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Computers/Electronics

Adaptive Suppression of Noise in Voice Communications

A digital signal processor effects SNR-dependent spectral subtraction.

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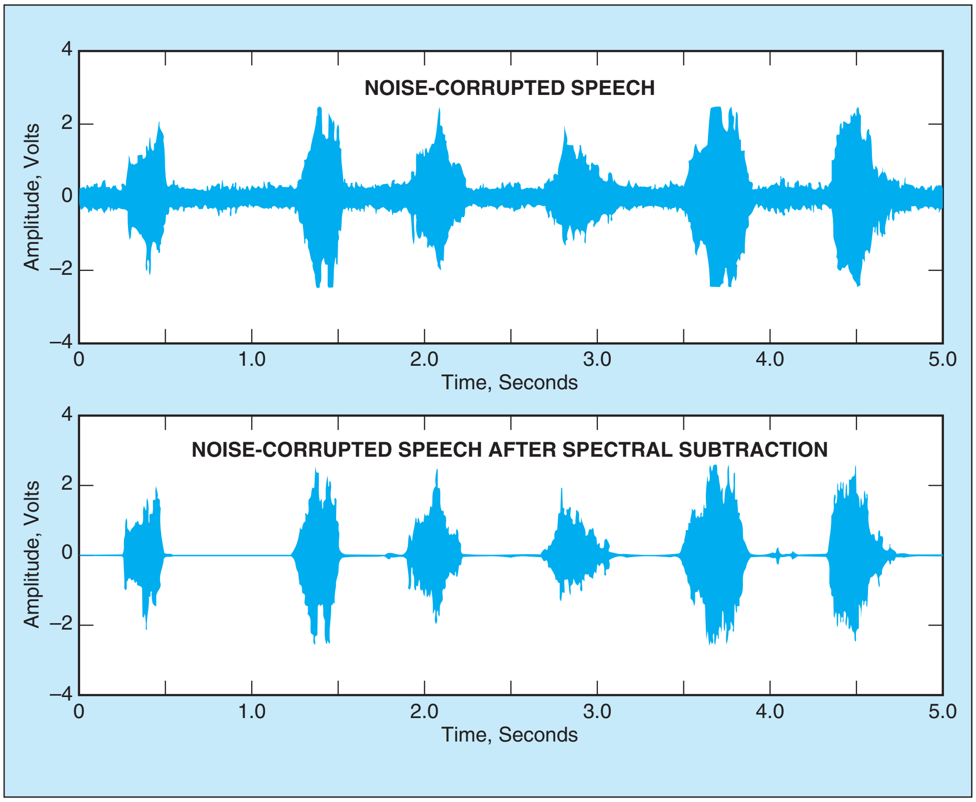
A subsystem for the adaptive suppression of noise in a voice-communication system effects a high level of reduction of noise that enters the system through microphones. The subsystem includes a digital signal processor (DSP) plus circuitry that implements voice-recognition and spectral-manipulation techniques.

The development of the adaptive noise-suppression subsystem was prompted by the following considerations: During processing of the space shuttle at Kennedy Space Center, voice communications among test team members have been significantly impaired in several instances because some test participants have had to communicate from locations with high ambient noise levels. Ear protection for the personnel involved is commercially available and is used in such situations. However, commercially available noise-canceling microphones do not provide sufficient reduction of noise that enters through microphones and thus becomes transmitted on outbound communication links.



The **Noise in a Noisy Speech Signal** was reduced nearly to zero by the spectral-subtraction process.

In operation, noise or noise-corrupted speech enters the adaptive noise-suppression subsystem through a microphone. The output of the microphone is sent through a high-gain amplifier and an antialiasing low-pass filter. The output of the filter is sampled by an analog-to-digital converter.

At this point, the DSP suppresses noise by executing a spectral-subtraction algorithm that incorporates a dependence on the signal-to-noise ratio (SNR). The noise level and the SNR for use in the algorithm are determined by a subalgorithm that includes the following steps: First, each frame of raw data put out by the analog-to-digital converter is examined to determine whether it is a voiced or an unvoiced frame. An estimate of the noise is obtained during each unvoiced frame. A running average of the noise is then computed and used to approximate the expected value of the noise.

Using the SNR computed as described above, the SNR-dependent algorithm pre-emphasizes the frequency components of the input signal that contain the consonant information in human speech. The algorithm then determines the SNR and adjusts the proportion of spectral subtraction accordingly. After spectral subtraction, de-emphasis filtering is performed and low-amplitude signals are squelched. The resulting digital signal is processed through a digital-to-analog converter, then through a smoothing/voice-band filter, which is a band-pass filter with low and high 3-dB roll-off frequencies of 300 Hz and 3 kHz, respectively. The resulting analog signal is used to modulate a transmitter in the communication system.

In a demonstration of the adaptive noise-suppression system, the words "test, one, two, three, four, five" were spoken into the microphone. The figure contains graphs of the original sampled noise-corrupted speech signal and the signal after spectral subtraction. Spectral subtraction increased the SNR by approximately 20 dB. A listening test verified that the original noise was virtually eliminated, and that little or no distortion in the form of musical noise was introduced.

This work was done by David Kozel of Purdue University, James A. DeVault of **Kennedy Space Center**, *and Richard B. Birr formerly of I-NET, Inc. Further information is contained in a TSP (see page 1).*

This invention is owned by NASA, and a patent application has been filed. Inquiries concerning nonexclusive or exclusive license for its commercial development should be addressed to the Technology Programs and Commercialization Office, Kennedy Space Center, (321) 867-6373. Refer to KSC-11937.