Perceptual audio coding for wireless acoustic sensor networks

Thesis outline

There has recently been a paradigm shift in the way acoustic signal processing, specifically speech enhancement and noise reduction, is being performed. This paradigm shift is the direct result of the availability of a plethora of electronic devices: mobile phones, laptops, tablets, etc., each with a (set of) microphone(s) as depicted in the figure below. Microphone signals can be exchanged between devices, forming a so-called wireless acoustic sensor network (WASN), which allows for better speech enhancement and noise reduction performance in the individual devices. These WASNs have also seen growing interest for use with hearing prostheses, such as hearing aids and cochlear implants, as the electronic devices in the WASN allow for higher quality microphone signals to be captured.



However, as the number of devices increases, there is a diminishing amount of available bandwidth that can be used. This diminished bandwidth, along with other device constraints such as broadcast power, makes the use of compression or coding crucial for continual operation of the WASN. This coding can be accomplished via lossless or lossy compression and is often dependent on the complexity of the algorithm compared to the available resources of the device as well as the capacity of the communication channel allotted to the device. Since the devices that compose the envisaged WASN all have different broadcast abilities, they can utilize different coding strategies to broadcast signals through the network.

The employed coding strategies should not only take into consideration the previously mentioned constraints, but also specific audio characteristics, such as the spectral differences between speech and noise signals, while at the same time limiting the amount of distortion introduced to the processed signal.

Your Task:

This thesis project looks to analyze different aspects of the communication between devices in a WASN and their impact on the noise reduction and speech enhancement performance. The starting point will be rate constrained hearing aids that utilize some type of audio coding between them. This will be further extended to incorporating higher capacity devices that do not need to rely on such aggressive coding strategies. Finally, multiple devices will be deployed in the environment which, depending on the device constraints such as energy and bandwidth, each utilize a different coding strategy.

Desired skill set

Students interested in this project should have knowledge of the central tenets of signal processing (FFT analysis, filtering, etc.) and any knowledge regarding audio processing is a plus. The majority of the work will be accomplished in the MATLAB environment, therefore, students should be well aquatinted with it.

Promoter: Marc Moonen
Supervisor: Joseph Szurley (B00.62)
Number of Students: 1-2
Work Load: Literature 40%, Implementation 30%, Validation 30%

Suggested reading: S. Srinivasanand, A.C. denBrinker,"Rate-constrained beamforming in binaural hearing aids," *EURASIP Journal on Advances in Signal Processing*, 2009.
O. Roy and M. Vetterli, "Rate-constrained collaborative noise reduction for wireless hearing aids," *IEEE Trans. Signal Processing*, vol. 57, no. 2, pp. 645–657, Feb. 2009.