EAR-IT network qualification

Qualify and Benchmark Test-beds for Acoustics in Deployment of Targeted Applications

RESSACS 2013, UBO

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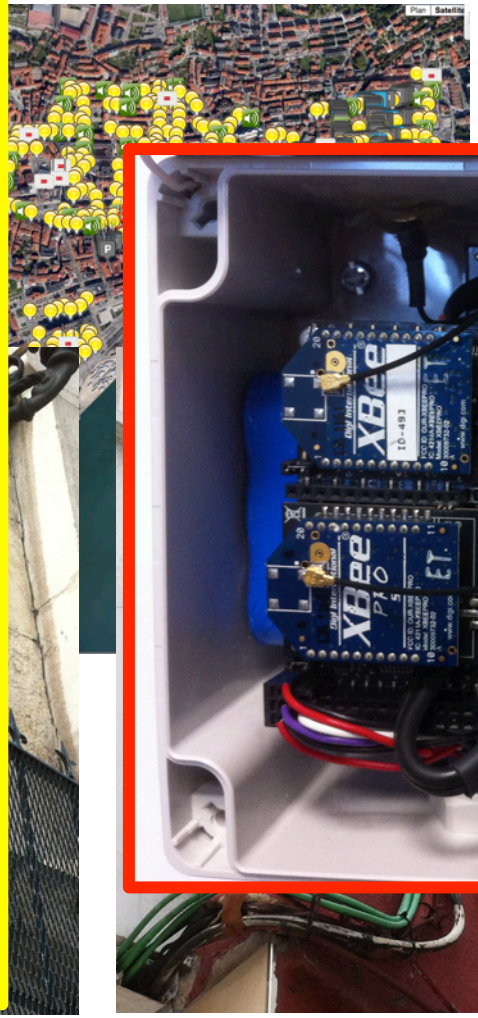
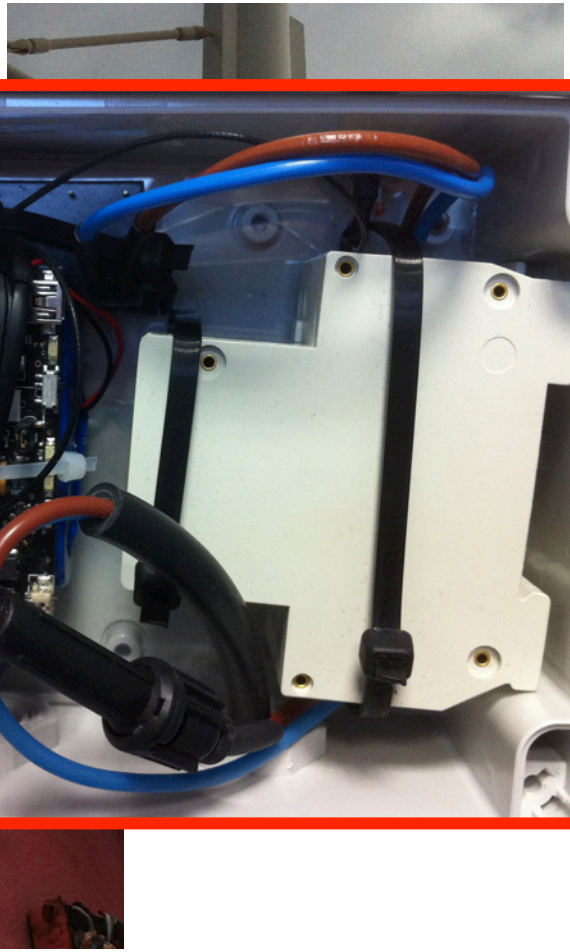
SmartSantander test-bed





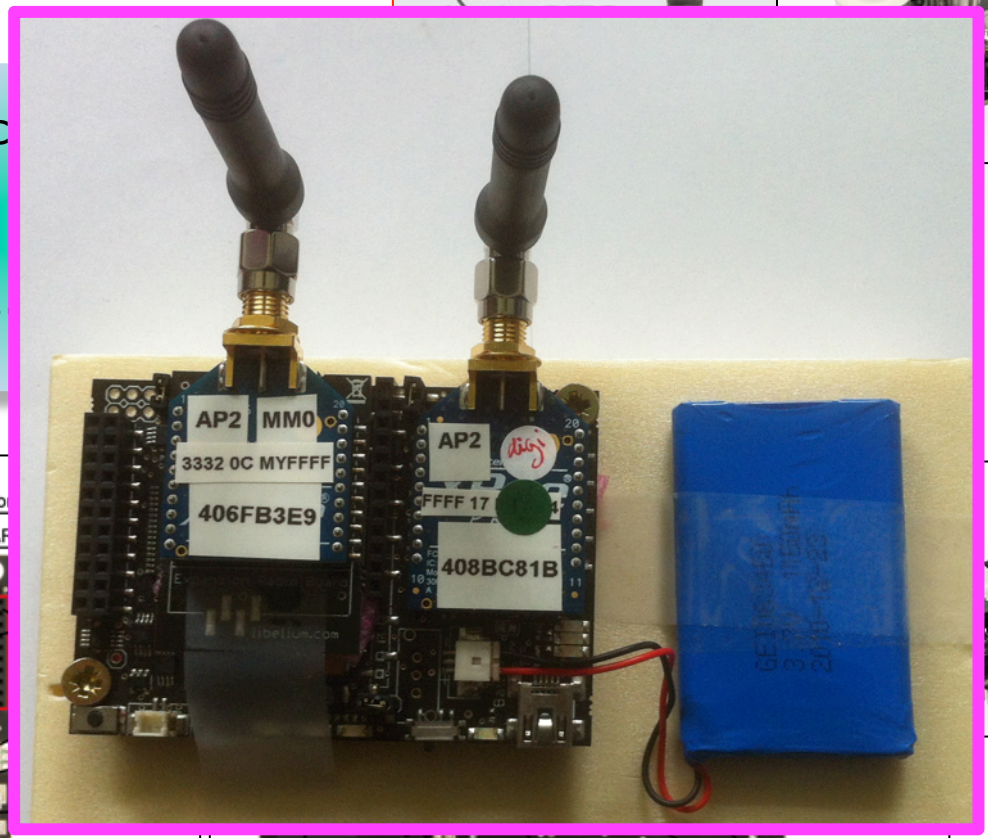
SmartSantander IoT node

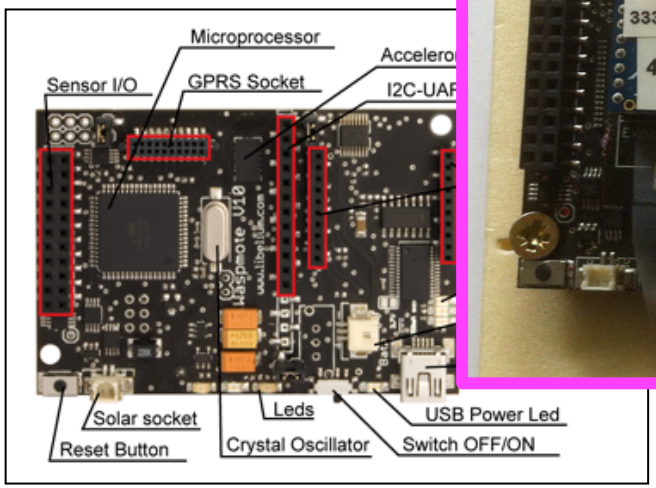
libelium
comunicaciones inalámbricas distribuidas

ATmega1281 mic

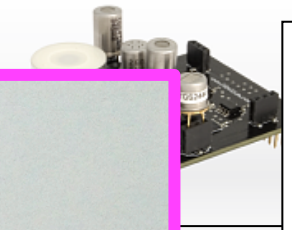
8Mhz, 4K RAM &

2.4GHz IEEE 802





Gases



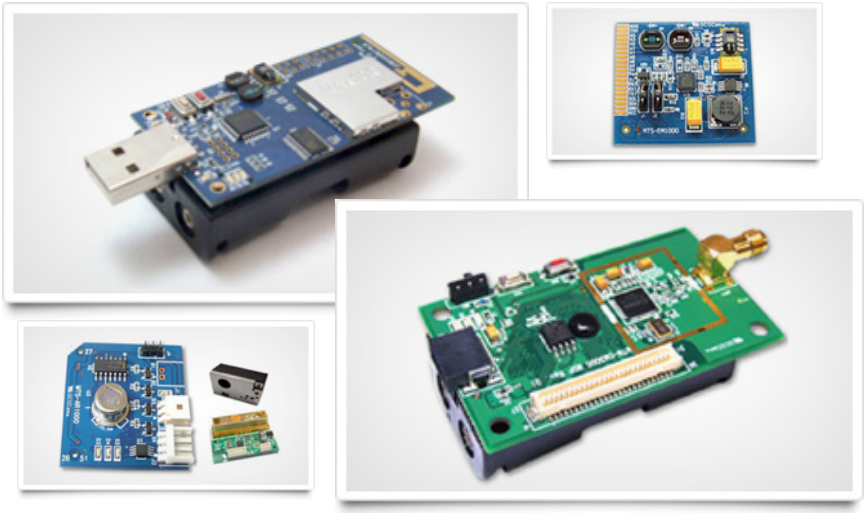
- Carbon Monoxide – CO
- Carbon Dioxide – CO2
- Oxygen – O2
- Methane – CH4
- Hydrogen – H2
- Ammonia – NH3
- Isobutane – C4H10
- Ethanol – CH3CH2OH
- Toluene – C6H5CH3
- Hydrogen Sulfide – H2S
- Nitrogen Dioxide – NO2
- Temperature
- Humidity

- Pressure/Weight
- Bend
- Vibration
- Impact
- Hall Effect
- Tilt
- Temperature (+/-)
- Liquid Presence
- Liquid Level
- Luminosity
- Presence (PIR)
- Stretch

the sounds of smart environments

IMAGES ARE FROM LIBELIUM COMPANY

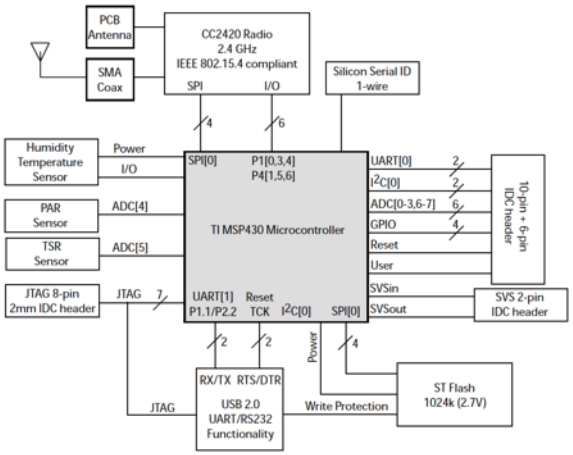
HobNet test-bed at UNIGE



MSP430F1611 microcontroller

8Mhz, 48K flash, 10K RAM

2.4GHz IEEE 802.15.4 CC2420

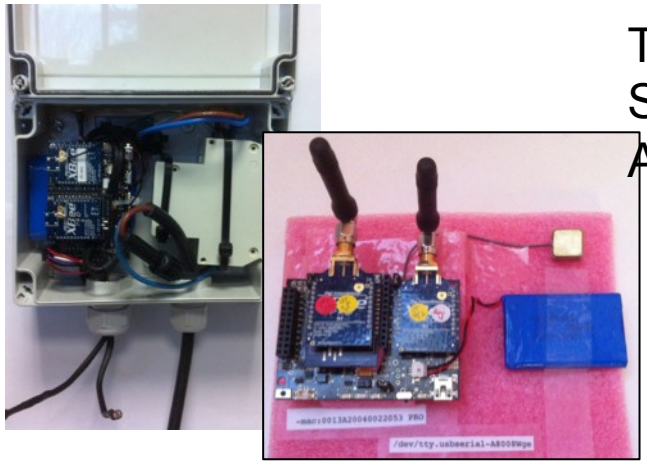


Purpose of network qualification

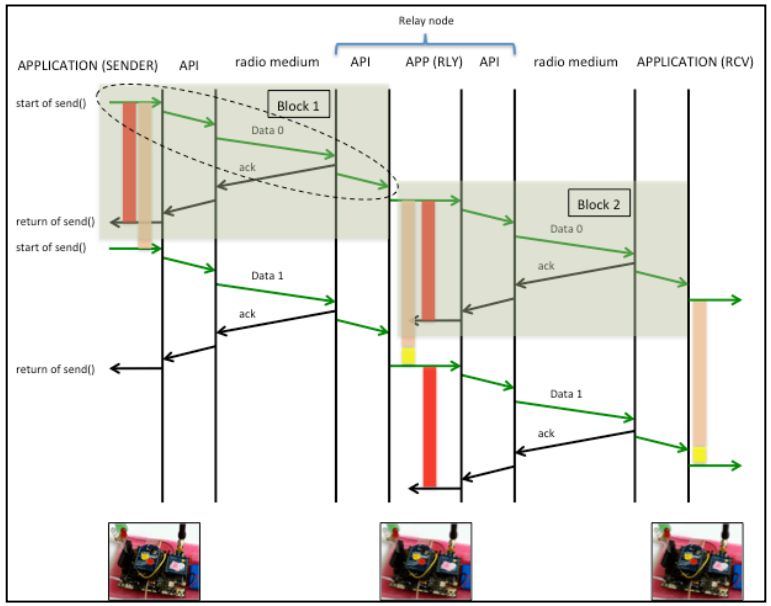
- What we have
  - Mesh configuration of IoT and gateways
  - IoT nodes rely on IEEE 802.15.4 radio
  - Radio modules are Xbee from Digi
- What we want to know
  - Upper bounds on sending and receiving throughput
  - Performances in a networked environment
  - Impact of API on performances
  - Where are the limitations?
  - To what extend audio traffic can be supported?

Qualification phase 1

- Phase 1
  - Determine upper bounds on performances of a single IoT node
  - Determine upper bounds on performances of multi-hop transmissions

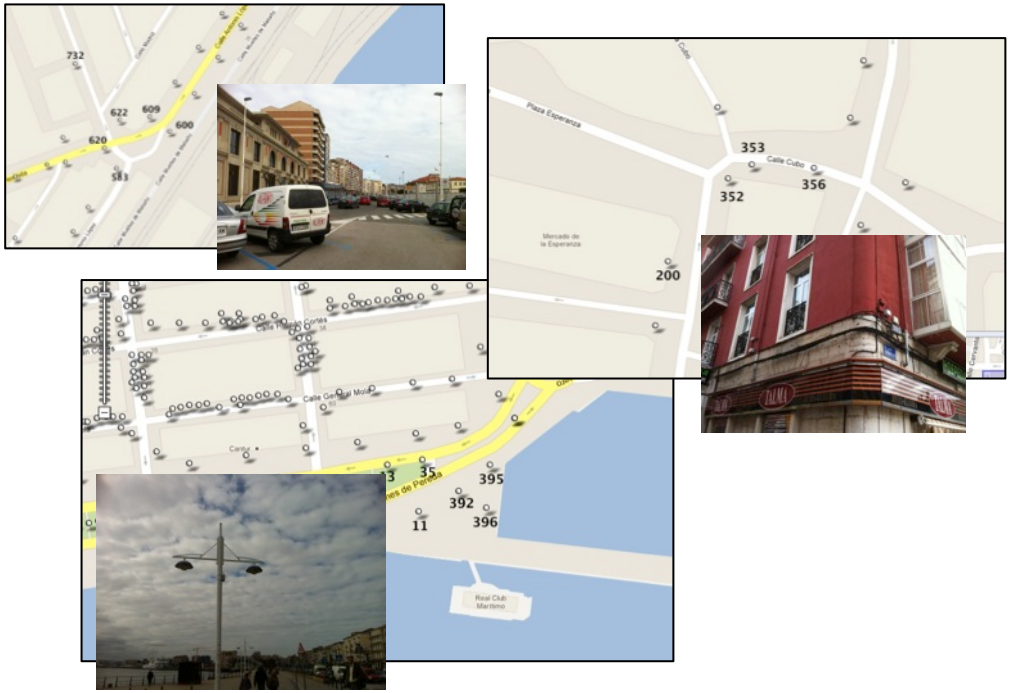


Traffic Generators
Sniffers
Advanced timing



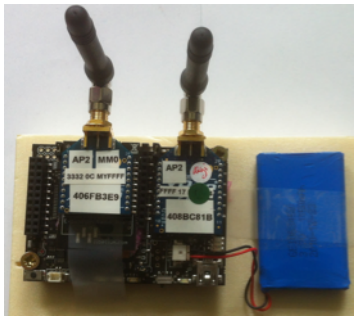
Qualification phase 2

- Phase 2
  - Performances in a networked environment: node density, traffic loads

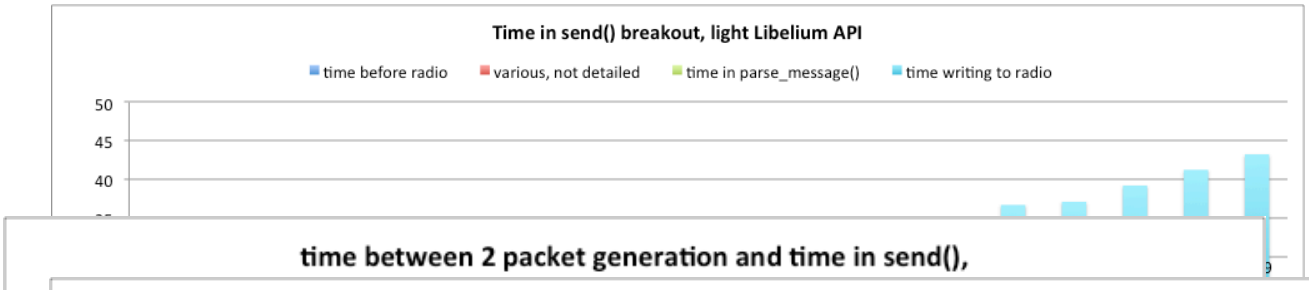


- Use representive locations in Santander for on-site test campaigns
- Deploy on IoT nodes traffic generators & sniffers
- Use mobile traffic generators & sniffers for dynamic traffic patterns
- Throughput, packet losses, latency,…

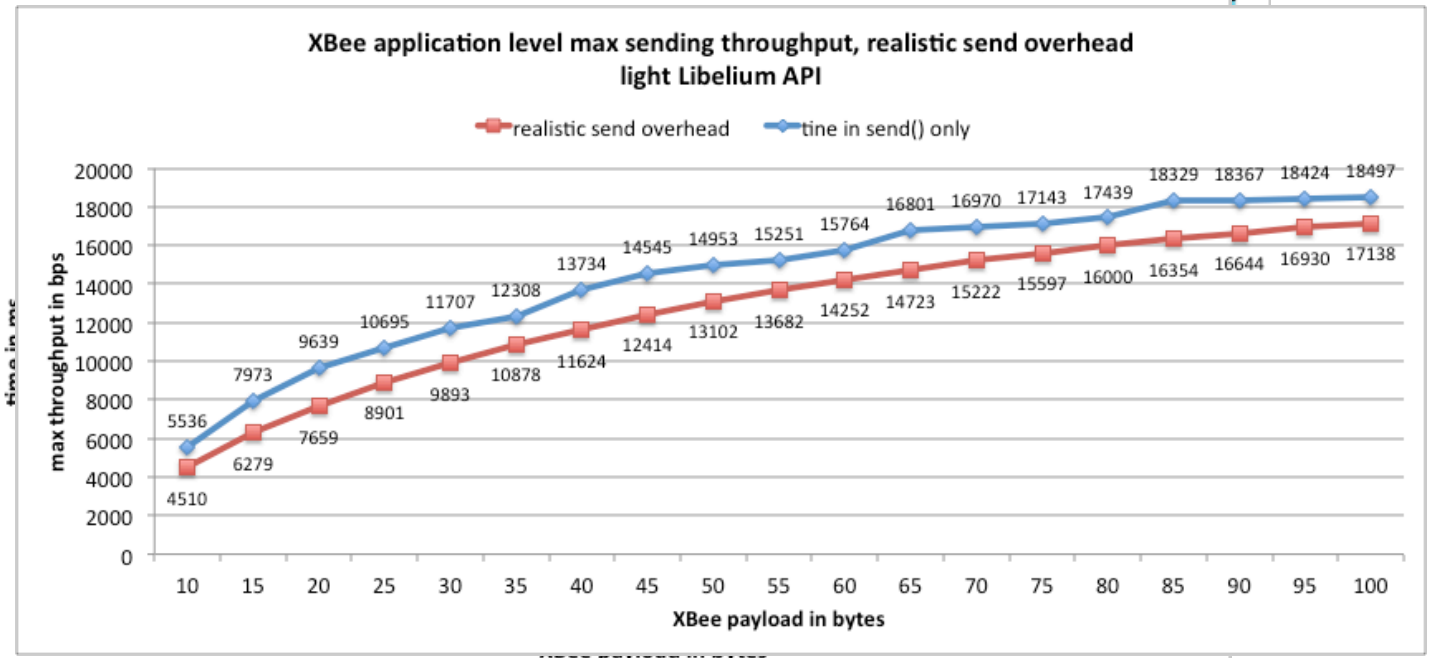
Node qualification: WaspMote (1)



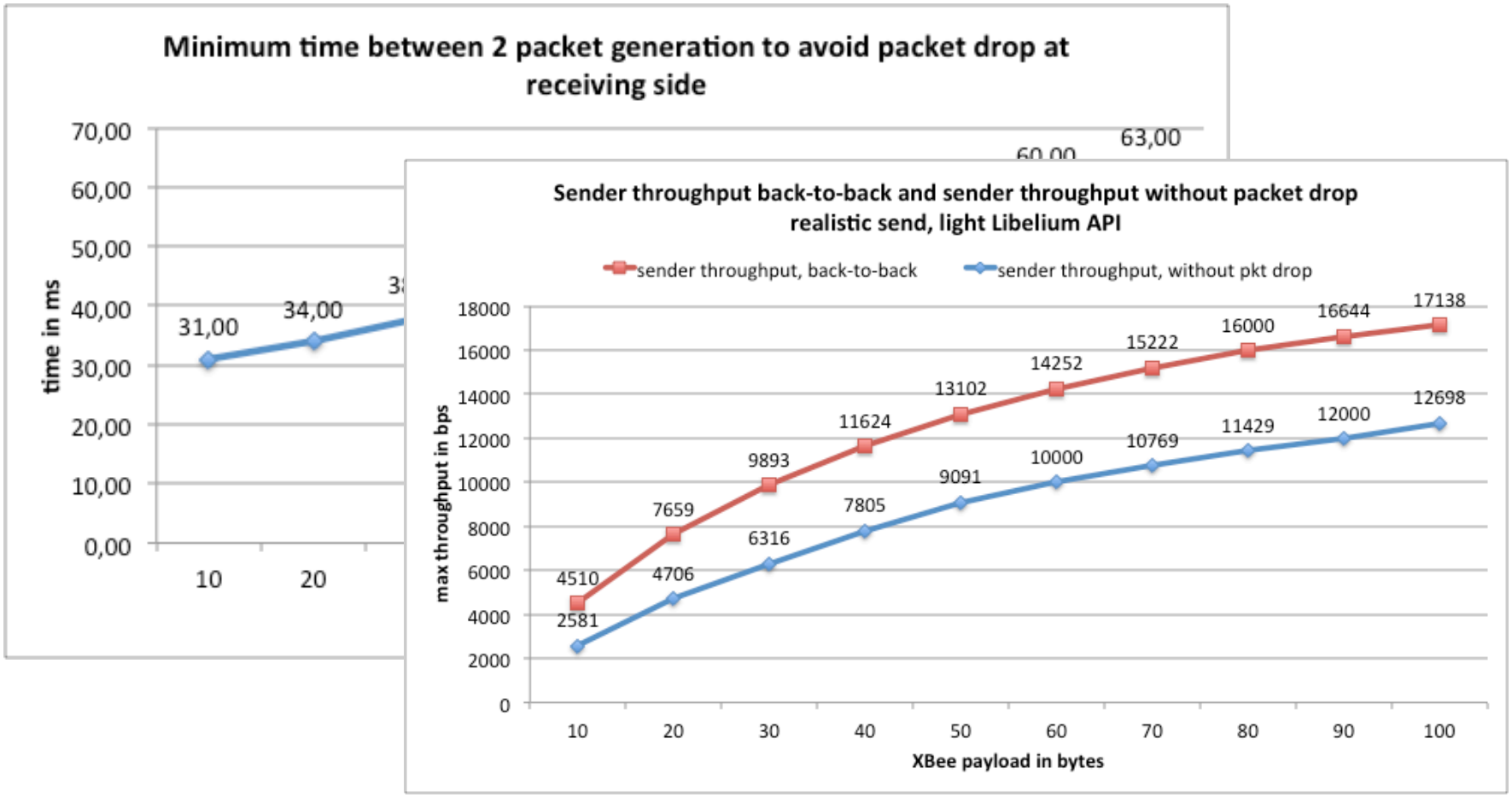
LIBELIUM WASPMOTE



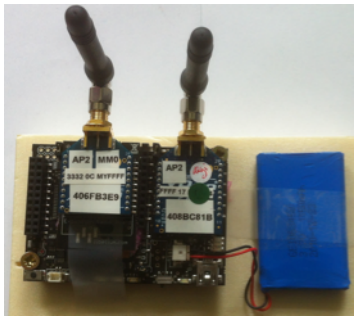
time between 2 packet generation and time in send(),



Node qualification: WaspMote (2)

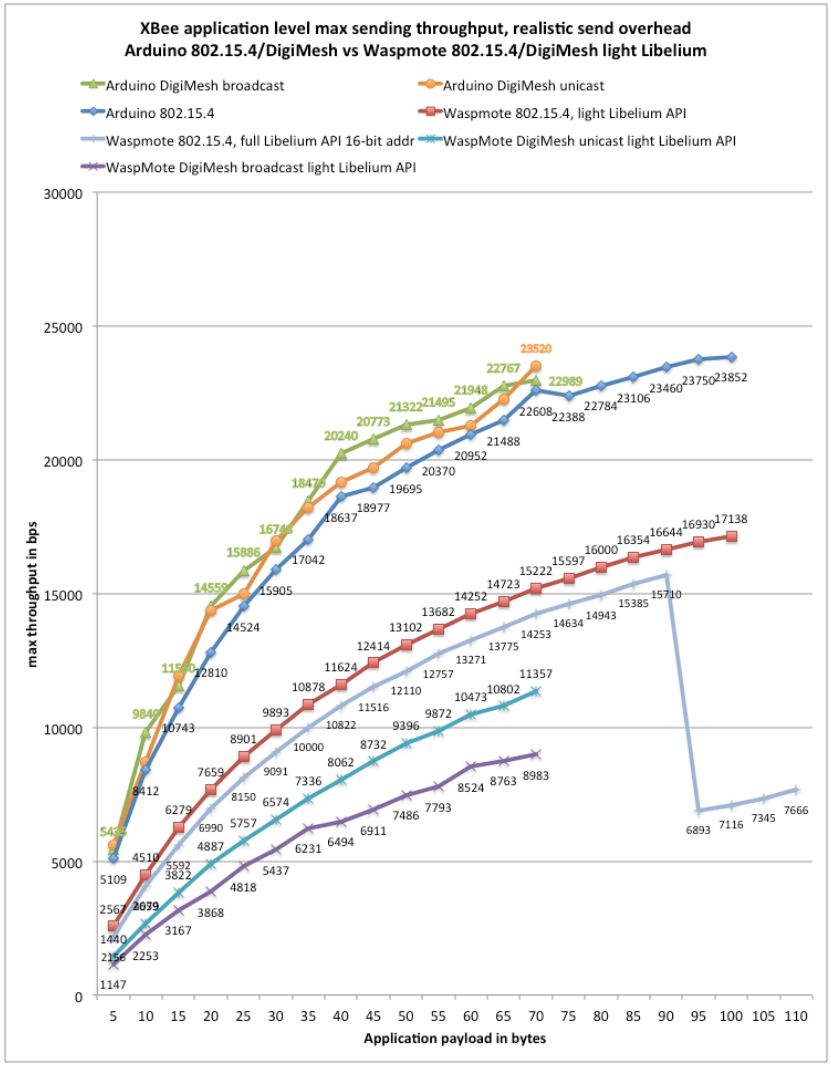


Technology comparison



LIBELIUM WASPMOTE

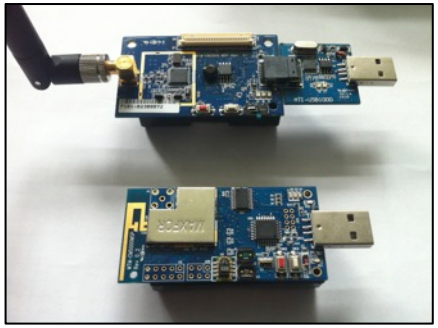
ARDUINO MEGA2560



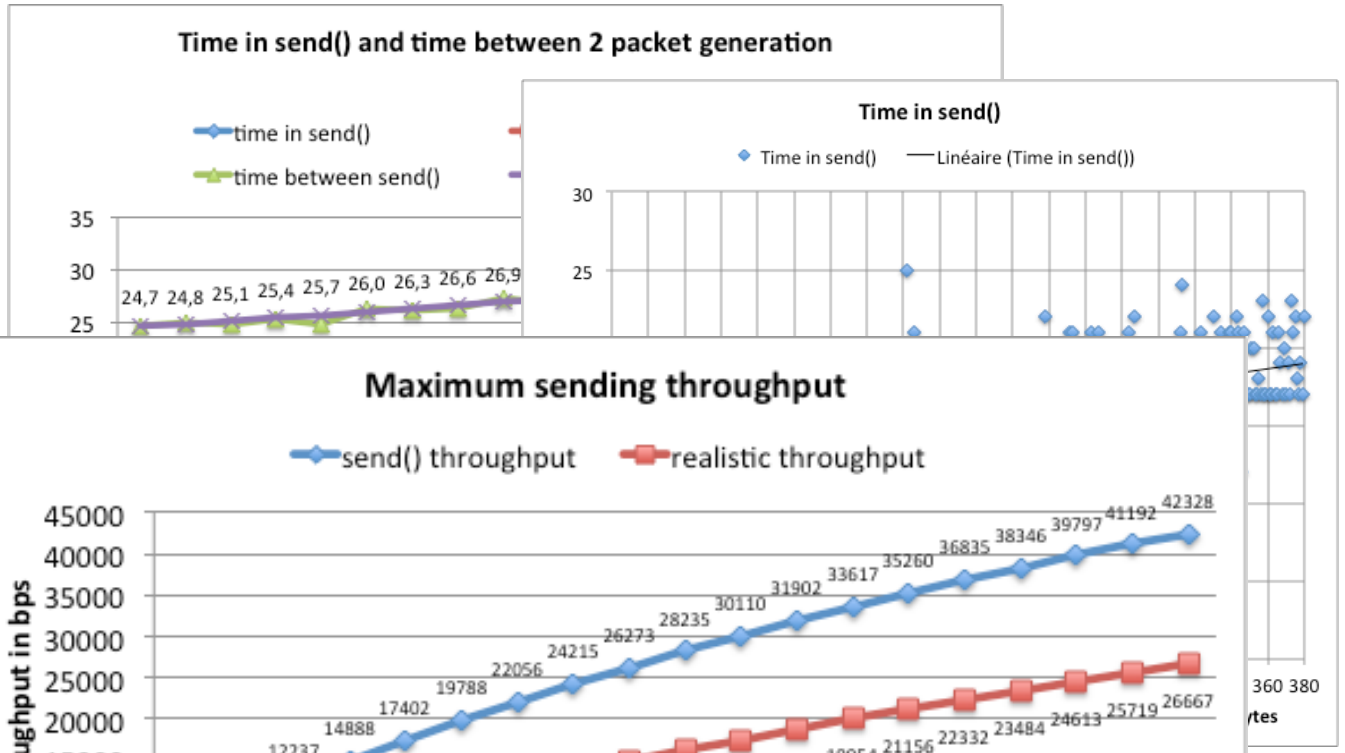
XBEE 802.15.4

XBEE DIGIMESH

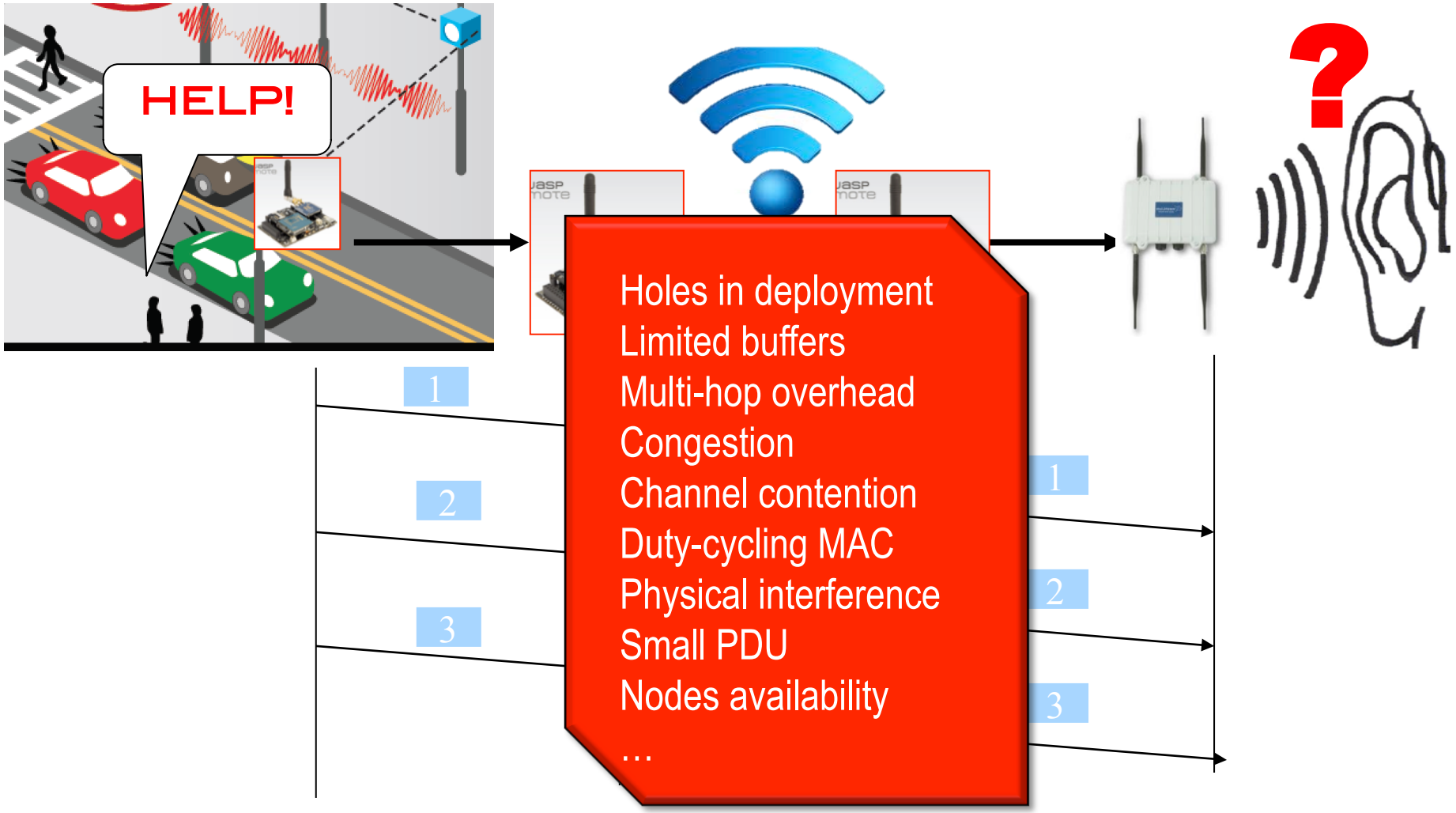
Node qualification: AdvanticSys



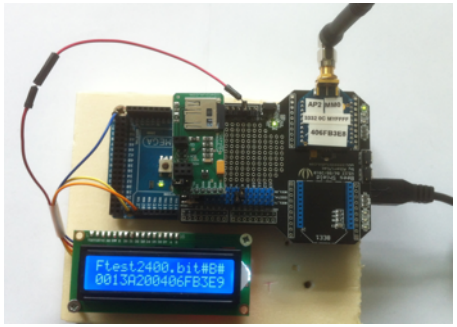
ADVANTICSYS
CM5000, CM3000



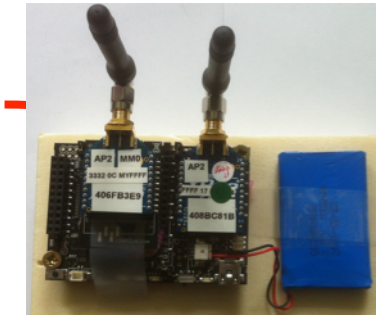
Multi-Hop Packet Forwarding?



Multi-hop audio test-bed

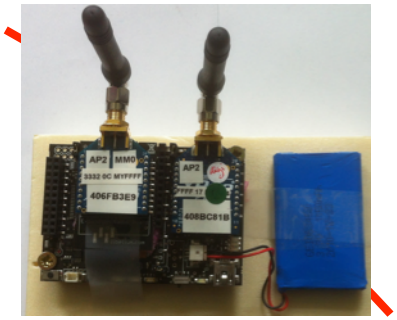


SENDS AN AUDIO FILE



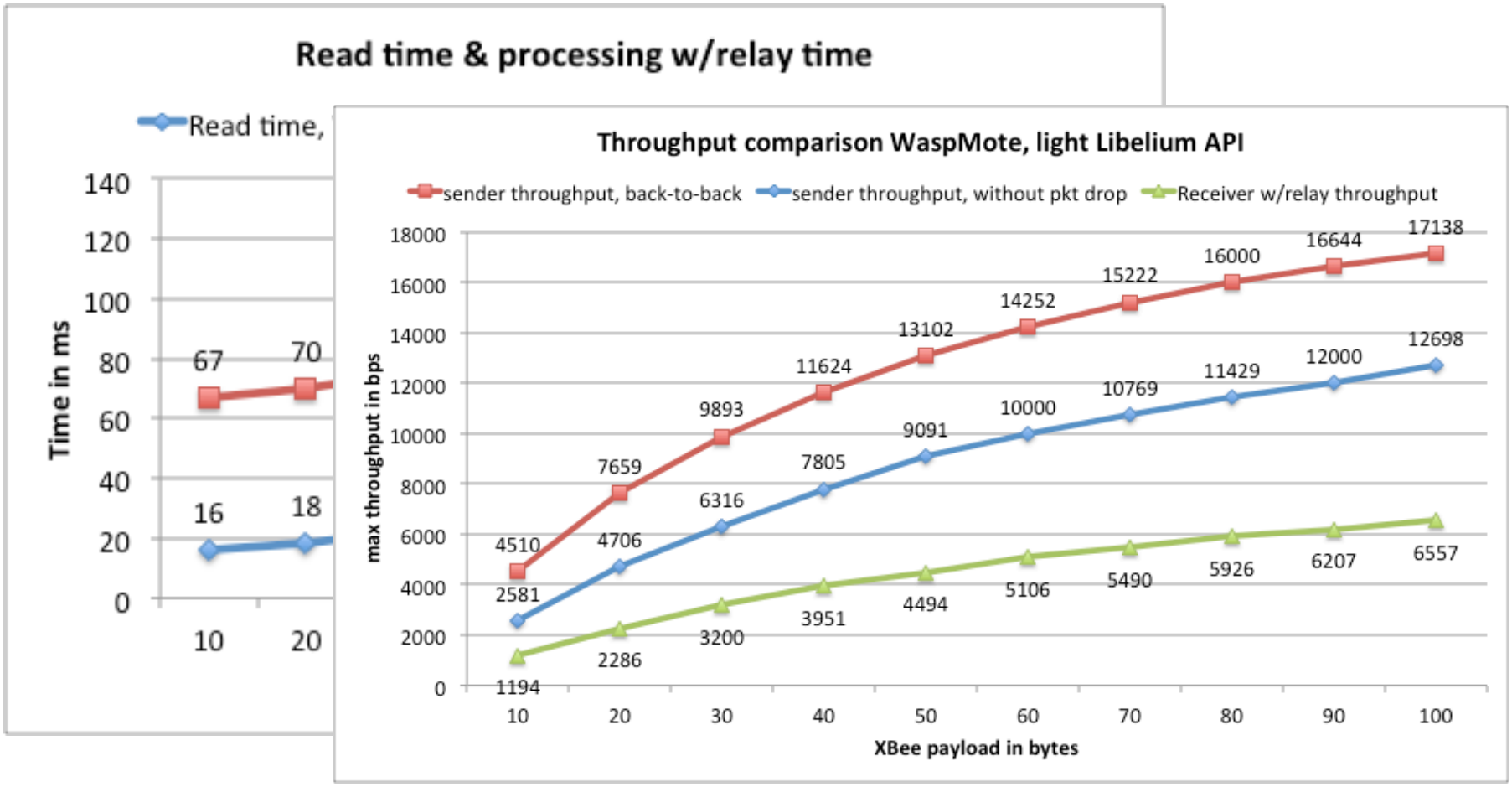
RELAY

RELAY



PLAY RECEIVED FILE

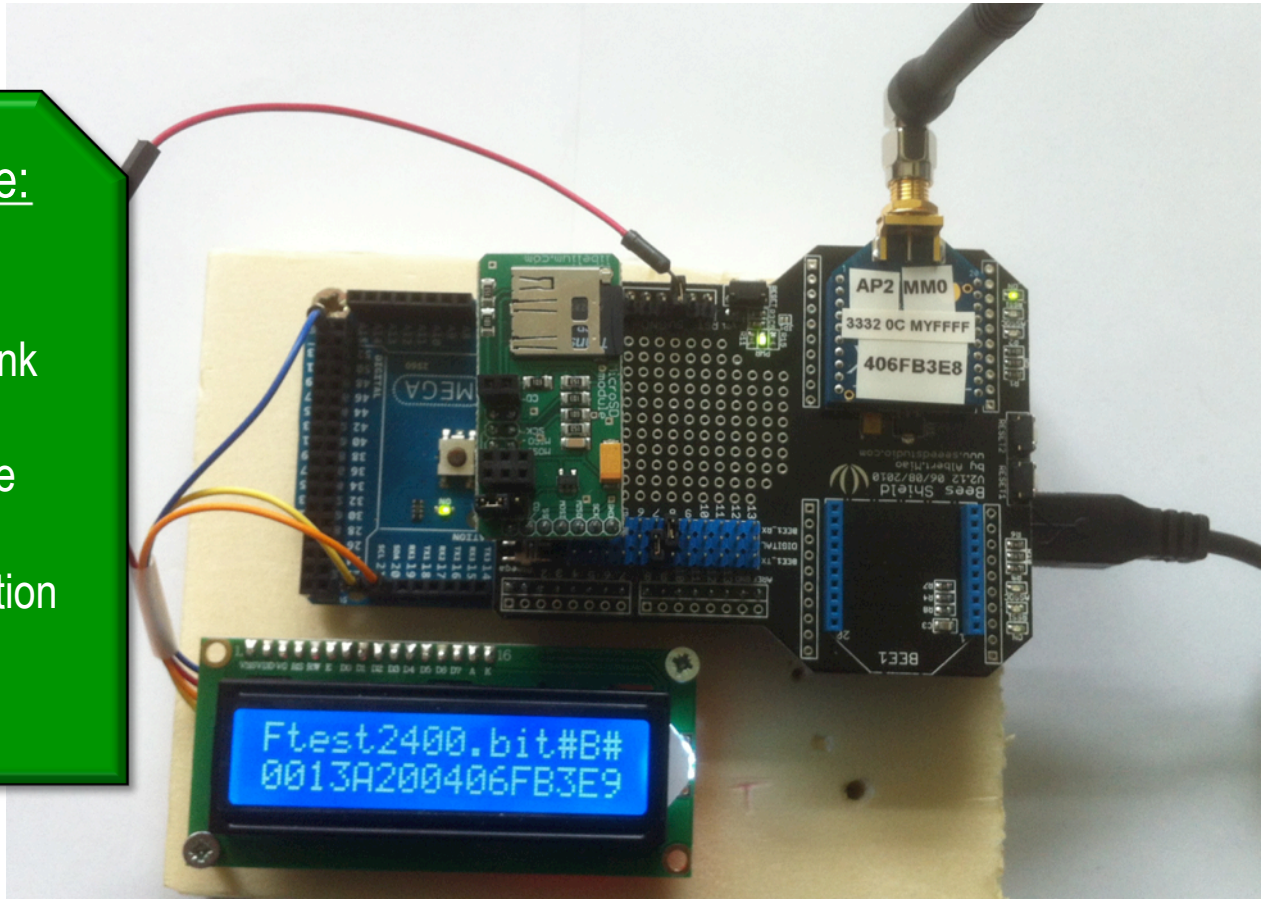
Multi-hop overheads



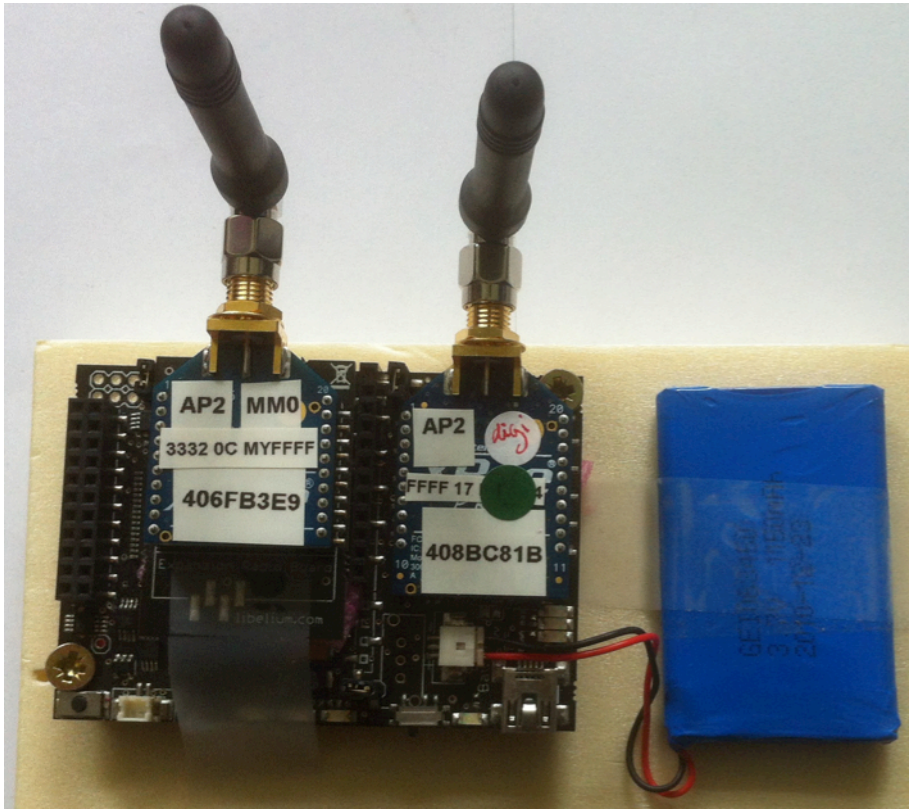
Sender node

Fully configurable:

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



Relay nodes



LIBELIUM WASPMOTE

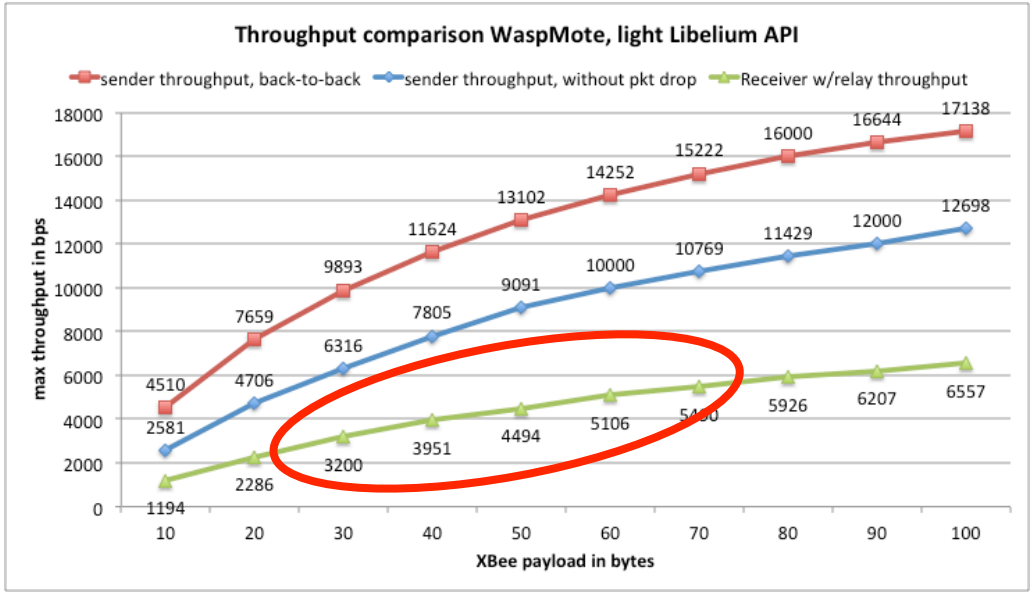
Fully configurable:

Destination node
Additional relay delay
Clock synchronization

Sink node

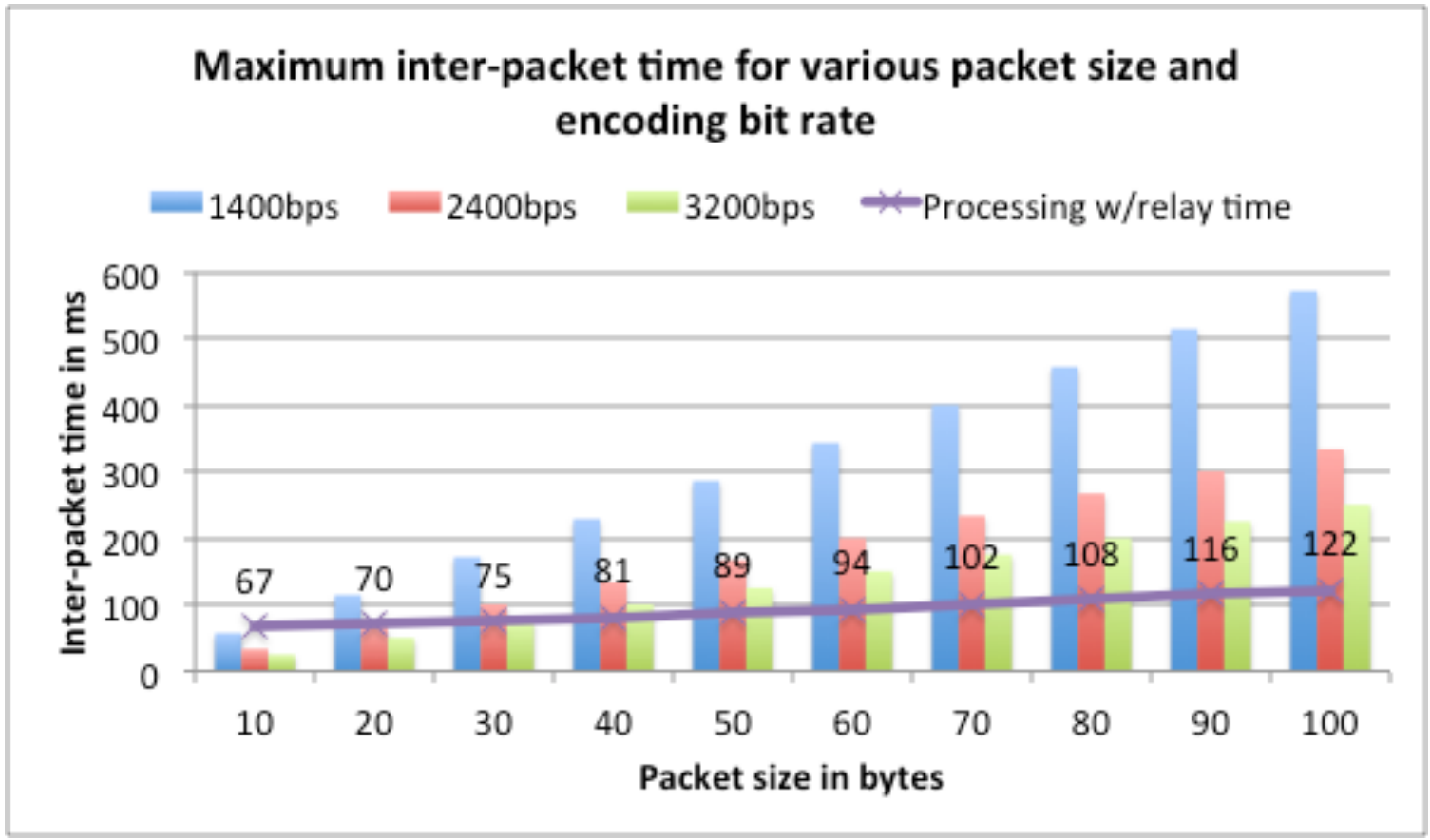
LINUX PC/LAPTOP WITH USB/SERIAL GATEWAY

Audio encoding



- Need a really low rate audio encoding scheme
- PCM is 64kbps, GSM 6.1 is 13kbps, can be lowered to 6kbps
- We use an open-source codec
  - codec2: http://codec2.org
  - Can be as low as 1400bps (1600, 2400 and 3400bps available)
  - All encoding/decoding tools are available in code source
  - Encoded file is robust against packet losses

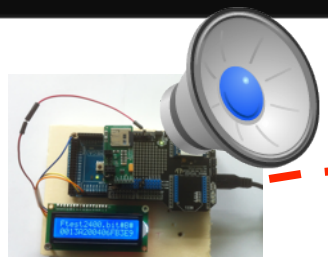
Can we meet the constraints?



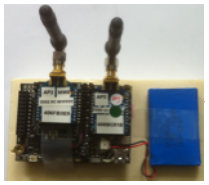
Test on SmartSantander

Test campaign – April 9th-10th 2013

Software for audio streaming

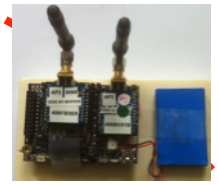


Sends an audio file



Relay

Relay



Sample Audio: 13s
PCM = 104000B
Codec2 at 2400 is 3900B

Play received file

Need to reduce the packet size to limit impact of packet losses

Store & Play

```
> XBeeReceive –B test2400.bit
> c2dec 2400 –B test2400.bit - | play -t raw -r 8000 -s -2 -
```

Streaming

```
> XBeeReceive –B -stdout test2400.bit | bfr -b1k -m2% - |c2dec
2400 - - | play -t raw -r 8000 -s -2 -
```

1-hop results

1-relay scenario									
bit rate	1400bps			2400bps			3200bps		
pkt size	40	50	60	40	50	60	40	50	60
n_{pkt}	59	47	39	101	81	67	134	108	90
t_{pkt}	105	110	120	105	110	120	105	110	120
n_{lost}	8	6	7	6	5	5	8	9	8
t_{pkt}	110	120	125	110	120	125	110	120	125
n_{lost}	1	0	0	0	2	2	3	1	3
t_s, s	6.5	5.6	4.8	11.1	9.7	8.3	14.7	14.4	11.2
t_{rcv}	6.9	6.4	5.2	11.6	10.1	8.8	15.4	15	11.7
t_{play}	4.7	4.5	3.7	8.4	8.2	6.1	13.1	12.8	9.8

Conclusions

- Receiver throughput is low and a maximum of 8kbps can be achieved without packet losses
- Low bit rate codecs for voices can be streamed from source to gateway provided that contention on radio links is low
- Multi-source is challenging


EAR-IT
the sounds of smart environment
Questions ?