Optimized design and analysis of sparse-sampling functional magnetic resonance imaging experiments of speech and hearing

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Functional magnetic resonance imaging (fMRI) offers an unparalleled opportunity to investigate the brain bases of speech and hearing. However, the high-amplitude (>90dB) acoustic noise that occurs during MR image acquisition presents a serious obstacle to research on speech perception and production. “Sparse sampling” is an alternate acquisition strategy that mitigates the interference of this acoustic noise by inserting a delay between subsequent image acquisitions, allowing auditory stimulus presentation or speech production during this silent period. Although this technique is routinely employed in auditory fMRI, there has been no empirical attempt to optimize the design of sparse sampling paradigms to maximize detection of whole-brain blood-oxygen level dependent (BOLD) signal. Moreover, the discontinuous nature of the sparse-sampling timeseries has led to the use of analysis models that fail to account for dynamic properties of the hemodynamic response, thus seriously underestimating BOLD signal and limiting the types of cognitive brain activity sparse sampling is able to detect. We present computational modeling and human neuroimaging experiments that explore the parameter space of sparse sampling experiment design and analysis – including delay duration, stimulation frequency, and hemodynamic response convolution – and offer significantly enhanced detection of brain activity during speech perception tasks versus conventional methods.