WHITE PAPER

TEXAS INSTRUMENTS

Dynamic Adjustment of System Parameters Improves Echo and Noise Cancellation

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Executive Summary

Speech quality is a critical factor in establishing consumer preference of mobile handsets, wireless networks, hands-free devices and other mobile communications systems. Acoustic echoes and noise are endemic in wireless communications. Signal processing is required to overcome these concerns and achieve a high-quality audio output acceptable to the market. This paper discusses sources of echo in wireless communications as well as a new approach to analyze and cancel unwanted noise and echoes.

The traditional approach to noise and echo cancellation is to utilize standalone echo and noise cancellation modules on the near-side or transmit path. This approach works well under constant conditions, but environment changes, such as a door opening or loud noises, degrades the audio performance as the audio system struggles to adapt.

A new approach integrates echo cancellation, noise suppression, and other sound enhancement technologies, making it possible to dynamically adjust systems parameters much more quickly to the environment change. In most cases, the adjustments can be made before the consumer can detect any degradation in sound quality. Likewise, the higher levels of integration provided by this new approach make it possible to compensate for larger echoes and noises, increasing the potential for achieving natural sounding full-duplex conversations.

These large improvements in echo and noise cancellation have come at a critical time as legislation in numerous U.S. states have instituted full or partial bans making it illegal for drivers to hold cell phones while driving. This ban already exists in the overwhelming majority of European countries and many others worldwide. These regulations are creating new demands for hands-free technology and overcoming echoes in the noisy environment of a car interior is the greatest challenge in design of hands-free systems. Designers need to have access to easy-to-use hardware and software that allows them to provide the same sound quality in hands-free audio products to which consumers have become accustomed in handsets.

Sources of Echo in Wireless Communications

Echoes in wireless systems arise from two basis sources: electrical and acoustic. Electrical echoes occur when there is poor design which allows the speaker signal to couple directly into the microphone signal. The best solution for this is to use good design practices.

The more challenging type of echo to remove is the acoustic echo. Acoustic echo occurs when the amplified speaker signal is echoed back through the microphone.

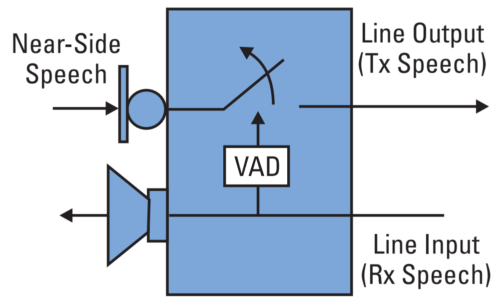

Figure 1: Half-duplex solution

Canceling (or removing) this echo can be challenging, as there are multiple factors which should be considered. Amplified speaker sounds can be reflected along multiple paths at different times. These indirect echoes can be noticeably delayed from the original signal since sound propagation in air is only about 300 meters per second. The echo reflection could also become distorted due to mechanical vibration which adds further complications.

Half-Duplex, Switching Technology

The most basic attempt to solve the echo problem is simply to disable the near-side speech path when far-side speech is detected. This eliminates acoustic echoes but results in only one person being able to talk at a time. For example, on a conventional walkie-talkie when you push the talk button you are no longer able to hear what other parties on the line may be saying. That is why two-way radio etiquette requires the speaker to say "over" when he or she has finished. Later technology replaced the push-to-talk button with a Voice Activity Detector (VAD) that automatically opens and closes the near-side speech path when far-side speech is detected. This limitation was accepted in the early days of mobile communications but will no longer be tolerated by users that are accustomed to full-duplex wireline conversations where they can interject thoughts, murmur their agreement or disagreement, and pause for emphasis without fear that that the microphone will be figuratively yanked out of their hands.

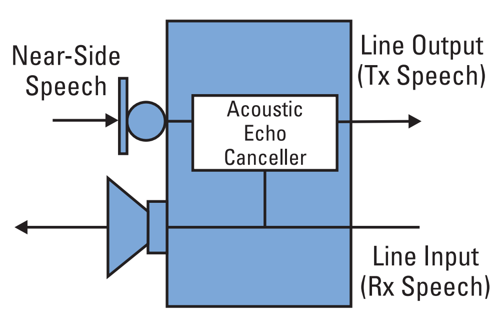

Figure 2: Conventional echo cancellation technology

Conventional Echo Cancellation Technology

This explains why nearly all cell phones, hands-free devices, and speaker phones provide some method of echo cancellation. The basic approach that is nearly universally used is to monitor the far-side signal and subtract it from the received signal. If the amount of echo were constant and known, this would be simple to accomplish, but the amplitude and timing of the echo depend upon the environment in which the wireless device is being used and this environment is subject to change. Conventional echo cancellation technology addresses this challenge by continuously monitoring the near-side and far-side signals. The acoustic echo canceller algorithm uses the reference signal from the near-side speaker to estimate the echo path and remove the echo from the near-side microphone signal.

The design and tuning of an adaptive filter is the essence of echo cancellation performance. The filter typically uses the known characteristics of an audio signal to calculate the echo estimate and then makes adjustments to the filter parameters in order to minimize the error. The normalized least mean squares (NLMS) algorithm is typically used to update the coefficients of the filter that is used to eliminate echoes. This algorithm is designed to minimize the mean square error of the canceller, where the error is the residual echo. The adaptation is generally normalized by the signal power in order to be independent of signal levels.

These calculations can be performed accurately enough in most cases to reduce perceptible echo. The problem is that the algorithm is dependent on a stable echo path between the speaker and microphone. The echo path changes whenever some acoustical obstruction is placed close to the phone (e.g., moving a phone from a hand to a desktop, touching the keypad, passing a paper over the speaker) or when the microphone-to-speaker distance is adjusted (e.g., a tethered microphone is repositioned). When these path changes occur, there is a certain delay before the algorithm can adjust to the new echo path. During this adaptation delay, acoustic echoes can be transmitted along the near-side signal path.

In designing an echo canceller it is important to understand the environment of the device in which the echo canceller will be working. Are the microphone and speaker in fixed locations? Will the unit be carried around? What is the longest echo path for the environment that the device will be working in? How much noise is expected? Will the noise vary (like in a car environment)? How loud will the unit need to be? How much echo return loss is there between the speaker and microphone? How loud will the near side talker be compared to the echo at the microphone? Understanding the answers to these questions helps to design the best traditional echo canceller which adapts to the known environment. However, when the environment changes, echo may occur while the filter coefficients are adapting to the new echo path. This adaptation can be 5 to 10 seconds or longer depending on the initial parameter settings.

In addition to the problem of echo affecting near side signal quality, background noise can also be an issue. The solution for this is to use a noise canceller. The typical noise canceller works independently from the echo canceller described above and any interactions are ignored. Unlike the echo canceller, the noise canceller does not have a reference signal to train on. It must either estimate the noise and remove it from the microphone signal, or estimate the voice. In either of these cases training only on the noise is extremely important to get the best possible performance. Using control signals from the noise canceller and AEC together creates an environment for more accurate voice activity detection and better convergence. Without this interaction, the system may inadvertently try to cancel voice instead of noise.

New Approach Provides Higher Level of Integration

A new approach to wireless audio sound quality shown in Figure 3 has been developed in an effort to address some of the limitations of the conventional approach. A basic difference between the new and the old approaches is that the new approach integrates echo cancellation, noise cancellation, and other audio signal processing functions under the control of a new full-duplex control module. This approach uses the same core NLMS algorithm with some special features, takes advantage of the broader system knowledge inherent in this integrated approach, and dynamically adjusts the system parameters for quick re-convergence of the NLMS.

The full-duplex control provides the key to the performance improvements offered by the new approach. Integrating the control elements for the audio section of the wireless

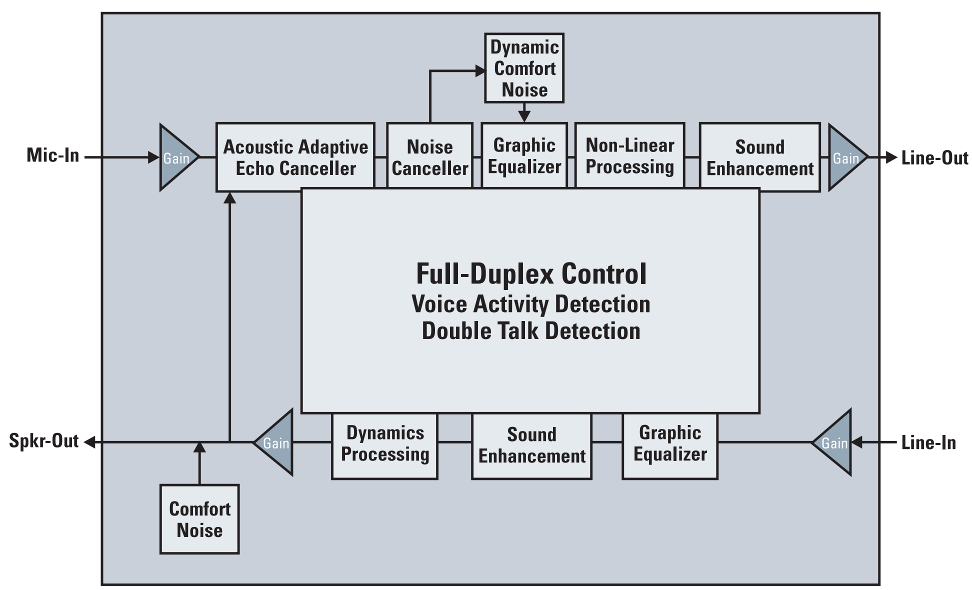

Figure 3: New approach integrates echo and noise cancellation with other audio processing technologies

communications device with the latest digital signal processing technology makes it possible to use nonlinear control algorithms that can adjust to sudden environmental changes, such as a door slamming in the background or the user making a sudden gesture with the hand holding the phone. Additional improvements in sound quality can be achieved by simultaneous optimization of the different control algorithms under a master controller. Finally, the use of more powerful signal processing architectures make it possible to add new capabilities such as filling in the background with natural sounding comfort noise to compensate for changes in the noise background that would otherwise appear as noise pumping.

Integrating the system processing for all of the key elements of the near-side and far-side audio paths to optimize signal quality on both ends of the conversation would be difficult with the previous generation of digital signal processors (DSPs). The recent availability of DSPs with the correct balance of performance and high levels of on-chip memory capacity enables the level of algorithm complexity and audio processing integration needed to rapidly optimize all of the different audio elements that contribute to the best wireless speech quality.

How the New Approach Works

The new approach uses the entire system to obtain knowledge of the current working environment and dynamically adjust the system parameters for best performance. The analysis and parameter adjustments are the task of the integrated full-duplex control. The full-duplex control evaluates the near-side and far-side signals, primarily to determine whether they are currently active and evaluates their quality from several different perspectives. Based on this information, the full-duplex control dynamically adjusts the full range of modules used to enhance the quality of both the near-side and far-side signals.

The full-duplex control on the near-side signal path, controls the parameters for the non-linear processor, echo canceller, and noise canceller to reduce echo and noise. The full-duplex control on the far-side signal path, controls the dynamics processing which modifies the audio signal to allow higher volume output with reduced speaker non-linearities. The graphic equalizer and sound enhancement are used in both signal paths. These graphic equalizers are used to adjust for transducer changes (both speaker and microphone) or to adjust the frequency characteristics of the audio signal. The sound enhancement is used to adjust the voice quality for the best possible speech clarification.

The difference in using this systems approach is that the full-duplex control can utilize the environment knowledge it gains from the system to allow a platform to have louder volume with less echo and quickly adapt to changing environments.

Designing the New Audio Processing System

The greatly increased level of integration in this new audio processing system presented a number of design challenges. The first was to identify a DSP that could deliver the higher levels of performance required by the new design and offer a programming environment that would make it possible to deliver a much more complex design than conventional echo and noise cancellers in time to meet short mobile communications systems design lead times.

Acoustic Technologies selected the TMS320VC5407 DSP from Texas Instruments because it provides the ideal combination of processing performance and high on-chip memory capacity which conserves processor cycles by reducing the need for cycle-intensive off-chip memory operations. The C5407 DSP delivers 120 MIPS of processing performance which provides headroom for addition of features and customization above the just under 80 MIPS required for echo cancellation, noise reduction, audio streaming, and Bluetooth algorithms.

The C5407 DSP provides 128K × 16 bit on-chip ROM configured for program memory and 40K × 16-bit on-chip RAM composed of five blocks of 8K × 16-bit dual-access program data RAM. The device automatically accesses the on-chip ROM when addressing

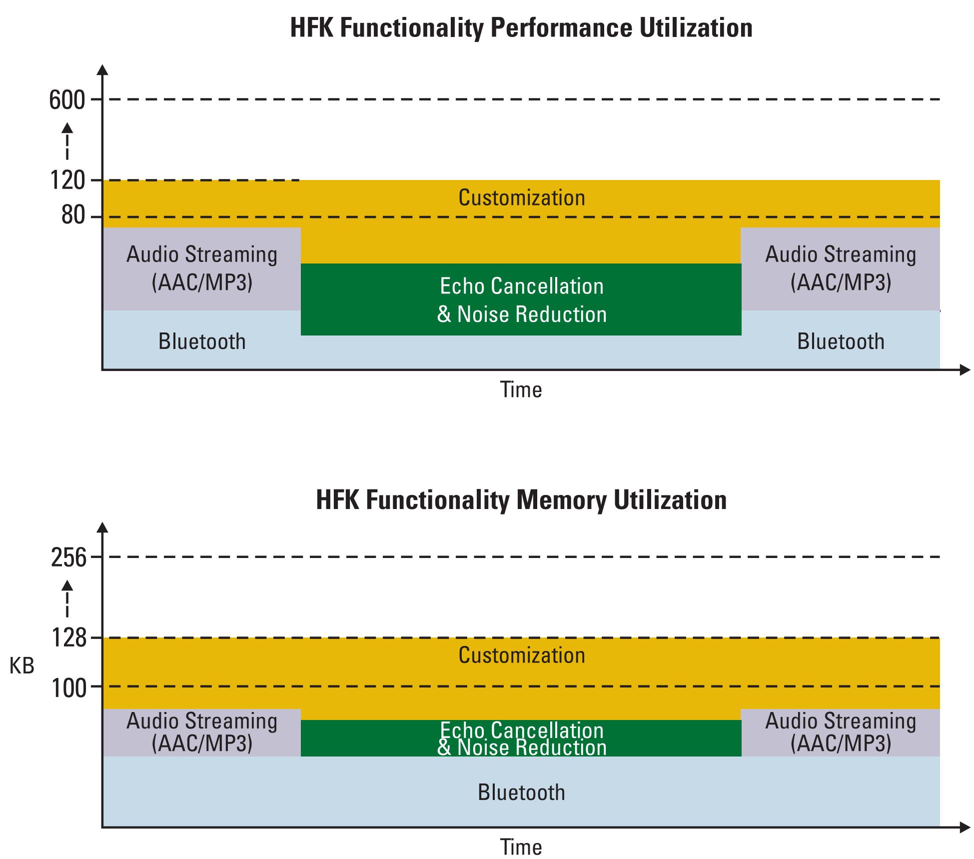

Figure 4: Performance and memory utilization

within its bounds. When an address is generated outside the ROM bounds, the device automatically generates an external address. Operating from on-chip memory provides higher performance because it creates better flow within the pipeline of the central arithmetic logic unit and wait states are not required. On-chip memory negates the need for external memory costs and internal memory consumes less power.

By selecting the C5407 DSP, Acoustic Technologies also obtained the scalability that is provided by the full TMS320C54x™ DSP and TMS320C55x™ DSP generations, providing up to 600 MIPS of processing performance. Designers can easily design for a price point within the generations as low as $2.99 per piece at 10,000 units. The scalability of the C54x™ and C55x™ DSP generations also makes it possible to easily add peripherals such as USB ports.

A wide range of available software from TI third parties dramatically helps reduce design cycle time. Members of the TI DSP Third Party Network offer diverse product portfolios that make it easy for original equipment manufacturers (OEMs) and original design manufacturers (ODMs) to add features such as audio streaming of MP3 and WMA files, Bluetooth functionality, speech recognition, phonebook download, etc.

The Hands Free Kit (HFK) Reference Design offered by Texas Instruments with Acoustic Technologies and Adamya Computing Technologies can save at least 40 weeks in development cycle time for hands-free cellular phone kits for automobiles. Acoustic Technologies provides echo and noise cancellation software and Adamya provides Bluetooth support. The basic design provides audio streaming from a cell phone, MP3 player, or car stereo interfaces and supports additional features such as identifying an incoming call, creating a ring tone, mute, volume adjustment, answer, hang-up, pause the current audio, and switch between car speakers and headset.

TI, Acoustic Technologies, and Adamya work with the customer to deliver software customization of the reference design to add extra functionality. TI will manage the software and hardware development. The complete development process can be completed in approximately 72 weeks at a total cost of approximately $260,000. The total bill of materials will run under $17.00 at 50,000 pieces or under $16 at 100,000 pieces.

The development process utilized model-based design methods that made it possible to model the complex acoustic behavior of multiple devices and generate test vectors for these device environments. The MathWorks' Simulink® was used to design and develop these models. Designers then created their own algorithm blocks in C code which could be integrated into this simulation environment for testing.

Engineers used the model to simulate the performance of the echo and noise cancellation systems in software by building scripts to depict typical operating scenarios. This approach made it possible to evaluate a wider range of designs and obtain far more insight on their performance while reducing the time and expense of the design process. They quickly modified the model and observed the performance change, making it possible to quickly optimize the design for optimal audio performance.

Once engineers were happy with the way the system behaved in the simulation, they generated a C code binary target for the TMS320C5000™ DSP platform. This binary target was easily created by using Code Composer Studio™ integrated development environment, which also allowed for source level debugging of the binary image while under test, allowing engineers to easily debug their design. The modeling target, in combination with Code Composer Studio IDE, allows engineers to validate the performance of their design on real hardware using simulated input. Later, minor optimization of code is allowed for evaluation with real audio inputs and outputs independent of the simulation model.

The best performance of echo cancellation and noise cancellation depends on a system solution which can dynamically adjust to the changing environments. The dynamic adjustment of system parameters provides fast response to environmental changes, avoiding intermittent echoes and noise that can be a problem with current generation technology. Testing this solution is difficult unless a good modeling environment is used. A key for the end product is choosing the right DSP technology which provides not only the signal processing horsepower needed, but also provides a development infrastructure which makes it possible to bring a product to market in a reasonable timeframe.

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