Perceptual based Voice Multi-Hop Transmission over Wireless Sensor Networks

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***Abstract*—Multimedia applications over wireless sensor networks (WSNs) are rapidly gaining interest by the research community in order to develop new and mission critical services such as environmental video monitoring and emergency speech calls. In this work we analyze the possibility of sending voice using a network of wireless tiny motes with the final goal of enhancing speech quality by protecting the most perceptually important packets. We first evaluate the speech quality for a modified version of the ITU-T G.711 standard implemented to fit the particular selected hardware. Hence, we propose a low-complexity measure to evaluate the perceptual importance of speech packets. When performing single-hop experimental data collection, we apply packet redundancy (protection) by using a cooperative mote (relay) which retransmits speech packets that are perceptually important to protect them against potential transmission losses. Collected experimental results are then used to assess multi-hop performance, showing that the combination of the selected hardware and the proposed perceptual marking algorithm achieves good speech quality levels, according to the MOS scale, while reducing the percentage of protected packets by 40% when compared to random protection.**

I. Introduction

Wireless sensor networks (WSNs) have been experiencing a rapid growth in the last several years due to the development of new ultra low-power microcontrollers and short-range transceivers capable of reaching better performance with respect to older devices. WSNs technology is nowadays extensively adopted in a wide range of applications, where it replaces old wired and wireless systems, which are more expensive and harder to setup due to the necessity of both power and connection cables. A reduced set of WSNs applications include environmental monitoring [1], human tracking [2], biomedical research [3], military surveillance [4] and, more recently, multimedia transmission [5].

Multimedia content diffusion over WSNs is a very promising and challenging research area, which has only recently received attention by the research community. The possibility of sending video and voice through tiny mote networks, using cameras and microphones developed for the last generation of mobile phones, fosters the development of new useful and mission critical services, among which environmental video surveillance, advanced health care monitoring and opportunistic safety applications are most notable.

In this work we focus our attention on voice transmission over sensor networks for opportunistic safety applications suitable in case of accidents and natural disasters. For example, if we consider the scenario faced by first response rescuers after an earthquake, not only are their efforts hampered by the collapse of buildings and roads, but communications are also dramatically affected by the disaster. The rescuers' efforts can be therefore severely hampered because of their inability to coordinate between each other. In this scenario, low cost and low maintenance motes can be easily installed, moreover, surviving wireless sensor nodes can be used to enable communications between rescuers and survivors. The depicted scenario is only one example in which opportunistic voice communications over WSNs can be successfully adopted. A similar scenario is proposed in [6], where the authors consider the problem of enabling a wireless sensor network for voice communications in a coal mine. In their work Mangharam et al. installed in a coal mine a wireless network based on the FireFly [7] sensor node platform in order to enable voice data transmission. The FireFly platform incorporates an 8-bit microcontroller and a transceiver with a maximum raw data rate of 250 kb/s. The speech signal is acquired using a sampling rate of 4 kHz, instead of the usual 8 kHz required to preserve speech intelligibility, and then compressed according to the ITU-T G.726 standard [8], which is capable of bit rates from 16 kb/s to 40 kb/s. The use of a 4 kHz sampling rate and G.726 based voice compression reduces the bit rate necessary for the speech application, but on the other hand affects the received speech quality, which results in speech quality between poor (annoying distortion) and fair (slightly annoying distortion) — according to the mean opinion score (MOS) [9] scale — already for low values of packet loss rate. Even if the nature of the proposed application is opportunistic, every attempt must be made to improve the speech quality level in order to avoid misunderstanding between rescuers and survivors.

In this paper we consider a different hardware and speech compression standard compared to [6] and introduce a perceptual driven voice packet protection mechanism based on cooperative sensor nodes. The protection mechanism — which can be adapted to work with any compression standard — enhances speech quality while containing energy consumption at the cooperative sensor nodes. The selected hardware is the UTMOST platform [10], developed at the University of Texas at Dallas in 2007, which incorporates a 16-bit microcontroller and a radio transceiver capable of a raw data rate of up to 500 kb/s. The selected standard for the speech signal compression is the ITU-T G.711 [11], with a transmission bit rate of 64 kb/s. The standard is modified to fit the 12-bit analog to digital converter (ADC) provided by the microcontroller. Perceptually important speech packets are detected and labeled at run-time by the sender node. Only labeled packets are protected against transmission losses by being retransmitted by a cooperative sensor node (the relay), which is in close proximity to both the sender and the receiver of the radio link. Speech quality gains achievable with the proposed protection mechanism are assessed in a multi-hop scenario using indoor experimental measurements.

The rest of the paper is organized as follows: in Section II we describe the UTMOST platform, in Section III the performance of the modified ITU-T G.711 standard is shown along with the perceptually important packet selection algorithm. The cooperative (re)transmission mechanism is described in Section IV, performance results are presented and discussed in Section V, conclusions follow in Section VI.

II. THE UTMOST PLATFORM

The UTMOST wireless sensor node, depicted in Fig. 1, is a low cost platform designed for both research and deployment. It incorporates several key improvements with respect to popular motes such as Telos [12] and Mica's [13] family devices, leveraging technologies in order to provide more functionality at a lower cost in many areas.

The core of the system is based on the Texas Instruments MSP430F1611 [14], an 8 MHz, 16-bit microcontroller with an ultra low-power consumption profile. The microcontroller functionalities are preserved adopting a dynamic voltage booster, which regulates voltage device and guarantees all basic functionalities when the battery's charge is lower than a critical threshold. Since WSNs are expected to remain in sleep mode for most of their life, their consumption at this mode is crucial to the longevity of the mote. The UTMOST platform shows a reduction of the system current in the sleep state from 5.1 μA to 3.2 μA, while only 330 μA are required in active mode for both sensing and transmission. The incorporated CC1101 [15] transceiver operates in the ISM/SRD band (779 MHz - 928 MHz), and is capable of a raw data rate of up to 500 kb/s, double the rate with respect to transceivers with comparable values of power consumption, which usually operate at the 2.4 GHz band. An advantage of operating in this lower band instead of the 2.4 GHz is to avoid the interference of 802.11 based networks, which are being deployed at a rapid pace and pose a threat to mote communications [16]. The UTMOST platform also incorporates a 512 Mb parallel NAND flash storage device, which is a lower power and faster alternative to the serial peripheral interface based NOR flash devices adopted by the majority of current sensor devices. Although it requires more I/O pins, the NAND flash technology results in significant power savings as well as increased speed and capacity. Both improved storage capacity

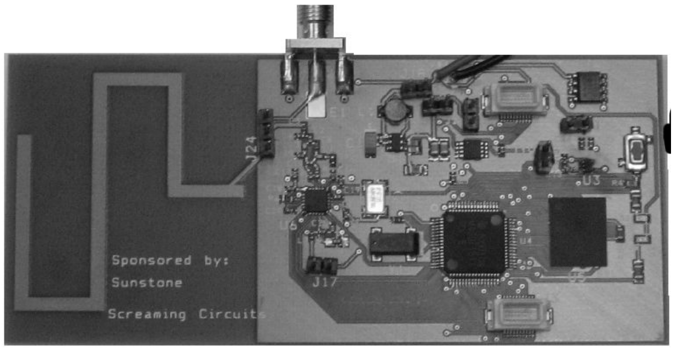

Fig. 1: The UTMOST platform.

and communications rates allow better support for demanding multimedia applications. Moreover, the ADC sampling rate guarantees the ability to preserve speech intelligibility, even though its 12-bit resolution of requires modified speech coding algorithms.

III. SPEECH CODING AND PERCEPTUAL SELECTION

As already mentioned, to send voice over the UTMOST platform, the ITU-T G.711 is selected from among the possible compression standards. The G.711 standard does not require high computational capabilities and can be easily implemented in a fixed-point architecture. Moreover, the two compression laws recommended, better known as A-law and μ-law, guarantee higher values of MOS with respect to other standards [17].

A. G.711 implementation for a 12-bit ADC architecture

Official ITU-T documents describe meticulously how the G.711 recommendation performs speech compression in the two laws previously introduced. In general, both of them perform a logarithmic compression of the speech signal starting by uniformly quantized PCM samples. The main difference between A-law and μ-law is the number of bits by which a uniform PCM sample has to be represented before performing the logarithmic compression. In the A-law case the compression algorithm maps a 13-bit value to a 8-bit value. In the μ-law case the mapping is from a 14-bit value to an 8-bit value. In both cases a transmission bit rate of 64 kb/s is required with a sampling rate of 8 kHz.

Due to the 12-bit conversion of the MSP430F1611's ADC, we implemented a modified version of the G.711 standard capable of mapping a 12-bit uniform quantized sample to

Gender	**A-law**		**μ-law**	
	Standard	12-bit	Standard	12-bit
Male	4.264	4.245	4.210	4.171
Female	4.356	4.337	4.308	4.259
All	4.311	4.292	4.260	4.216

TABLE I: MOS quality comparison both for A-law and μ-law between the original ITU-T G.711 standard and the proposed 12-bit implementation.

the corresponding 8-bit compressed value. The required bit reduction is obtained for both compression laws by changing the logarithmic quantizer. Table I shows a performance comparison between the ITU-T G.711 standard implementation and our 12-bit implementation. As we expected, the reduction in the number of bits used to represent a uniform quantized speech sample affects the speech quality. However, even in the worst case represented by the μ-law compression, the quality reduction in the MOS scale is less than 0.05. The speech quality measures are obtained using a subset of the NTT Multi-lingual Speech Database, by means of the perceptual evaluation of speech quality (PESQ) method [18] and then converted to the MOS scale (Excellent=5, Good=4, Fair=3, Poor=2, Bad=1) according to [19].

B. Perceptual selection

In order to mitigate speech quality degradation due to packet losses that occur over the wireless channel, protection mechanisms may be used in the network core or edge, e.g., forward error correction with interleaving techniques [20] and cooperative transmission among devices [21]. In general packet protection mechanisms lower packet loss rate by requiring additional bandwidth and power consumption, thus a trade off between the percentage of protected packets and the resulting speech quality is advisable.

The selection of the speech packets to be protected is performed by the sender. A packet is selected for protection as a function of its perceptual importance and a desired protection percentage. The perceptual importance of a speech packet can be expressed as the distortion that would be introduced by its loss. Bigger values of distortion reflect higher perceptual importance values. A common measure used to evaluate the distortion introduced by a speech packet loss is the spectral distortion (SD), defined as the power spectrum distance between the original speech packet and its reconstructed version obtained by means of concealment algorithms. The SD equation is shown in (1), where S_X and $\widehat{S}_X$ are the power spectra of the original and concealed speech packet X, respectively.

$$SD = \sqrt{\frac{1}{N}\sum_{i=1}^{N}[10log_{10}(S_X(i)) - 10log_{10}(\widehat{S}_X(i))]^2} \quad (1)$$

The SD computation requires an analysis-by-synthesis approach at the sender, where the packet loss is simulated and concealment algorithms applied for evaluating the reconstructed packet. A discrete Fourier transform (DFT) is then performed on both the original and the reconstructed packets. The described approach is not suitable for tiny mote devices with limited computational capabilities, thus a low-complexity distortion measure is needed, at the cost of some loss in performance. A first simplification of equation (1) is reached by choosing a simple packet loss concealment based on silent insertion of lost packets. Even if this choice does not guarantee higher performance than state of the art algorithms, it can be easily implemented with minimal computational requirements. With the selected concealment the log power spectrum $10log_{10}(\widehat{S}_X)$ can be considered equal to zero, thus the SD for a single packet depends only on the power spectrum of the original signal.

Equation (1) with silent insertion concealment shares similarities with Parseval's equality (the power spectrum of a signal is equal to the square of the magnitude of its samples in the time domain), thus a low-complexity perceptual measure can be performed using only the ADC uniform quantized samples which compose a speech packet. The adopted measure is reported in equation (2), and it's based on the sum of the absolute values of the acquired samples, $X(i)$.

$$P(X) = \sum_{i=1}^{N} |X(i)| \quad (2)$$

Fig. 2 shows the proposed measure as a function of the spectral distortion for a subset of the NTT Multi-lingual Speech Database and using a packet length of 20 ms. There is an encouraging direct dependency between the two measures, suggesting that the measure in (2) may well capture the actual spectral distortion trend.

As previously introduced, the selection of the most important speech packets is performed according to two main parameters: the perceptual importance defined in (2) and the desired percentage of protected packets. Once the perceptual importance of a packet X is evaluated, it is classified as perceptually important if $P(X) > T(p)$, where $T(p)$ is a threshold which is a function of the protection percentage p. A number of thresholds has been evaluated from the cumulative distribution function (cdf) of $P(X)$. For example, if we impose a protection percentage equal to 30% this means that the 70% of the packets will not be protected, thus packets with $P(X)$ less than the threshold with a cumulative probability equal to 0.7 will not be selected. Even if the threshold based marking algorithm guarantees protection only for the most perceptually important speech packets, it does not guarantee that a specific

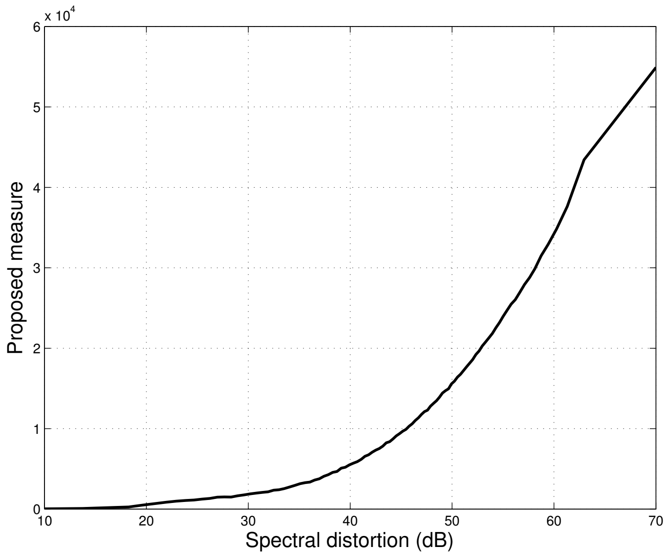

Fig. 2: Proposed measure versus spectral distortion.

speech flow achieves the desired protection percentage. This occurs because the threshold values have been selected to work with a certain group of speakers, so they are optimal in general. In order to overcome this problem and reduce fluctuations, we changed dynamically the compared threshold after every window of M packets. More in particular, for every M packets a run-time protection percentage is updated and the threshold is increased or decreased in order to reach the desired p target.

IV. Cooperative transmission

The speech packets, which are labeled as perceptually important by the sender, are protected against potential transmission losses with the help of cooperative sensor nodes (relays), which are in the vicinity of both the radio link sender and receiver [21], [22]. Relays are not to be confused with the intermediate nodes of a multi-hop route, as clarified next.

In a wireless sensor network, all (sensor) nodes share the same transmission medium and channel access is regulated by means of some access protocol. When a packet is sent from the end-sender to the end-receiver it travels through the network hop (radio link) by hop (radio link) across an ordered subset of selected intermediate nodes, which are chosen by the adopted routing protocol. Each intermediate node performs packet store-and-forward till the packet reaches the end-receiver. The broadcast nature of the wireless medium results in other nodes (besides the selected intermediate nodes) possibly receiving the packet as well, which gives rise to an opportunity for neighboring nodes to act as relays and assist (cooperate) in the packet delivery over individual radio links. Relays can help in mitigating the high packet loss rate (PLR) at radio link receivers caused by unpredictable fluctuations of the wireless channel quality as follows.

A basic cooperation system, depicted in Fig. 3, consists of three entities, the radio link source, the destination and the relay node. When a packet is sent from the source to the destination, the relay node — usually placed between the source and destination — can receive and make a copy of the transmitted packet. A moment later, the relay can forward an additional copy of the packet to the destination, effectively lowering the PLR of the received data flow over the radio link. The relay node provides a spatially distinct path for transmission in which the distribution of losses is different from the one in the source to receiver link. It is reasonable to expect that a number of relay nodes are available in WSNs because of the natural high density required for wireless sensor networks to operate successfully. The nodes chosen to act as relays may be selected while computing the end-to-end multi-hop route [23].

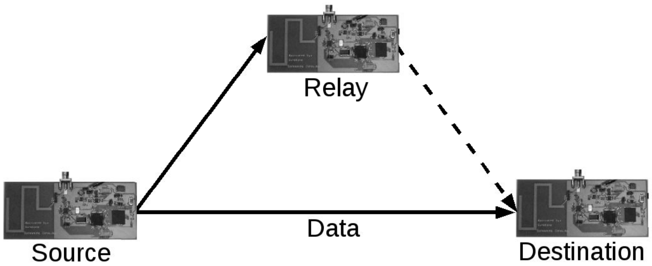

Fig. 3: A three nodes based cooperation system.

In the implemented cooperative transmission mechanism, a trade off between the number of relay transmissions and the final speech quality is reached by protecting only the most perceptually important packets, which are selected and labeled at the source according to the procedure described in Section III-B. The protection label information is inserted in the packet header, thus the relay node needs to receive and process only a minimum amount of data before deciding whether to discard or retransmit the incoming speech packet.

V. Performance results

Voice transmission experimental results were collected in terms of speech quality using the UTMOST hardware and the modified ITU-T G.711 speech compression standard. The performance of the perceptual driven cooperative transmission mechanism is compared against the performance of a cooperative transmission in which protection is given to speech packets at random.

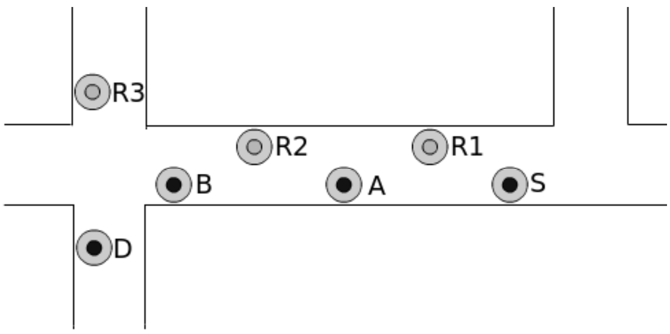

Fig. 4: Experimental data collection scenario.

A. Wireless sensor network setup

Speech transmission traces were collected in an indoor experiment conducted on the University of Texas at Dallas campus. The 3 hop scenario used in the experiment is qualitatively depicted in Fig. 4. Seven UTMOST nodes were placed at the two sides and at the corners of a corridor. The distance between two nodes on the same side of the corridor is approximately 20 m, while between the two corridor sides the nodes are shifted by approximately 10 m. The two nodes installed at the corner sides, R3 and D, are about 15 m apart. The adopted configuration on the one hand guarantees a good environmental monitoring, on the other hand guarantees that in case of transmission along one corridor side the relays nodes along to the other side are placed between each sender and receiver.

The CC1101 transceiver of each UTMOST node is set to the lowest output power to reduce the energy consumption from batteries, i.e., -30 dBm. The selected transmission frequency is equal to 915 MHz, using a PCB antenna which at the selected transmission frequency shows a quasi omnidirectional

	Perceptual protection					Random protection				
ΔMOS	MOS	Protected packets (%)	Target protec. (%)	Link PLR (%)	Receiver PLR (%)	MOS	Protected packets (%)	Target protec. (%)	Link PLR (%)	Receiver PLR (%)
0	2.246	0.00	0.00	11.14	11.14	2.246	0.00	0.00	11.14	11.14
0.193	2.530	9.36	10.00	11.14	9.98	2.337	10.00	10.00	11.14	9.48
0.409	2.859	19.46	20.00	11.14	8.79	2.450	20.00	20.00	11.14	8.93
0.528	3.096	29.25	30.00	11.14	7.92	2.568	30.00	30.00	11.14	7.81
0.789	3.501	39.42	40.00	11.14	6.62	2.712	40.00	40.00	11.14	6.70
0.981	3.855	49.56	50.00	11.14	5.56	2.874	50.00	50.00	11.14	5.59
1.068	4.125	59.49	60.00	11.14	4.35	3.057	60.00	60.00	11.14	4.50
0.993	4.262	69.43	70.00	11.14	3.47	3.269	70.00	70.00	11.14	3.41
0.781	4.330	79.29	80.00	11.14	2.31	3.549	80.00	80.00	11.14	2.28
0.478	4.343	89.27	90.00	11.14	1.24	3.865	90.00	90.00	11.14	1.16
0	4.353	100.00	100.00	11.14	0.06	4.353	100.00	100.00	11.14	0.06

TABLE II: MOS performance results for A-law compression as a function of the protected percentage of speech packets for both perceptual and random protections.

radiation pattern in all planes with a maximum gain between 2.9 dB and 4.6 dB [24].

In order to better characterize the adopted scenario, preliminary experiments were conducted sending speech packets every 20 ms, for a speech flow bit rate equal to 64 kb/s, in order to evaluate the packet loss rate on each direct link between sensor nodes. The measured PLR for a one hop transmission between two nodes along the same corridor side varies from a minimum of 3.3% to a maximum of 6.0%. The measured PLR for a transmission between nodes on opposite corridor sides is in the 1.6% to 3.0% range. Even if every link offers relatively good (low) values of PLR, its behavior is highly time variant with long bursts of lost packets.

B. Speech quality performance

The speech quality performance is evaluated both for the perceptual and random selection of speech packets as a function of the desired protection percentage. The wireless sensor network is configured to send speech packets from the end-sender (S) to the end-destination (D) through intermediate nodes A and B. All the other nodes act as relays for each radio link sender and receiver pair. The total packet size is equal to 170 bytes, i.e., 160 bytes of speech samples plus 10 bytes of header. Packet are sent every 20 ms, reaching a link bit rate of 68 kb/s. Packet loss traces were collected for each single radio link (hop) in isolation and accounting for the coordinated actions taken by the sender, receiver and relay[1]. End-to-end speech quality measures are evaluated offline averaging MOS values obtained by applying the experimental loss traces to selected speech traces of the NTT Multi-lingual Speech Database. The speech quality measures have been performed by means of the PESQ method and then converted to the MOS scale according to [19].

Results of the performed speech quality analysis are shown respectively in Table II and Fig. 5 for a numerical and graphical comparison. Only results for the A-law are presented since μ-law results shows the same behavior. The total packet loss rate from the end-sender to the end-receiver for the three hops communication with no cooperative transmission among nodes is up to 11%, which results in Poor speech quality level. The speech quality of the received stream increases according to higher percentage values of the protected packets, which results in lower values of PLR at the end-receiver node for both perceptual and random selections, approaching the overall MOS value of 4.382 (reachable with no transmission

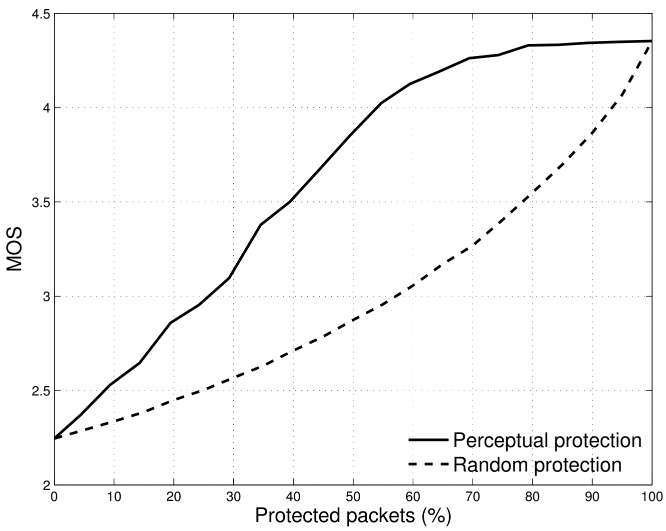

Fig. 5: MOS comparison for both perceptual and random protections as a function of the protected percentage of speech packets.

[1]The choice of collecting loss traces in isolation, one sender, receiver, and relay at the time, may partially reduce the interference among electromagnetic waves, thus potentially positively affecting the link PLR. However, both protection techniques (perceptual driven and random selection) are similarly affected by this fact and their performance comparison is still meaningful.

losses). Moreover, the perceptual selection of protected speech packets yields better speech quality values when compared to random selection for every imposed protection percentage. The performance gain ranges from a minimum of 0.193 to a maximum of 1.068 reaching the Good speech quality level (MOS value of 4) already with 55% protection level. The same Good speech quality level is reached by the random selection only when protecting 95% of the speech packets, as illustrated in Fig. 5.

The presented results show how it is possible to enable voice transmission over wireless sensor networks with good speech quality. Even when the variable nature of the transmission medium adversely affects the speech quality, voice packet protection mechanisms can guarantee higher quality performance. Moreover, in the analyzed scenario (where a trade off between speech quality and sensor node available resources is necessary) a careful selection of the speech packets to be protected, which is driven by their perceptual importance, yields high speech quality gains with a reduced percentage of protected packets. For examples, with reference to the above discussed results, the perceptual driven selection requires 40% less packets to be protected compared to the random selection in order to yield Good speech quality level, thus considerably reducing both the relay's bandwidth requirements and energy consumption.

VI. Conclusions

In this paper we analyzed the problem of enabling high quality speech transmission over wireless sensor networks with both hardware improvements and algorithm development. In order to better support multimedia applications over WSNs we selected the UTMOST platform, developed at the University of Texas at Dallas, which is capable of lower power consumption values and higher raw transmission bit rates when compared to commercial devices. Speech compression was obtained by means of an optimized version of ITU-T G.711 speech compression standard. A reduction in packet losses was achieved by using a developed low-complexity perceptual based algorithm combined with a cooperative transmission technique among neighboring sensor nodes (relays), which retransmit only the perceptually most important packets.

Using experimental results a three hop end-to-end transmission was studied, showing that the adopted hardware and the developed algorithms jointly yield high speech quality values. Moreover, the proposed perceptual driven selection of speech packets for protection was shown to yield higher MOS values compared to packet random selection for the same percentage of protected speech packets. It was also noted that the perceptual driven selection reaches Good speech quality level by protecting 40% fewer packets when compared to the random selection, thus limiting both bandwidth requirement and energy consumption at the relays.

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