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HIGH CALL QUALITY DEMANDS EFFECTIVE ECHO CANCELLATION. NEW RECOMMENDATIONS AND CUSTOM-DESIGN APPROACHES POTENTIALLY COMBINE ECHO CONTROL WITH NOISE-REDUCTION AND LEVEL-CONTROL FEATURES, PAVING THE WAY FOR IMPROVED ECHO-CANCELLATION-EQUIPMENT DESIGN.

Tackling the echo-control challenge

Underestimating echo-control devices' contributions to modern telecommunications networks is easy. However, echo control, a highly exacting science can fundamentally affect how customers perceive network performance and therefore judge an operator's service. If handled effectively, echo control can significantly enhance call quality. Call quality contributes to customer satisfaction and reduces "churn"—the process of repeatedly losing and regaining customers.

These claims may sound bold, but field evidence clearly reveals the necessity for echo-cancellation equipment. In recent years, the increasing complexity of echo-cancellation applications for both wireline and wireless network environments prompted the need for devices that you can customize for a given application. Developments in echo-cancellation and voice-quality-enhancement equipment focus on the advent of integrated designs that use state-of-the-art ASIC and DSP technology. Many telecommunications engineers focus on the next generation of challenges, and Voice over Internet Protocol (VoIP) applications are emerging as the industry's most compelling new sources of revenue. Operators and carriers who tackle the thorny issue of echo control in the early stages of their infrastructure planning process usually achieve fast time to market and a high-quality platform from the outset.

From a design perspective, the future of echo control increasingly involves optimizing the echo-cancellation circuitry to handle a given network deployment's specifics. This customized approach combined with the requisite blend of features and integrated performance gives system designers access to a powerful set of tools that is an important means of achieving and maintaining carrier-grade voice quality. Industry pundits may talk about the data revolution's taking hold, but voice remains king in the revenue stakes. Various sources at the World Telecom Conference of the International Telecommunications Union (ITU) in October 1999 consistently ranked voice services as approximately 90% of revenue streams. Thus, you cannot underestimate the importance of preserving and delivering the best possible voice quality.

THE CAUSES OF ECHO AND THE GOAL OF TRANSPARENCY

Echo presents a major challenge in just about every type of network environment. Two types of echoes present in modern communications are line and acoustic. Line echo arises at a four- to two-wire interface, or "hybrid." Acoustic echo occurs when sound reflects off hard surfaces, such as walls, tables, and the inside of your windshield. Another concern, multipath echo, also a line echo, results from multiple-echo paths, or so-called tail circuits, in a phone call. For example, multiple tails can occur when one participant, caller number one, or the first tail, adds a paticipant to a conference call. Multiple tails are particularly likely to occur if the new participant, or second tail, is outside the building where caller number one is located.

The two echo types differ in cause and effect and offer a challenging problem when both combine on the same call in the same network, as is the case with digital cellular. In wireless networks, or digital-cellular calls, there is no hybrid on the mobile end of the call. The source of echo is acoustic coupling in the mobile handset, and this ambient echo depends on the environment in which the handset is located. Multiple delays can occur from the speaker's voice rebounding off surfaces at various distances from the handset.

By its very nature, the "wireline" end of the call has a "leaky" hybrid interface. As voice signals pass from the four-wire to the two-wire portion of the network, the higher energy level in the four-wire section reflects back on itself within the hybrid, creating echo. The parameter for defining the effectiveness of the hybrid to attenuate echo is echo-return loss (ERL). A high ERL indicates a low reflected-signal back to the talker and vice versa.

The extremely sensitive human ear responds to minute variations in pitch, volume, and tone. Echo with a delay greater than 50 msec is both extremely noticeable and intrusive, and it causes loss of concentration and loss of comprehension in extreme cases. You can detect virtually any echo, though customer complaints, such as poor quality or long delays, may appear to be unrelated.

In reality, both subjective and mechanical factors govern a customer's perception of quality. You aim to have communications mechanisms that are completely unnoticeable, or transparent, during the call. If you strain to hear the person you are talking to, the network is not transparent and interferes with the conversation. This transparency requirement has led manufacturers and carriers to introduce a number of measures. For example, a technology such as automatic level control (specified by ITU Recommendation G.169) can restore a conversation's proper speech levels and naturalness. Noise reduction is also an important feature. Consider the situation in which you are talking from a moving car. In this case, the voice can be hard to understand because of degradation that occurs from speech coding, road noise, and ambient car noise. Achieving transparency in this case requires you to use a noise-reduction feature that removes the intrusive background noise.

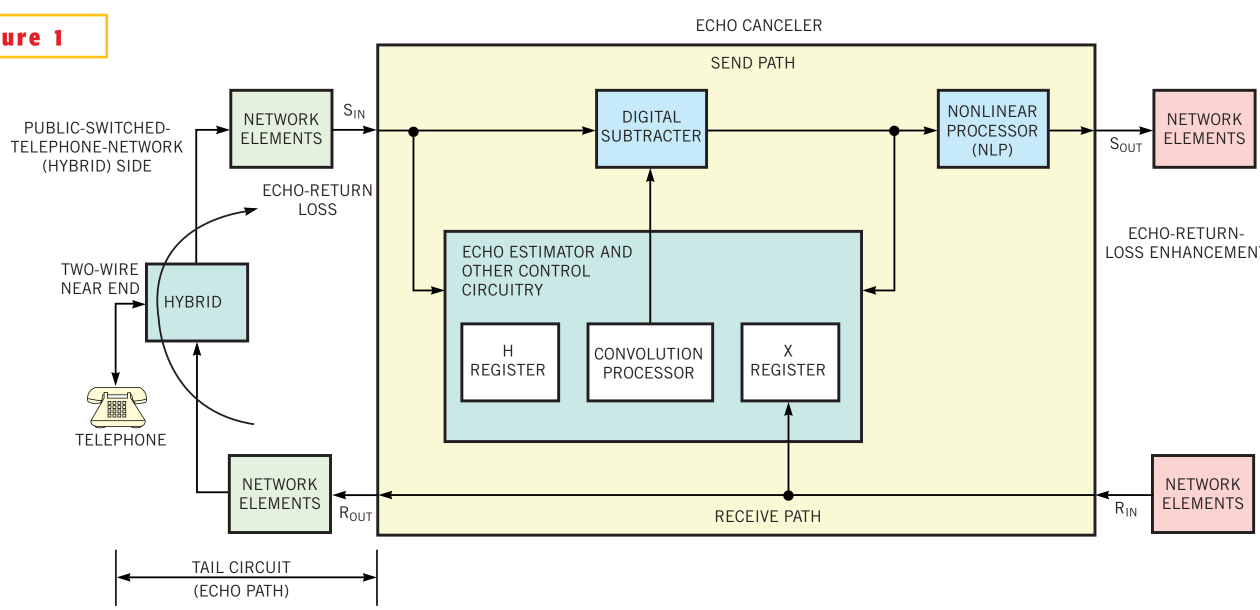
ECHO CANCELLATION IN ACTION

The measure of an echo canceler's ability to remove echo is ERL enhancement (ERLE). To completely remove the echo from the circuit, echo cancellation creates a model of the echo path, synthesizes a replica estimate of the echo, and cancels the echo by subtracting the estimated echo from the true echo. This process allows full-duplex speech between the near and distant callers and results in natural, interactive speech. You can place echo-cancellation circuitry anywhere in the digital circuit. However, the traditional location comes immediately before the A/D converter or as close to it as practically possible.

The modeling and subtraction process involves taking a sample of the received signal, storing the sample in the X register, and creating a model of the echo using an X register and the convolution processor (**Figure 1**). When any reflected signals pass back through the echo canceller, the canceller recognizes these signals as undesirable echo and subtracts the echo model from the actual signal to cancel out the echo.

Although the fundamental requirement is to remove the echo, the critical issue is how quickly and efficiently the equipment removes the echo. The echo-controller equipment's ability to merge level-control and noise-reduction techniques is also important. These techniques typically are prerequisites to handle the varying signal levels and high-background-noise levels that contribute to customer complaints. An echo canceler quickly must "converge," that is, adapt itself to the given network parameters at the start of a call and ensure that you are not subjected to any echo, even during the first second of the call. Subjective tests prove that the listener's perception of voice quality in the first few seconds of a

Figure 1



Echo cancellation involves modeling the echo path and subtracting a synthesized version of the echo from the actual signal.

call affects how long he or she will continue the conversation.

The echo canceler must also be able to introduce other quality enhancements, such as spectrally shaped and amplitude-matched comfort noise. Comfort noise is injected to overcome the "quiet" periods in a call introduced by the echo canceler's nonlinear processor (NLP). Without a comfort-noise injection, the resulting silence when the NLP is activated, causes the line to appear to go dead. The NLP is activated when the echo canceler has determined that there is no near-end speech, which results in full attenuation of all speech signals (real near-end speech and echo from the far end of the conversation). Making the correct determination of when to engage the NLP is one of the most difficult aspects of echo cancellation and often separates excellent echo cancelers from mediocre ones.

THE SCOPE OF VOICE-QUALITY ENHANCEMENT

Design engineers use sophisticated application-specific software to deal with the complexities of modern networks and to achieve high-performance echo-cancellation. The location of the echo canceler also allows for an extended role for the equipment. Usually located at the network point where many system elements converge, the ideal location of the echo canceler allows it to deploy sophisticated performance monitoring and call-quality-improvement techniques, such as noise reduction. Recent techniques capably reduce background noise by as much as 10 dB with no or minimal speech distortion, which is a significant improvement. State-of-the-art noise reduction works well on stationary noise, or noise that does not vary significantly over a period of time. The techniques also improve, to a slightly lesser degree, nonstationary background noise, such as background conversation, airport or train-station noise, and radio noise. The technology works by "learning" the spectral frequencies of the background noise and filtering accordingly. This technique improves speech intelligibility by detecting speech, modeling background noise during the gaps in the speech, and then eliminating the spectral components of the background noise.

This value-added facility of echo-cancellation equipment presents operators with a powerful argument for using the echo canceler in a wider, more creative role. Deploying echo cancellation alone wastes opportunity. But, if you combine it with noise reduction and application-specific software, it yields a number of cost-effective, space-efficient advantages.

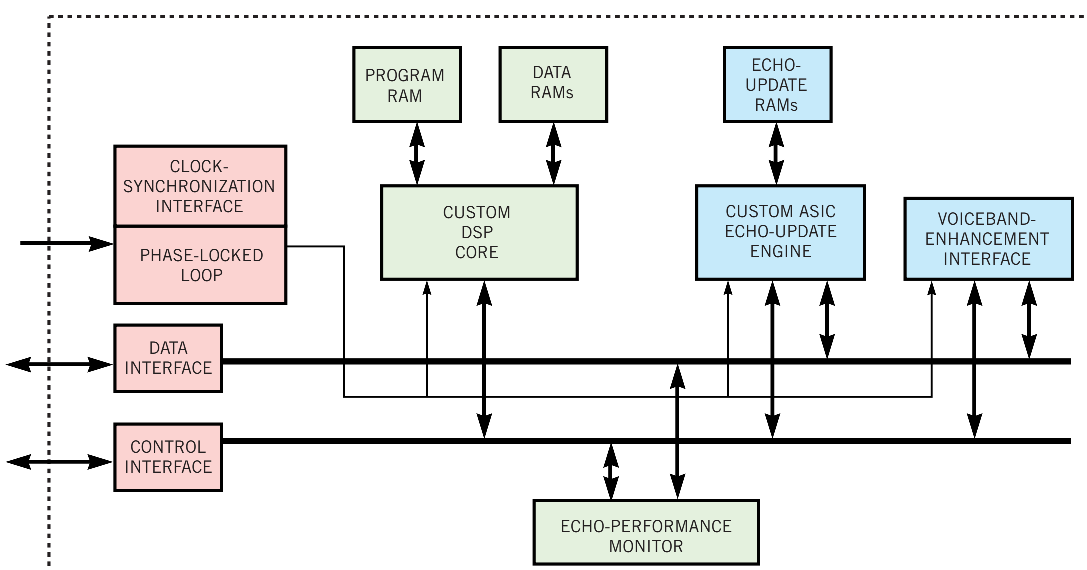
A CUSTOM VOICE-PROCESSING ENGINE

ASIC-based echo control proves to be valuable, particularly in wireless networks that have the inherent potential for more severe voice degradation than a typical public-switched-telephone-network application. However, the application's deployment of echo cancelers in both directions on every channel for optimum performance, necessitates high-investment levels. Consequently, you will need large volumes of ASICs that also have a proportionally higher price tag. This requirement has driven the development of a new combined ASIC/DSP µP that combines the performance, cost, size, and power advantages of ASICs with the benefits of the core-DSP approach.

The latest ASIC features are programmable, enabling designers to easily and cost-effectively incorporate additional features and functions. Considerable advances in the development of optimized code combined with specialized hardware can create a high-performance echo-cancellation combination. The wealth of field application data gathered from the use of ASICs in this area led to the development of several sophisticated algorithms, which you can modify and optimize for specific applications by incorporating a series of decision-logic and control-logic instructions. The performance improvements can be typically as many as four times that of general-purpose DSP-based designs.

The hybrid ASIC/DSP design in **Figure 2,** a custom voice-processing engine, can deliver impressive performance in

Figure 2



A custom voice-processing engine can perform a range of programmable and optimized functions.

complex wireless-network applications and can incorporate a range of new added features. A custom DSP environment can perform diverse programmable yet optimized functions, such as double-talk detection and nonlinear processing. However, the modeling process ideally suits an optimized ASIC, which can offer cost, size, and power benefits. Tone detection, level control, and noise reduction are potential value-added functions, which the combined approach to echo cancellation can deliver.

This custom-voice-processing approach also highlights the value of more effective integration of echo-cancellation technology into the core fabric of the network. By adopting an integrated approach, you can gain major advantages of performance, power, size, and functionality, considerably enhancing the echo-cancellation function in voice-enhancement and performance-monitoring applications. The merging of specialized ASIC- and DSP-based designs enables cost-effective incorporation of complex enhanced algorithms into custom voice-processing engines.

CONTROL ECHO IN VOIP APPLICATIONS

As you move from circuit-based networks into a packet-switched environment, the echo-control issue continually arises. For example, the VoIP application presents a particular challenge for echo control. The IP network requires a substantially higher standard of echo control than conventional time-division-multiplexed (TDM) networks, especially if the packet-switched network operates at the available bandwidth's limits. Potential impairments to voice quality cause a buildup of problems, including lost packets, network latency, and speech-compression-coding effects. Add echo to the mix, and you have a big problem. Providing an echo-free signal to the input of a VoIP voice coder critically affects the fidelity of voice across a packet network. As the coder attempts to build an accurate model of the input-speech signal, removal of echo in the signal is imperative for achieving the true potential of carrier-grade voice in the VoIP domain. Compared with TDM networks, VoIP applications introduce a range of problems, such as packetization delay, loss, and jitter that impact voice quality. Coding compression can also cause voice-quality impairments. The combination of these potential impairments results in a considerably more complex design scenario than for TDM networks.

COMPETING EFFECTIVELY WITH THE WIRELINE SERVICES IS ARGUABLY THE BIGGEST CHALLENGE CARRIERS FACE.

The technical challenges involved in VoIP applications have implications for operators. The primary concern of first-tier operators is a high-quality connection, providing end users with an easy-to-use and consistent service. Second-tier operators understandably have a different focus, namely just making the connection in the first place. So evidence calls for differentiated services, combined with the goal of transparency to ensure a quality end-user service. The delivery of carrier-grade voice quality is clearly the objective, and competing effectively with the wireline services is arguably the biggest challenge carriers face. For VoIP applications, market differentiation does not result from coder deployment because one G.729 coder performs comparably with another. Without a doubt, voice quality will be a major deciding factor in the competition between newer packet-voice technologies and traditional wireline services. To be successful, packet-voice technology must rely on the deployment of sophisticated voice-quality-enhancement technology.

G.168: WHAT PRICE COMPLIANCE?

You can therefore argue that voice quality will reign supreme in VoIP applications and the level of quality will determine an operator's level of success. Internationally recognized standards are an essential foundation, with the ITU-T G.168 recommendation acting as the linchpin for echo-cancellation and voice-quality-enhancement-equipment testing. The latest version of this recommendation, G.168 (2000), is scheduled to appear this month. Although G.168 provides a framework, this recommendation has some flaws and fails to reflect the real-world situation for echo control in a number of important areas. Achieving compliance with G.168 is indisputably a prerequisite in the selection of equipment, but compliance may not tell you everything you need to know about the equipment's performance in your network.

The G.168 recommendation is unique among equipment and function recommendations of the ITU because it does not define the echo-canceler algorithm. The recommendation enables manufacturers to develop their own algorithms and features, as long as the algorithm can pass the G.168 tests. The recommendation does not define the echo canceler's performance; rather, it provides a standard test configuration with pass/fail criteria. Also, G.168 does not define the interface to the network, and manufacturers can develop their own proprietary user interfaces. The only control requirements for echo cancelers in G.168 are tone-disabling via 2100-Hz modem- or fax-tone signaling and bit transparency when the system disables echo cancellation.

The fact that manufacturers can develop proprietary echo-cancellation algorithms and system interfaces makes echo-canceler performance and control requirements essential, and it focuses attention on the revision of G.168. Study Group 15 is considering a manufacturer-sponsored three-part solution, designed to address the lack of specified requirements in G.168. The first element is a true equipment recommendation; the second, a set of standard test methods; and the third, a document giving guidance to users of echo cancelers. With these steps in place, the industry can efficiently evaluate the all-important element of echo cancellation.□

AUTHOR'S BIOGRAPHY

Mike Kropotkin, the manager of application engineering for the Coherent OEM Division of Tellabs (www.tellabs.com), leads the Applications Engineering Group in assisting customers in integrating coherent OEM echo-cancellation and voice-quality-enhancement products in their own products. Kropotkin has 17 years of experience in product development in the telecommunications industry and has a BSEE from Fairleigh Dickinson University (Teaneck, NJ).