FEATURE

Adaptive Balance in Wireless Local Loop Systems

Achieving wireline quality in wireless devices is not an easy challenge to overcome. Wireless local loop requires more elaborate transhybrid balance than traditional wired telephony.

By Duncan Ashworth and Herbert Chen

Since the advent of the telephone system, copper wire traditionally provided the link in the "local loop" between the telephone subscriber and the local exchange. In spite of the 600 million telephone lines currently in use, four-fifths of the global population have never used a phone. This situation is rapidly changing as countries realize the relationship between a solid communications infrastructure and national prosperity. Countries with developing economies are racing to deliver cost-effective plain old telephone service (POTS) to hundreds of millions of impatient future subscribers. Installing twisted-pair bundles throughout the countryside or through crowded metropolitan areas is a slow, often uneconomic proposition. In some countries, wire laid between the customer premise and the local exchange in daylight hours is ripped up at night, to be sold for the value of the copper. Additionally, backhoes don't follow Moore's law. Installing a twisted-pair–based public infrastructure requires immense amounts of time and money.

Wireless local loop (WLL) technology, used to complete the "last mile" of the subscriber loop, appears assured of being the technology solution of choice for the exploding worldwide telecommunications market. However, while WLL avoids the costs and delays associated with laying copper cable as an infrastructure or bypass solution, the convenience of a wireless solution presents new technological challenges. Key among them is achieving wireline voice quality in a wireless solution.

For equivalent performance, WLL systems require more elaborate transhybrid balance networks than traditional wired telephony systems. Due to the longer time delays in the speech path and the unpredictability of the terminating impedance, it is impossible to choose a single compromise balance network that will consistently deliver ideal transhybrid performance in a WLL system. An adaptive balance algorithm patented by AMD and embedded in the DSP microcode of the codec/filter products addresses this problem. This algorithm constantly monitors the return echo and adjusts a summing-cancellation signal so that the transhybrid balance is always optimized.

The network

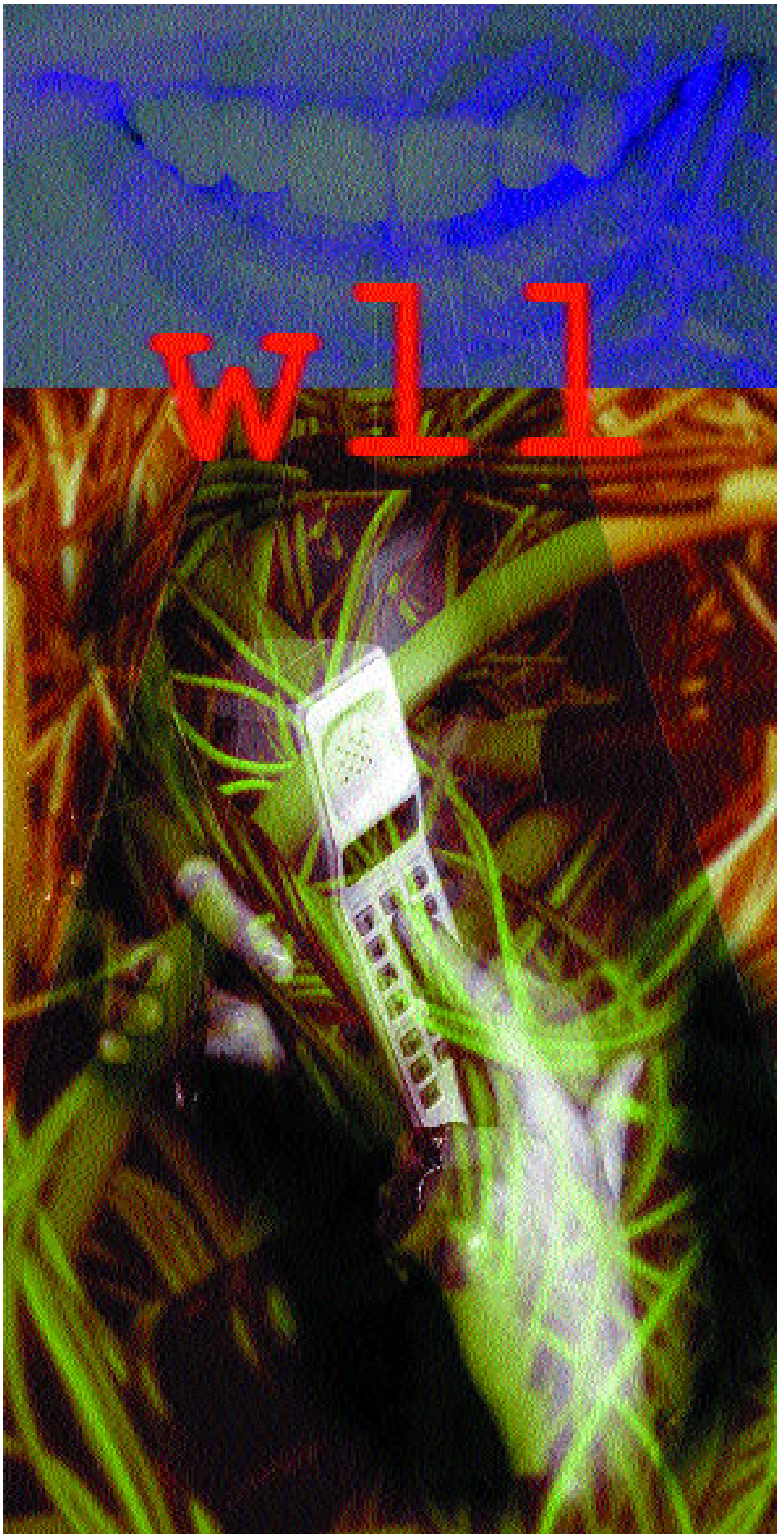
In a nonmobile wireless local loop system, the wireless basestation typically connects to the local exchange via an interface standard such as V5.2 or TR303. The customer premise equipment consists of a box on the side of a house, which contains the radio unit and synthesizes the POTS interface. The customer terminal equipment (telephone, fax, etc.) then plugs into the house side box in the same manner as with a wireline system.

The primary differences between a WLL system and a wired system are that the WLL system typically incorporates some type of speech compander, such as adaptive differential pulse code modulation (ADPCM), and the final wired loop from the box at the house to the terminal equipment is very short. As a consequence of using a wireless system, round-trip delays of up to 40 ms can occur with the dominant source of delay being the baseband protocol processor.

A potential source of echo is caused by the mismatch in impedances between the customer's terminal equipment and the balance network residing in the POTS interface. In short loops, such as those in a WLL network, the 2-wire impedance is dominated by the terminal equipment only, since the line is so short its impedance is insignificant.

By contrast, in a traditional wireline system, the line can be very long, in which case the characteristic impedance of the line dominates, and the impedance of the terminal equipment becomes irrelevant. For this reason, it is usually possible to achieve acceptable performance by using a fixed but compromised balance network in a wired system.

Transhybrid balance is the name given to the degree of echo cancellation. Transhybrid balance is the ratio



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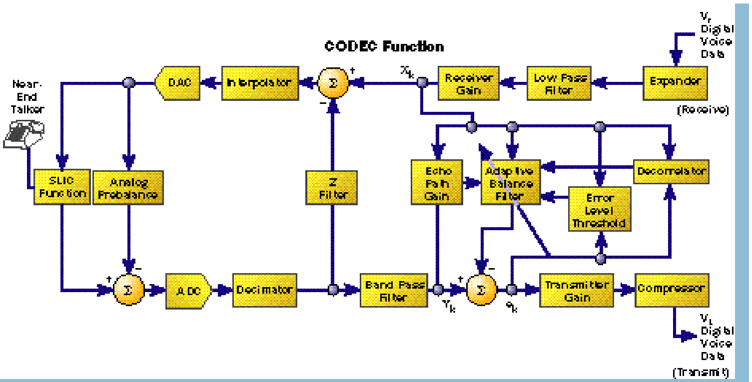

FIGURE 1: *Subscriber line interface.*

of the reflected signal to the transmitted signal (as viewed from the 4-wire side), and is defined by the equation: THB = 20 log (Vt/Vr), dB where Vt is the voltage of the echo signal and Vr is the transmitted voltage (Figure 1).

In a WLL system, it is desirable to perform the transhybrid balance function near the terminal equipment where the echo delay is a minimum. An attempt to perform this function in the wireless basestation or at the local exchange would require a more complicated network because of the increased delays the signal encounters while passing through the companders in the basestation and the box at the house. Due to the complicated nature of automatic transhybrid balance, it is best handled by a digital signal processor (DSP). If the codec function is DSP-based, the balance network can be implemented in the programmable digital domain as shown in Figure 1.

The general implementation of automatic balance is handled in the following manner: The adaptive balance filter performs an estimate of the echo-path impulse response and dynamically adjusts a set of digital-filter coefficients to create an echo replica. This echo replica is inverted and summed into the receive path to cancel the echo component in the receive signal, Y_k.

Echo problems in WLL systems are more pronounced than in wireline systems, due to the additional delay. Subjective tests indicate that the tolerable amplitude of the echo diminishes as the delay increases.[1] In other words, more echo cancellation is required for longer delays (Figure 2). The adaptive balance network consists of four primary blocks. The central block in the adaptive balance filter implements the adaptation algorithm itself. The three remaining blocks – echo path gain, error level threshold, and decorrelators – are control blocks that the adaptive balance filter on and off as described below.

The adaptation algorithm

One of the common adaptation algorithms is the least mean squares (LMS) algorithm.[2] Here, the criteria is to minimize the mean value of the squared residue error, and the filter coefficients are updated according to the stochastic steepest descent algorithm. The error signal at time sample k (Figure 1) is simply the difference between the echo replica $x_k \bullet h_k$ and the received echo signal y_k or:

$$e_k = y_k - (x_k \bullet h_k)$$

To reach the minimum mean square error, it is necessary to follow the gradient of the energy of the error with respect to h_k. When the gradient reaches zero, then minimum mean square error is obtained. Expressed algebraically:

$$\frac{de_k^2}{dh_k} = -2 \bullet e_k \bullet x_k = 0$$

Therefore, the adaptation of the filter coefficients should be:

$$h_{k+1} = h_k + 2u \bullet e_k \bullet x_k$$

with u being a constant step size.

If there is more than one tap in the balance filter, for instance N taps, the adaptation algorithm becomes:

$$h_{k+1,j} = h_{k,j} + 2u \bullet e_k \bullet x_{k-j}$$

$$\text{for } j = 0 \text{ to } N$$

A modification of the LMS algorithm that results in a much simpler calculation is called sign-based LMS algorithm or sign algorithm, and is described by the equation:[3]

$$h_{k+1,j} = h_{k,j} + \Delta \bullet \text{sign}(e_k)$$

$$\bullet\ \text{sign}(x_{k-j})$$

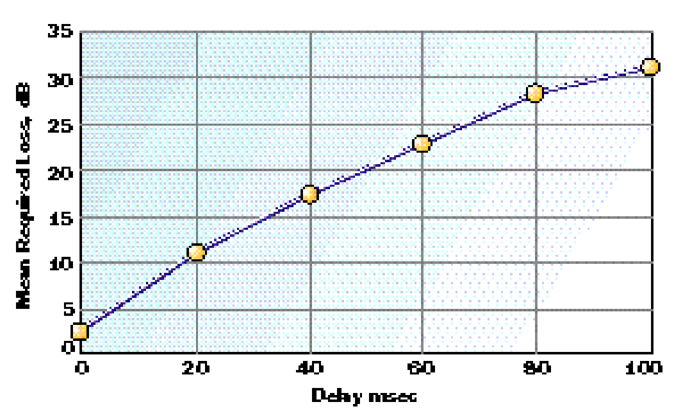

FIGURE 2: *Talker echo tolerance for average telephone user.*

where Δ is a predetermined incremental value.

The sign algorithm uses the sign of the error signal sample e_k and the sign of the receive signal sample x_k. An exclusive OR function performed on these signs yields the direction of the coefficient value change as shown in Table 1.

One advantage of employing the sign algorithm to perform coefficient updating is that only addition and subtraction are required to perform an update. Accordingly, the updating is readily performed by an simple arithmetic logic unit (ALU). Furthermore, the sign operations can be easily implemented by an exclusive OR logic gate. Two major inferior performances relative to conventional LMS are expected by using the sign-based algorithm. One is the higher updating noises caused by the coefficient adaptation and the other is the lower converging speed due to finite-adaptation step size. Therefore, a mechanism is required to stop adaptation once a convergence is reached to prevent excessive adaptation noise. The speed of convergence can be accelerated as well by choosing a non-linear adaptation step size.

The transfer function of an 8-tap sign algorithm-adaptive balance filter (Figure 3) can be described by the equation:

$$H(z) = h_0 + h_1 \bullet z^{-1} + h_2 \bullet z^{-2} + h_3 \bullet z^{-3} + h_4 \bullet z^{-4} + h_5 \bullet z^{-5} + h_6 \bullet z^{-6} + h_7 \bullet z^{-7}/(1-a\bullet z^{-1})$$

while the error signal at time instant k becomes

$$e_k = \sum_{j=0}^{7} (y_k - h_{k,j} \bullet x_{k-j})$$

The 8-tap FIR digital filter with coefficients h_0 to h_7 provides the main transfer function. The FIR filter by its definition has an impulse response of finite length due to its limited number of taps. The 8-tap FIR filter, in some cases, may not be

TABLE 1: Coefficient change.

sign(e_k)	sign(x_k)	Direction
+	+	Increment
+	-	Decrement
-	+	Decrement
-	-	Increment

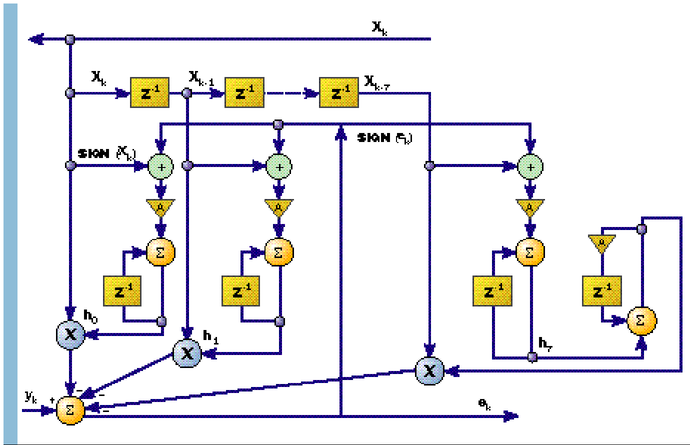

FIGURE 3: *Adaptive balance filter using sign algorithm.*

able to replicate a long trailing impulse response. Hence, a one-pole IIR filter with h_7 as the input is cascaded to the FIR filter for extending the impulse response to a much longer duration by choosing an appropriate coefficient a.

Convergence control

An uncontrollable or excessive adaptation of a balance filter is undesirable due to the resulting divergence of the filter coefficients. This is especially true for a single-tone excitation, which may reach a converging point and then start to diverge. Therefore, a residual-error level threshold (ELT) detector must be provided to stop the adaptation process once the error energy level of e_k is less than the receive energy level x_k by a certain threshold.

Double-talker control

Double-talker control is necessary to prevent erroneous adaptation when both parties on the line talk at the same time, since the near-end talker's voice can be misinterpreted as an echo. Without proper control, adaptation will be misguided due to the merging of the near-end speech and echo, resulting in noisy transmission.

A double-talker condition exists in every telephone connection during conversation phase as well as in signaling phase when the dial tone and DTMF signal are present at the same time. A double-talker detector is also critical for a reliable high-speed modem connection because of the duplex operation performed and its sensitivity to the adaptation noise. Failure to detect a double talk and then inhibit the adaptation during a modem connection can result in a high bit-error rate or a connection hang-up.

A conventional way of detecting a double-talker condition is to compare the signal level at the near- and far-ends using energy-averaging filters. If the near-end signal level exceeds the signal level of the far-end by a predetermined threshold, a double-talker condition is flagged and the adaptation is inhibited. Since the near-end signal level is varying and contains an echo, too low a threshold (oversensitive detection) will turn off the adaptation because of high echo level; while too high a threshold (undersensitive) will continue the adaptation even when the signal level of the near-end talker is high. Therefore, it is very difficult to provide a reliable double-talker detection by using this simple conventional way.

Because of this deficiency of the conventional method, a sign-based decorrelation detector is added to provide reliable double-talker detection.[4] During the adaptive process, the decorrelation controller will detect cross-correlation between a far-end signal and echo residue. If the detected correlation value is below a certain threshold, indicating that the two signals are decorrelated, the adaptation will be stopped (Figure 4). At such a point, a proper echo cancellation has been achieved. Conversely, when the detected correlation value exceeds a threshold, the adaptation will be resumed until the correlation values fall below the threshold again. The decorrelation detector is independent by its nature of the near-end talker, as long as the far-end and near-end talker signals are uncorrelated. In any case, the controller maintains its capability to detect a signal correlation and implement adaptation control even in the presence of a double-talker condition.

The sign-based decorrelation detector (DCR) detects the correlation value by using the sign bit of the two signals. The circuit implements the following sign-based decorrelation equation:

$$\sum_{k=0}^{\infty}\sum_{j=0}^{7} \text{sign}\ (e_k) + \text{sign}\ (x_{k-j})$$

Certainly, the manipulation of sign bits is much simpler than the multiplication required in a conventional cross-correlation calculation.

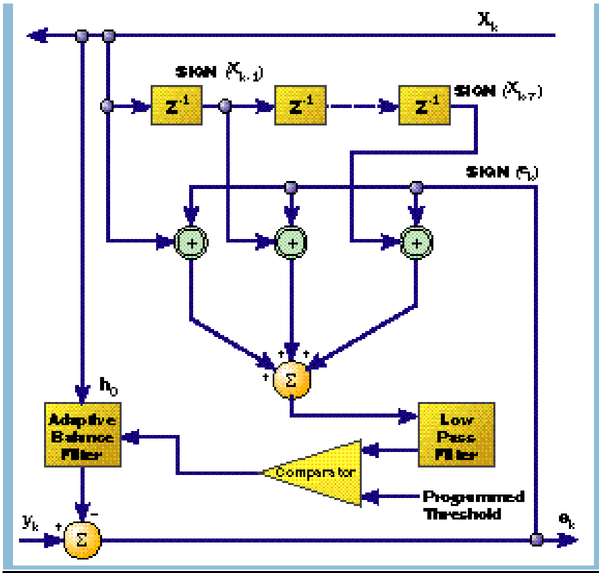

FIGURE 4: *Decorrelation detector.*

Prebalance circuit

Another peripheral circuit required to assist the adaptive-balance filter is the prebalance circuit. Two practical reasons that a prebalance circuit is needed are to ease the dynamic range requirement of the A/D converter and to reduce the echo level at the adaptive-balance input. A pre-balance circuit is a coarse echo canceler located in front of the fine tuning adaptive-balance. The pre-balance network is selected such that it can greatly cancel the echo reflected from a nominal impedance termination, for instance 600 ohms or 900 ohms.

In theory, a pre-balance circuit alone is able to reduce the echo by 6 dB in the worst case and 40 dB for the best cases under various line conditions. One extreme case is the caller ID, or on-hook transmission, where the loop is nearly open and a simplex modem signal is received. A totally reflected high level of echo is expected to clip the A/D converter input if an analog prebalance is not used. It also pushes the adaptive filter coefficients to a high value where a larger step size occurs if a non-linear incremental adaptation is employed. Moreover, lacking prebalance will complicate the solution to the loop instability described in the next section. A simple prebalance circuit with an analog-summing amplifier will solve these problems.

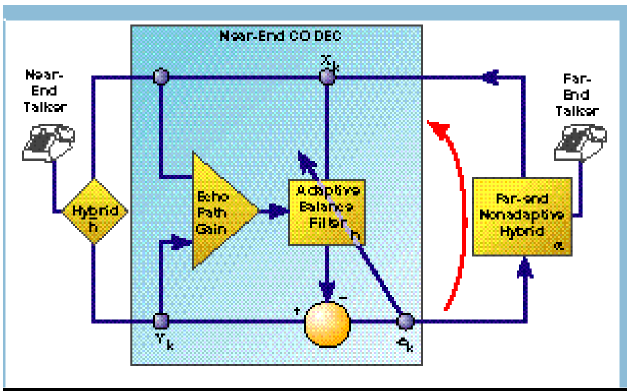

FIGURE 5: *Model of bursting loop instability.*

Bursting loop instability

An undesirable phenomena called "bursting" has been observed when an adaptive balance network is employed in the local loop.[5] Bursting is characterized by periods of successful echo cancellation followed by periods of wildly oscillating noises. In other words, bursting is an alternating sequence of adaptation converging and diverging. The 4-wire loop gain is $|\alpha \bullet (\bar{h} - h)|$ where α is the gain across the hybrid at the far-end, $\bar{h}$ is the response of the near-end hybrid, and h is the response of the adaptive filter, as shown in Figure 5. If the 4-wire loop gain is larger than 1, the resulting feedback loop is unstable. The instability is triggered by the near-end signal, which goes out to the far-end and returns back to the receive side. This is especially true when the far-end transhybrid loss is poor, which means α is large. In this situation, the adaptive-balance filter is trying to cancel the near-end signal, resulting in divergence with high noise. A "self-stabilization" process is followed because the spectrum-rich diverging noise at the receive end will force the adaptive filter back into convergence. The bursting happens more often when the transmission from far-end is quiescent and the near-end signal is periodic.

In the absence of bursting control, an annoying noise will be heard during a telephone conversation. A control circuit called echo path gain (EPG) is added to prevent the bursting behavior. The EPG circuit consists of two signal-level detectors and one comparator. If the signal power level at near-end exceeds that at the receive path by a predetermined threshold, the filter adaptation will be stopped. The threshold value should be selected equal to or smaller than the worst case transhybrid loss at the far-end. However, a low threshold (oversensitive) may result in a false detection triggered by a normal echo, rather than by a near-end signal. This oversensitive EPG problem can be solved by the prebalance circuit, which reduces the echo and makes the detection of near-end signals more accurate.

Modes of operation

Balance filters can be divided into three general classifications regarding their operating modes. The first mode is that of a fixed balance network that is similar to conventional schemes. The second is an adapt-and-freeze mode whereby the adaptive balance filter automatically determines its best set of balance coefficients and then stops adapting. This is typically done at the beginning of a call. The final mode is the fully adaptive mode whereby the algorithm is continuously on (when meeting the threshold criteria described above) and is always trying to optimize the transhybrid loss. This is useful in cases when the load in the terminal equipment changes during the call, such as when someone picks up an extension phone. In the case of a WLL system, this dramatically changes the impedance seen by the codec/filter, and, therefore, the continuous adaptive balance feature is most desirable.

Wireless local-loop systems are more cost-effective to deploy than wireline systems under certain scenarios. However, echo problems require new solutions to provide acceptable performance. These echo sources, caused by long time delays and varying impedances, can be automatically compensated for near the customer terminal equipment to give optimum performance even with changing loads.

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