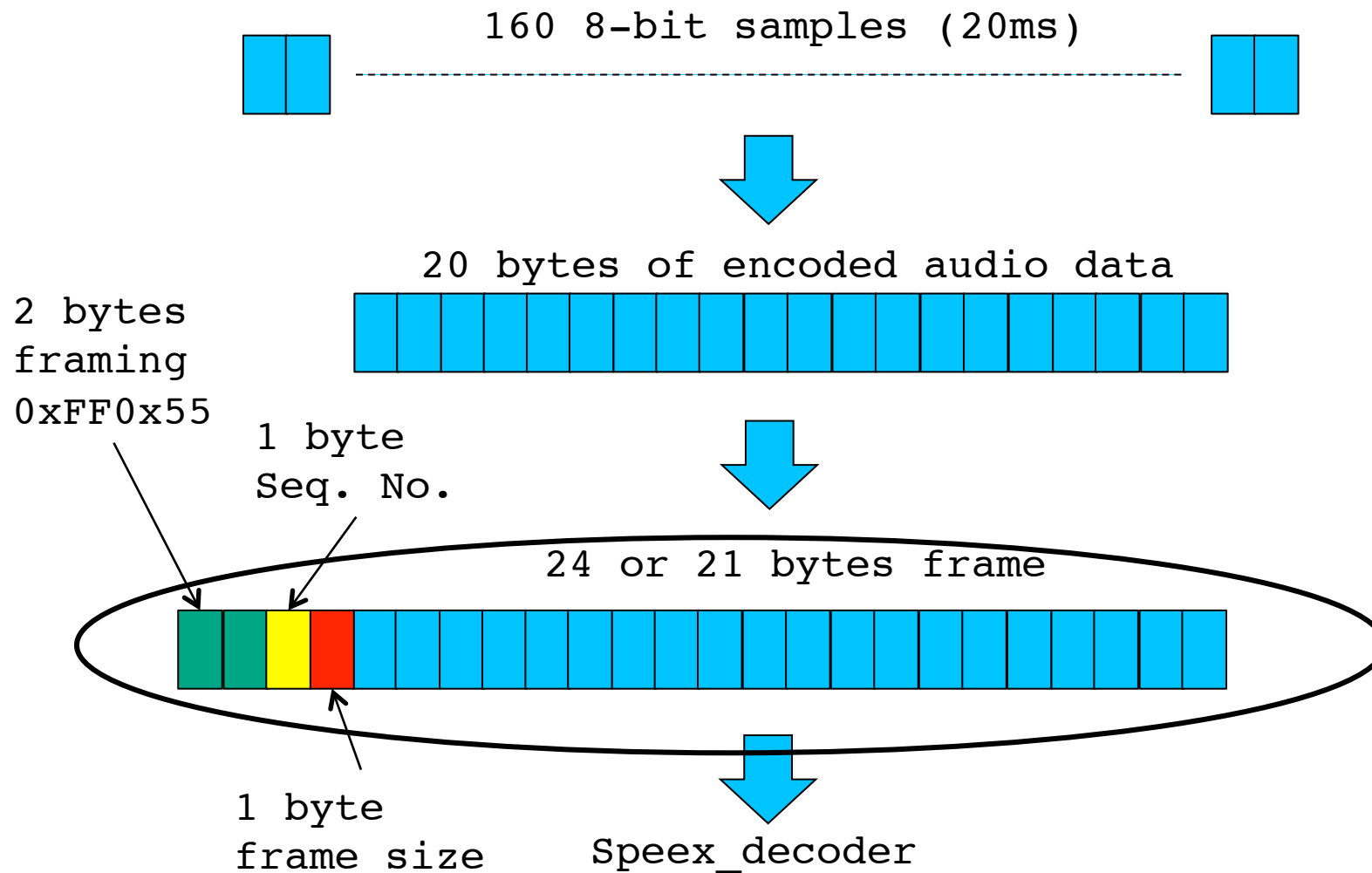
- Speex audio & Codec2 audio



Between 6 and 20 bytes of encoded audio data

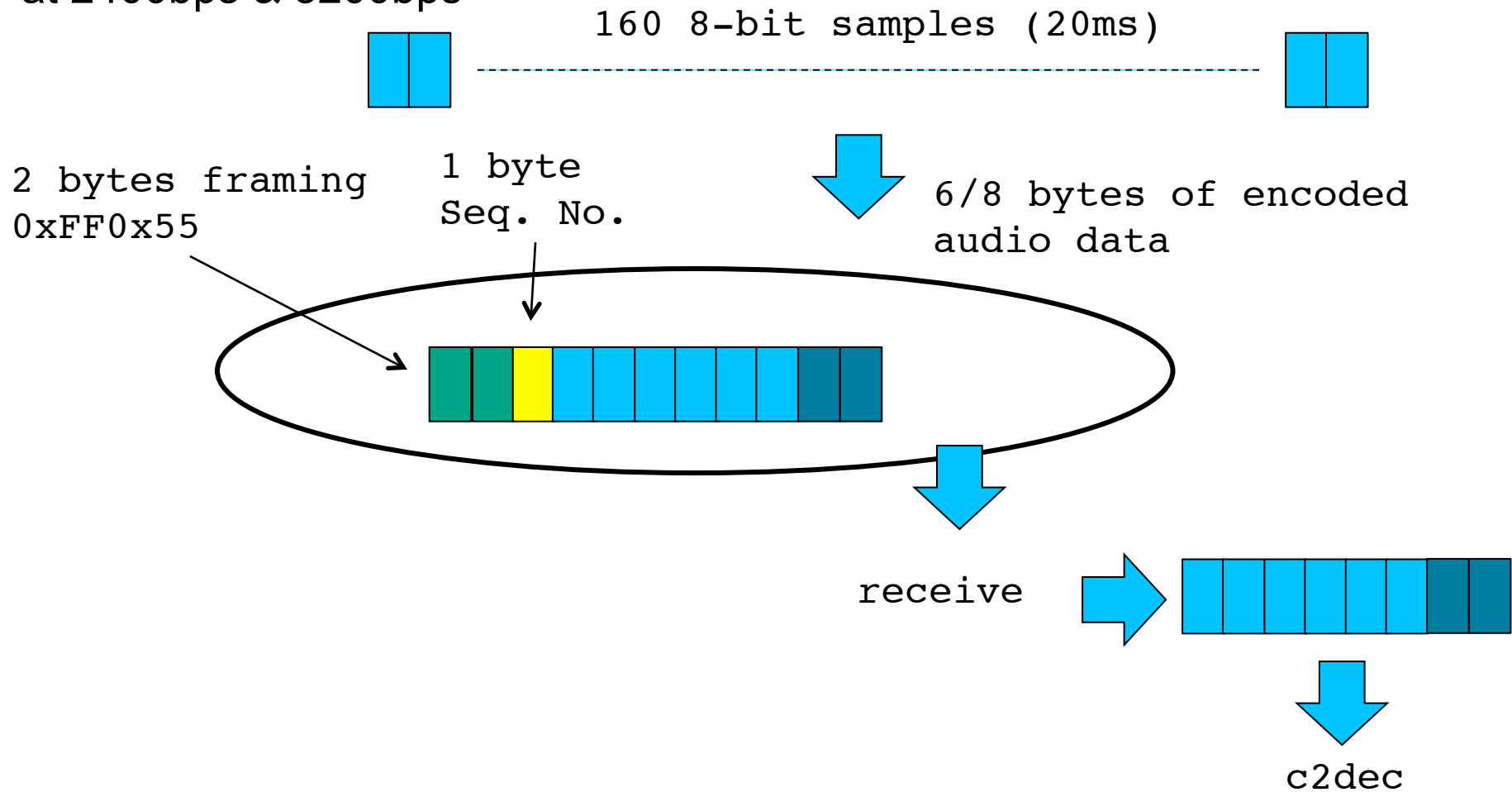


speex at 8kbps



Codec2 at 2400 & 3200bps

at 2400bps & 3200bps



Minimum requirements

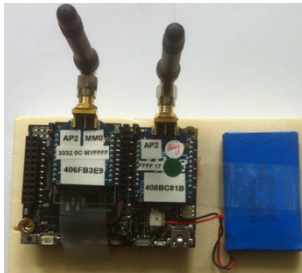
Sender: sending time

Relay: relaying time

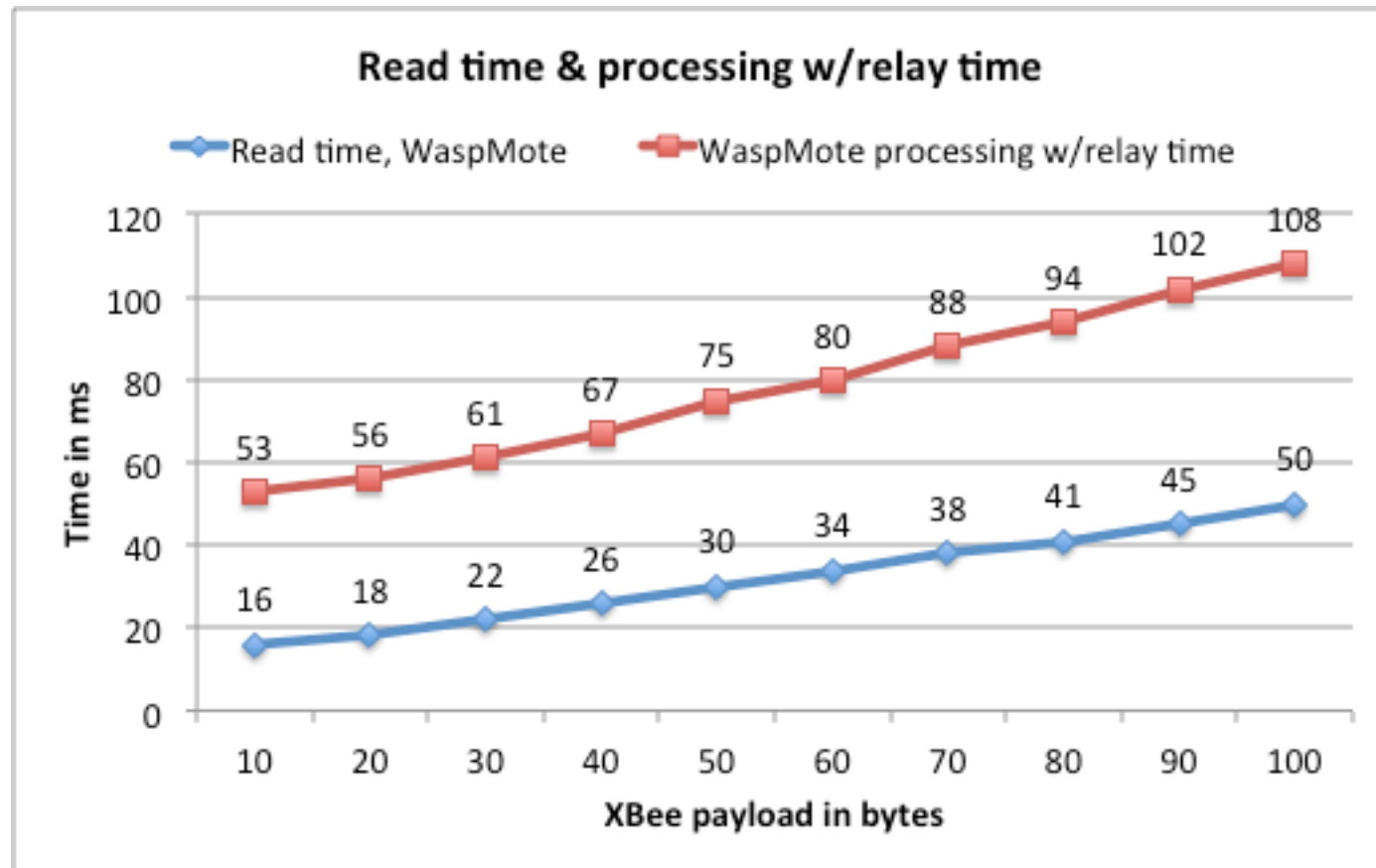
Audio frame aggregation can be performed to increase the time window for sending or relaying

Codec	Minimum sending rate
Raw 4KHz 8KHz	100 bytes every 25ms 100 bytes every 12.5ms
Speex 8000bps A1 A2 A3 A4	24 bytes every 20ms 48 bytes every 40ms 72 bytes every 60ms 96 bytes every 80ms
Codec2 2400bps A1 . . An (1≤n≤11) 3200bps A1 . . An (1≤n≤9)	9 bytes every 20ms . . 9*n bytes every n*20ms 11 bytes every 20ms . . 11*n bytes every n*20ms

Measured relay node performances



LIBELIUM
WASPMOTE

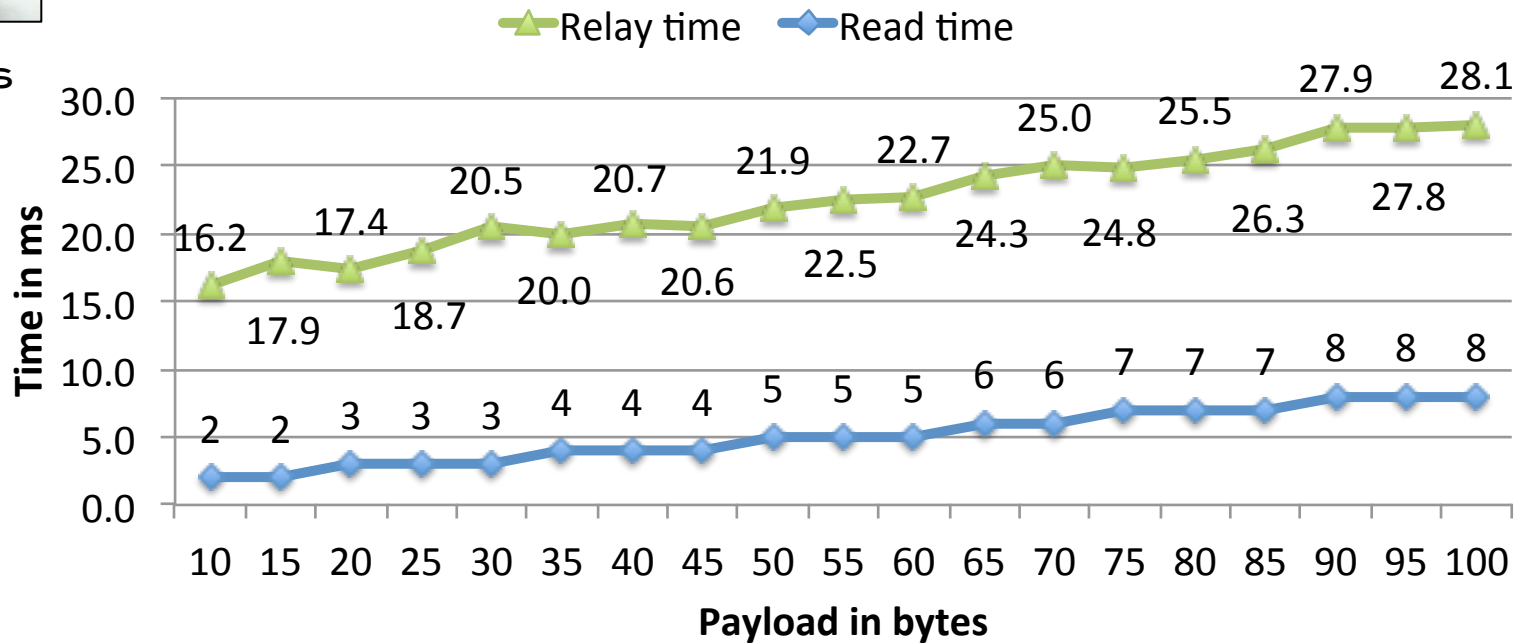


Measured relay node performances

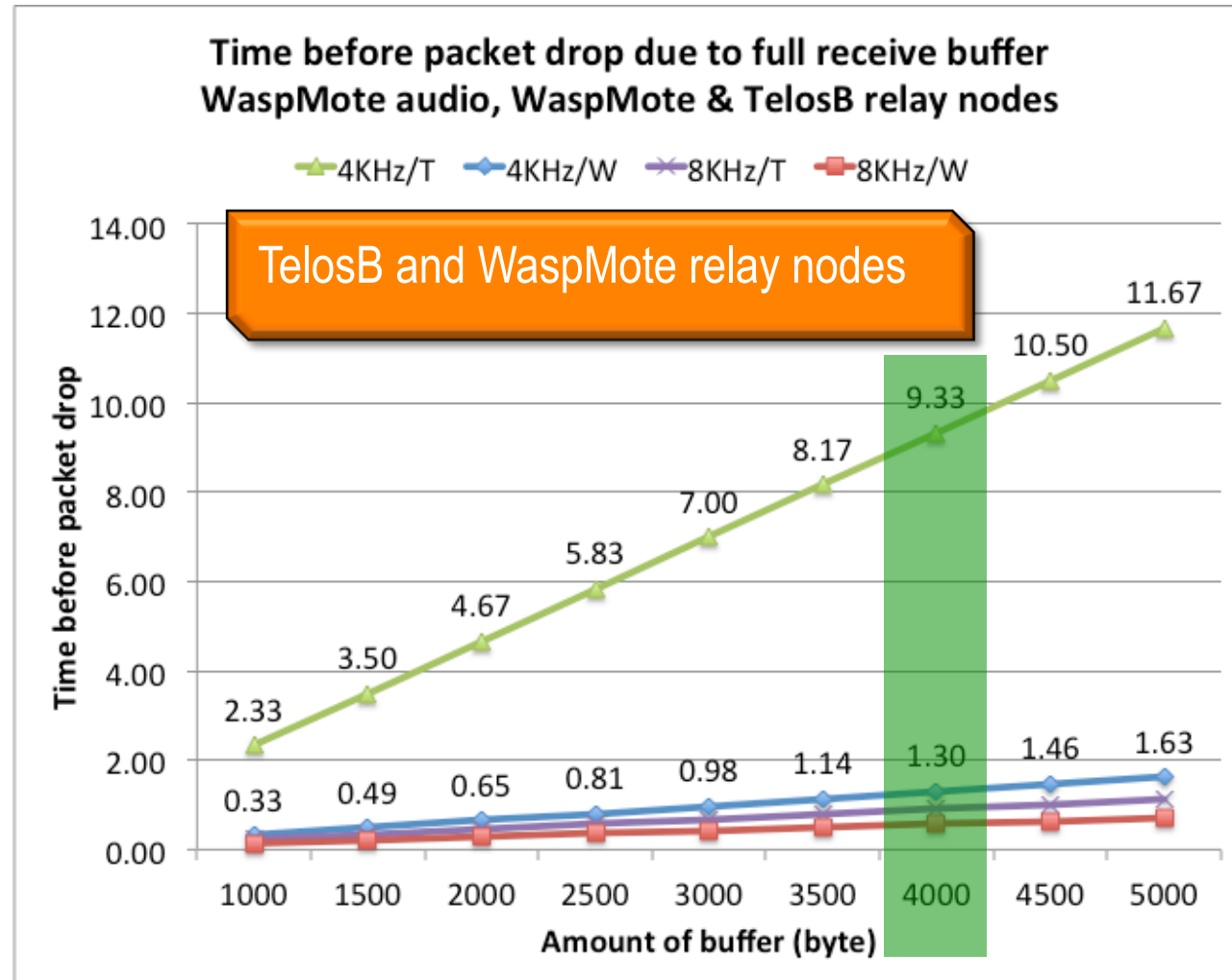
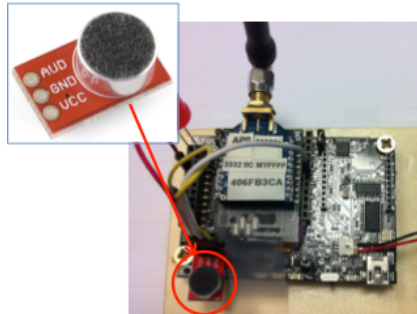


ADVANTICSYS
TELOS B

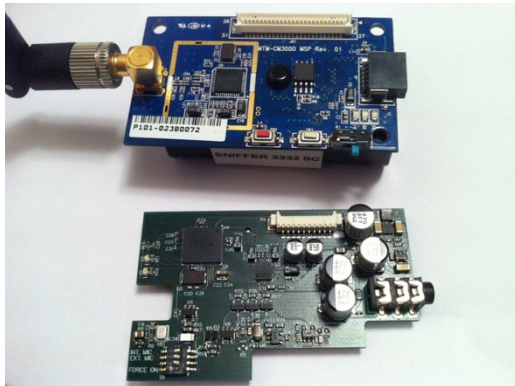
Pkt read time & Pkt relay time, TelosB



Buffer requirements at relay

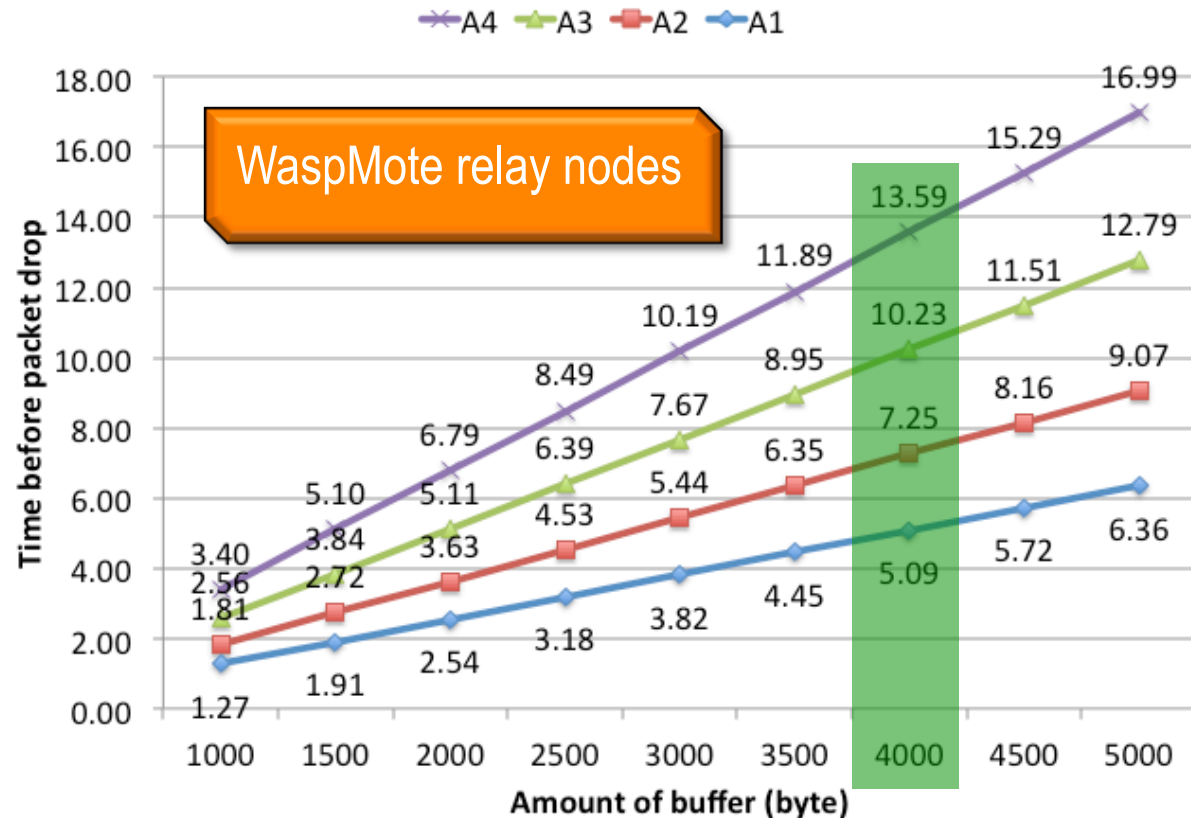


Buffer requirements at relay

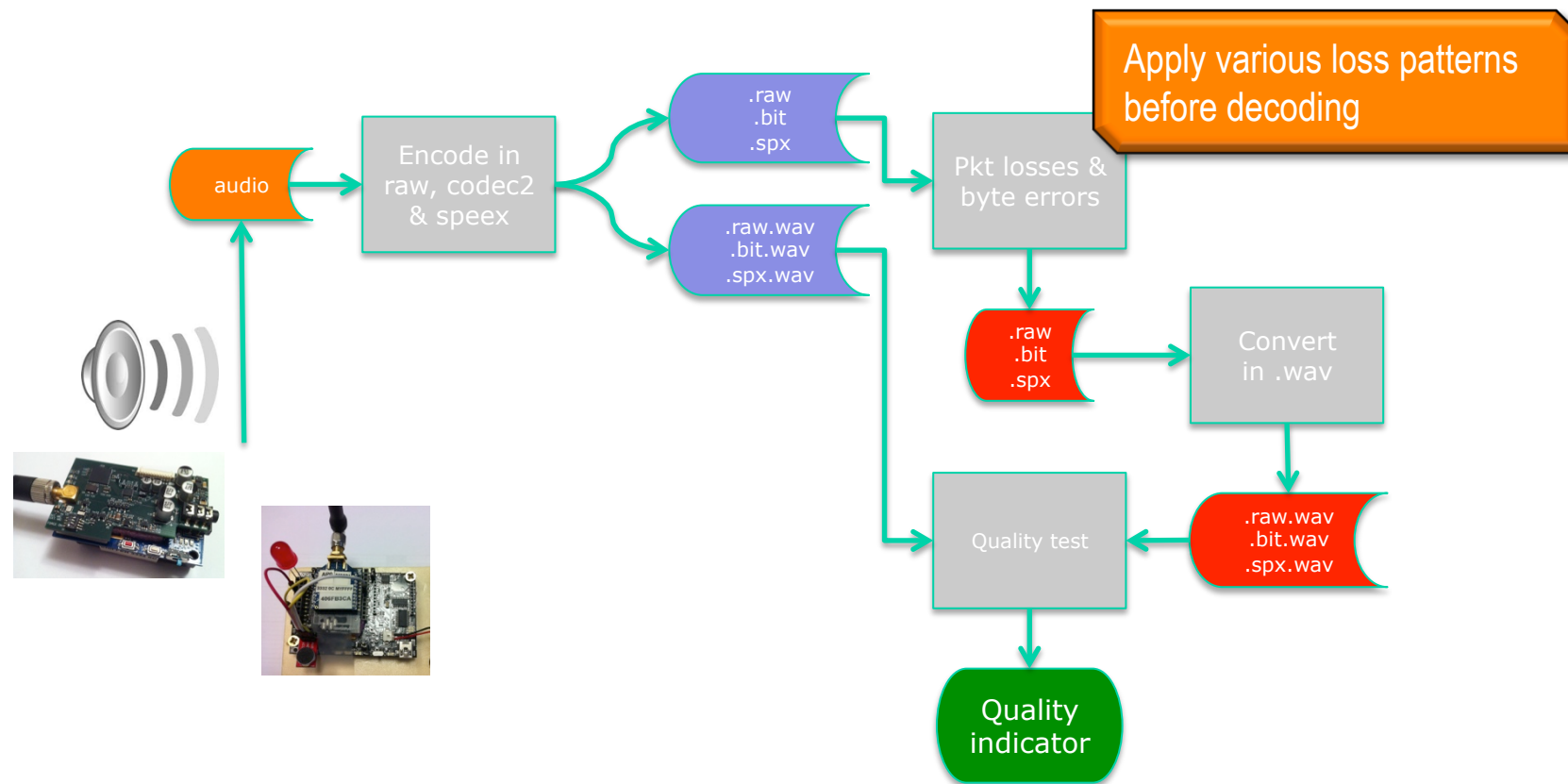


With TelosB relay node, compressed audio relaying can be performed without buffering needs

Time before packet drop due to full receive buffer
TelosB audio board, WaspMote relay node



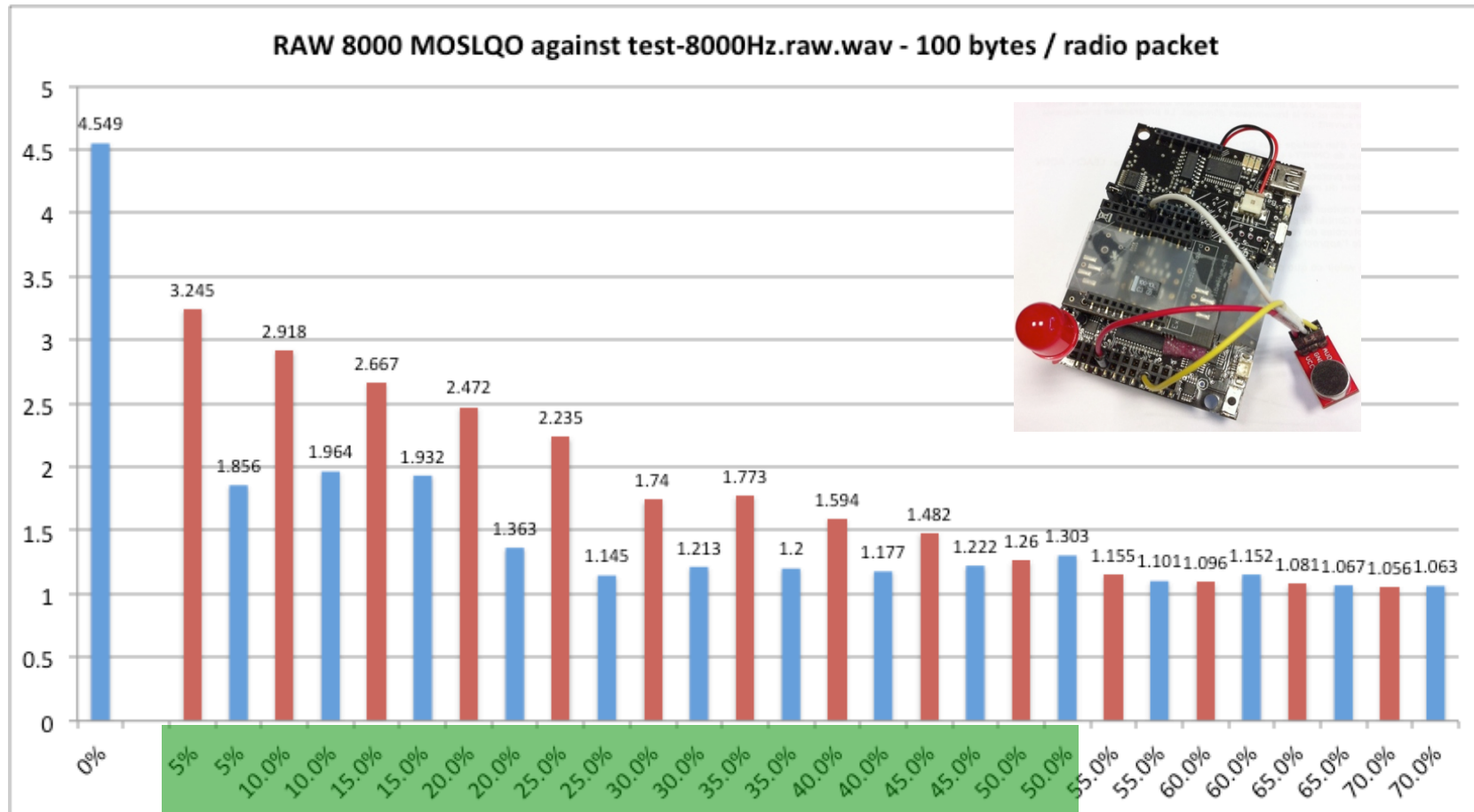
Impact of packet losses



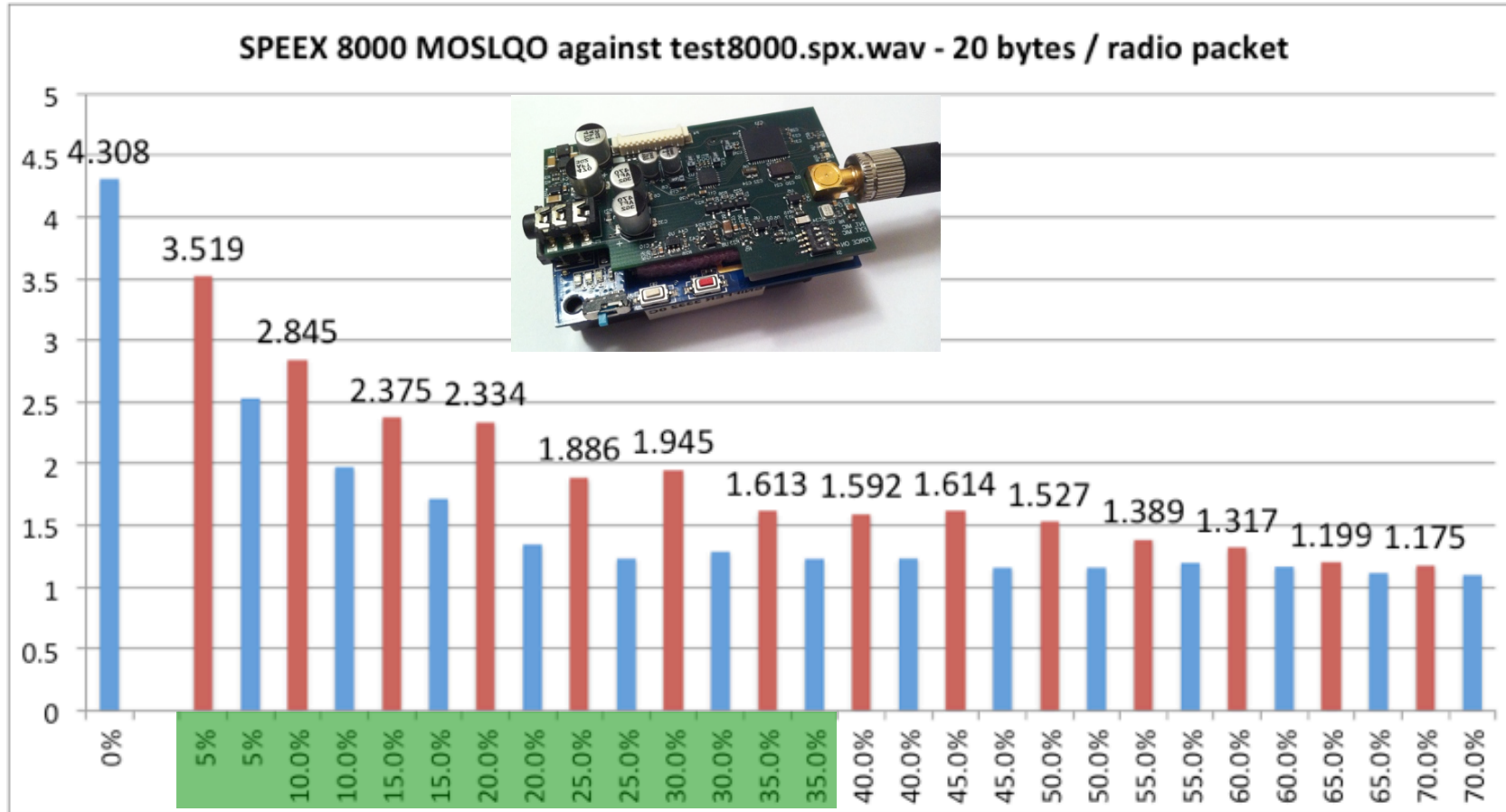
Audio quality: PESQ & MOS (1)

- ITU-T P.862 Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs.
- We can use ITU-T PESQ tool to determine the MOS value for loss-free encoded audio (codec2, speex, ...). MOS-LQO values greater than 2.6 are considered good.

Test8000.raw



Test8000.spx, 20B/pkt (A1)



Maximum supported pkt loss rate

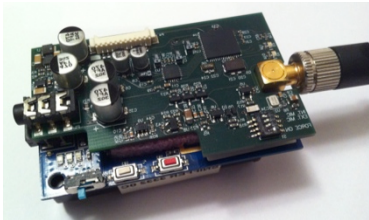
Codec	Maximum packet loss rate for speech understanding
Raw 4KHz & 8KHz	50%
Speex 8000bps	35%
Codec2	
2400bps	20%
3200bps	30%

Conclusions on 1.2

- Low-resource devices (sensor, IoT, ...) are currently deployed in a number of projects, especially in SmartCities context
- The EAR-IT project focuses on acoustic data, deployed on large scale test-beds
- We define performance indicators as well as quality and usability indicators for use of acoustic sensors
- Minimum requirements for NETWORK and AUDIO have been determined according to available hardware and implemented acoustic solutions
- In next task WP1.3, we will present the benchmark methodology to verify that a given test-bed can support these minimum requirements
- ENERGY indicators will be also detailed in WP1.3.

2-hop demo w/audio board

0x0090



SPEEX AUDIO ENCODING
8KBPS

A1/2/3/4/6 aggregate audio frames
D0020 set the 16-bit dest. mac addr
C0/1 power off/on the audio board

python script,
standard Unix scripting tool

speex decoding tool from
open-source speex distrib



```
python 115200SerialToStdout.py | speex_sampledec_wframing | play --buffer 100 -t raw -r 8000 -s -2 -
```

0x0020

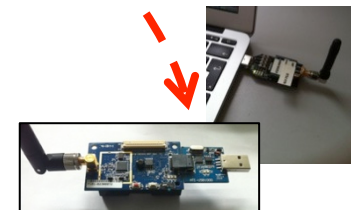


D0100 set the 16-bit
dest. mac addr

RELAY

DECODE & PLAY
RECEIVED AUDIO

0x0100





the sounds of smart environment



Questions ?