Adaptive pacing of language production task performance using time-aware methods

Thomas J. Grabowski1,2, Matthew Bauer1, Derek Foreman1, William W. Graves1, Dori DeFoie1, Sonya Mehta1, Lizann Bolinger3

1Department of Neurology, University of Iowa
2Department of Radiology, University of Iowa
3National Research Council Canada, Institute for Biodiagnostics, Winnipeg MB

Objective: The physiologic correlates of language processing in impaired subjects remain largely unexplored because of the fundamental difficulty of interpreting data that are contaminated by abnormal or failed performance. The problem of mixed performance can be addressed with event-related design, which can be made compatible with monitoring speech production (Birn et al., 1999; Barch et al., 1999), and which allows selective analysis of successful or unsuccessful trials. It may also be desirable to adjust the rate of task performance (Dalesaar et al, 2003) for impaired subjects, to avoid collision of trials and to minimize differences from the control group in emotion and arousal. At the same time, efficiency and avoidance of paradigm-correlated speech/movement artifacts require a stochastic inter-trial interval (ITI). We used a “time aware” fMRI acquisition and processing environment (I/OWA, Smyser et al, 2001) to implement an interactive approach to single word language production studies in fMRI, capable of adapting in real time to individual subject performance.

Methods: I/OWA continuously records the receiver unblank and other information channels pertinent to monitoring subject behavior or modeling MRI signal, and uses a streaming data model to support real-time data processing and paradigm control. Sound from the scanner bore (speech and gradient noise) is collected at 16 kHz. Clear speech is generated by streaming subtraction of epochs of the raw sound stream acquired before the first utterance from subsequent epochs, aligning the samples with reference to a 16kHz-sampled receiver unblank data stream. The investigator pushes a button when a response has been successfully spoken. The visual stimulator delivers the next stimulus not sooner than a specified stochastic delay calculated from the previous stimulation time, contingent on receiving “permission” to do so by the investigator button push. Thus, timing is determined algorithmically and content is endorsed by the investigator. If no response is generated, the system “times out” after a specified interval and delivers the next stimulus. Because the delivery of the next stimulus is contingent not only on a response, but on its content, as well as on a constraint on ITI, we refer to the arrangement as “adaptive pacing” rather than “self pacing”.

Results & Discussion: This system is in routine use. Results of real-time speech filtering are shown in the Figure. Processing overhead per trial is approximately 0.25s (much shorter than ITI). Preliminary studies in normal subjects using a single word production paradigm show preservation of efficiency and avoidance of temporal correlation of task reference functions with speech envelopes. The advantages of the approach compared to a simple “voice key” are robustness to erroneous utterances, and preservation of a stochastic ITI. The benefits of adapting pacing are expected to be the accrual of more successful trials, better efficiency (informative data per unit time), and less contamination of data by correlates of other processes, notably emotion.

Conclusions: The system described is suitable for application to patients with language production disorders (anomia, stuttering, aphasia), including longitudinal studies during which there may be improvement or progression of impairments.

References & Acknowledgements: Supported by R33 EB001484