

AUTOMATIC IMPEDANCE SYNTHESIZER

This invention relates to telephone line circuits and related circuitry forming the interface between analog and digital telephone lines and trunks and a digital switching network. Specifically, this
5 invention relates to a digital line circuit providing automatic matching of the impedance of lines/trunks terminated by an electronic hybrid and to the automatic synthesis within the hybrid of the line matching impedance without using discrete components and with
10 minimized power dissipation.

In the prior art of two-to-four wire conversion, the problem due to impedance mismatch at the telephone central office between the subscriber lines/trunks and the terminating impedance is well
15 known. It causes poor return loss characteristics and reflections or echoes due to poor operation of the two-to-four wire hybrid, which for proper operation, needs the line and terminating impedances to be equal in both phase and magnitude over the
20 bandwidth of the telephone channel. Passive fixed terminating impedances to date are a compromise, except at a specific frequency, since such compromise impedance is either a series or parallel combination of a resistor and capacitor. Such a simple impedance
25 results in a poor match for the line impedance. Typically, the hybrid which performs two-to-four wire conversion depends on a relatively close match between the line and the terminating impedances and for optimum

performance, a good match over the range of frequencies of interest is required. Compensating for this mismatch has been attempted in the prior art with balance or "building-out" networks which are a part of the hybrid, and which, for a given line, represents a custom tailoring of the circuit.

A further problem due to impedance mismatch at the far end of the line is that if that end is improperly terminated, unwanted reflection or echo is returned to the near end. If the near-end terminating impedance equals the line impedance and if the transhybrid loss is zero, then optimum performance is obtained from the near end equipment. The far end echo can also be minimized by using known echo cancellation techniques, which depend upon a prior knowledge of the locally transmitted signal, and an assumed non-correlative relationship between the near end transmitted signal and the far-end received signal.

Using adaptive equalization, with or without training signals, the correlative portion of the far-end reflection present in the near-end receive signal can be regenerated, using an adaptive equalizer, and subtracted from this locally received signal.

Conditions for proper operation of the adaptive equalizer must be met. The equalizer must have an adequate S/N ratio to allow for equalizer convergence, and a linear network characteristic. Sufficient energy must be present across the band to allow for correct

feedback control signals for digital filter tap adjustments. There must be an absence of "double-talk" during the equalization process. Next, and most significantly, the digital local office which must interface to analog subscriber loops must now add two two-to-four wire converters to interface these loops. Previously, with analog central offices, no hybrids were required.

These newly introduced hybrid interfaces can introduce reflections or unwanted return signals. Previously, for analog switches these additional hybrids were not present. Thus, without improved
5 performance in the hybrid, the digital office is potentially poorer in performance than its analog predecessor. The problem of "singing", or more explicitly, potential instability of the network in a Nyquist sense results from the unwanted feedback
10 due to the two-to-four wire conversions; and the system can conceivably oscillate if proper precautions are not taken. Classically, the VNL (Via Net Loss) Plan takes this condition into account by appropriately inserting attenuations throughout the network in a
15 regulated manner, and specifying the transhybrid loss to meet some minimum criteria at those points in the network where two-to-four wire conversions take place.

Previously, in arriving at the attenuation which could be inserted in the existing networks,
20 consideration was given to those offices (or circuits) which used two-to-four wire converters; namely, trunks. For analog local offices not needing two-to-four wire converters, zero attenuation was allowed, and the insertion loss allowable was and is only a
25 few tenths of a decibel. Thus, the problem of designing a digital local office to provide equivalent analog performance in an analog environment is aggravated by the existing VNL Plan. Experimental results show that the addition of attenuation (4-db)
30 in the local office to overcome this problem served only to reduce the "Grade of Service", i.e. in comparison, telephone users can detect the poorer performance due to the added attenuation.

Automatic equalizers per se are well known in the field of digital data transmission, with U.S. Patent Nos. 3,579,109 and 3,984,789 being illustrative. A digital adaptive equalizer is described in U.S.

5 Patent No. 3,633,105. U.S. Patent No. 3,798,560 describes an adaptive transversal equalizer using a time-multiplexed second-order digital filter.

An object of the invention is to provide automatic line impedance matching at call set-up by providing an electronically synthesized terminating impedance at the line circuit which matches the subscriber line impedance.

According to the invention there is provided a circuit for automatically digitally synthesizing an output matching line terminating impedance including:

- (a) analog to digital conversion means for converting analog communication signals occurring at circuit output terminals to digital signals;
- (b) means for summing said digital signals with input digital communication signals providing a composite digital signal input having negative feedback;
- (c) digital filter means having said composite digital signal input coupled thereto for directly establishing a predetermined output impedance characteristic, said digital filter having alterable filter coefficients, and having a digital signal output;
- (d) digital to analog conversion means for converting said digital signal output to an analog voltage signal; and
- (e) means for converting said analog voltage signal to an analog current having a high output impedance characteristic compared to the matched line and to the synthesized impedance such that said analog current is coupled to said circuit output terminals and to said analog to digital conversion means.

Embodiments of the invention will now be described with reference to the accompanying drawings, in which:-

Figure 1 shows a known electronic hybrid circuit.

Figure 2 shows a digital line circuit embodying
5 the invention.

Figure 3A illustrates the digital synthesis of an output admittance.

Figure 3B is a simplified block diagram of impedance matching in a digital hybrid.

10 Figure 4 is a transconductance amplifier.

Figure 5 shows a recursive automatic equalizer.

Figure 6 is a block diagram of a generalized digital filter.

Figure 7 is a block diagram of an equalizer.

15 Figure 8 is a block diagram of a generalized digital two-to-four wire converter.

Figure 9 is a block diagram of a preferred embodiment of an equalizer used in conjunction with the invention.

20 Figure 10 is a first memory portion of the equalizer of Figure 9.

Figure 11 is a second memory portion of the equalizer of Figure 9.

Figure 12 is a flow chart of the equalizer
25 operation.

Figure 13 is the multiplexer/distributor used with the present invention.

Figure 14 illustrates the interface connections between the equalizer of Figure 9 and the multiplexer/
30 distributor.

Figure 1 is a state-of-the-art electronic hybrid, which:

(1) matches the line impedance seen at its terminals 12 and 14, and shown as a lumped impedance
35 Z_L at 16.

(2) minimizes the unwanted return signal from transmission path 1 by providing a cancelling signal on transmission path 2 such that at the output of the receiving amplifier 18 the transmitted signal is reduced or minimized. The operation of hybrid 10 is as follows:

Impedances Z_8 at 20, Z_L and Z_3 , Z_4 at 22 and 24, respectively, form a bridge circuit with the transmitting amplifier 26 acting as the driving source and the receiving amplifier 18 acting as the null amplifier. If the bridge is properly balanced, the output of the receiving amplifier 18 is zero for any signal being transmitted. At the same time, any signal appearing across the tip and ring lines is received at the output of amplifier 18. Thus, unwanted transmitted return signals are eliminated, which provides the two-to-four wire conversion function of the hybrid. The impedance seen by Z_L is Z_8 in parallel with Z_1 at 28. If Z_1 is much greater than Z_8 , then the transmission line sees Z_8 as a line terminating impedance. In practice, Z_8 is fixed at a set value, typically 900 ohms in series with 2.2 microfarads, and Z_3 and Z_4 are varied or selected to give the proper balance, thus minimizing the unwanted return signal. The drawbacks of this circuit are:

(1) Since Z_8 does not equal Z_L for all transmission lines, the return loss of the circuit 10 as defined by:

$$\text{Return loss} = 20 \log_{10} \frac{Z_L + Z_8}{Z_L - Z_8} \quad (\text{Eq. 1})$$

is not infinite, which is the optimum value to minimize reflections from signals arriving from the far end.

(2) Z_3 and Z_4 must be selected on a compromise basis that the same circuit 10 can accommodate different lines, or Z_3 and Z_4 must be selected manually or automatically by an appropriate set of control
5 signals which activate an analog switching network to place appropriate values of Z_3 and Z_4 in the circuit, according to some control process.

(3) The majority of the impedances are complex
10 impedances and thus need both resistive and capacitive elements.

(4) The circuit is basically analog and does not lend itself to low cost, low power, high density, digital LSI techniques.

(5) The circuit needs high cost, stable and
15 precision components to perform properly over the life of the equipment with which it is used.

(6) The input/output on the 4-wire side of the circuit are from the decoder and coder circuits in the codec used in a digital system. Thus to function
20 properly, the line circuit for a digital office needs a separate codec and hybrid, in addition to the normal codec filters.

Figure 2 is a block diagram of a digital line circuit embodying the invention wherein the digital
25 two-to-four wire hybrid function is performed with the elimination of the previously associated discrete analog component. Also, the circuit provides automatic digital synthesis of line terminating impedance for any transmission line characteristic, eliminating the
30 previously used analog component. These features enable an LSI circuit to implement the complete electronic hybrid and impedance matching functions as part of an overall codec function, which functions can be implemented on a single LSI chip without extensive analog external
35 balancing or adjusting circuitry. This results in a

low cost, low power, high density and a highly reliable telecommunications line-circuit. The techniques upon which this circuit is based will now be described.

- 5 Referring back to Figure 1, if Z_g could be made exactly equal to Z_L over the range of lines for which the impedance is intended, then Z_3 and Z_4 could be equal resistive elements, eliminating return signals and providing an exact impedance match to maximize
10 return loss. Figure 2 achieves this using digital filtering and feedback techniques without using the heretofore used analog components. In Figure 2, the impedance seen looking into the tip and ring terminals 102 and 104 is designed to match the line impedance.
- 15 The line terminating impedance Z_g of Figure 1 is digitally synthesized by the digital loop of circuit 100 formed when switch 106 is operated providing feedback from the coder path 108 to the decoder path 110 via line 112.
- 20 Figure 3B is a simplified block diagram of such digital synthesis. The transconductance or g amplifier 200 is a device which converts an input voltage V_x to an output current $-gV_x$. It has an infinite output impedance or zero output admittance. Such a circuit is
25 well-known. In its simplest form, it can be equated to a pentode vacuum tube or Field Effect Transistor (FET) in which the plate or drain currents are proportional to the grid or gate-to-drain voltages respectively. An example of such a circuit is shown in
30 Figure 4 at 200, wherein a high open loop gain operational amplifier (gain = 10^6) is used with a resistance whose conductance is g mhos.

Amplifier 200 in practice is the normal output amplifier of the decoder 114 and is not an added
35 component, being shown functionally separately for

39

explanatory purposes. The decoder 114 is a device which converts a digital signal into an analog voltage and this function can be implemented by well-known digital to analog devices. The H filter is a digital
5 filter with programmable coefficients which can be implemented using well-known techniques in digital filtering, as will be explained. The H filter 116 is preceded by a digital summing node 118, see Figure 3B. The function of the prefilter 120 is to limit the
10 bandwidth of the decoder 114 input signal to the region of interest, i.e., below the sampling rate of the coder 122, to avoid spectral folding and to optimize the signal-to-noise characteristics of the coder 122. Prefilter 120 may comprise a simple two pole passive
15 low-pass filter since it is analog in nature.

The coder 122 is preferably a sigma delta type described by Candy, et al. The coder post filter 124 performs bandwidth limiting and decimation, and provides an overall low-pass, flat response in combination with
20 the prefilter 120, such that the composite gain is equal to unity in the frequency band of interest, and rolls off monotonically outside this band. With the system thus defined, the response and output admittance at the points indicated at Figure 3B can be shown to be:

25
$$\frac{V_o'}{V_{in}} = \frac{-gH}{Y_L' + gH} \quad \text{where } Y_L' = Y_L + Y_{bf} + Y_{pf} \quad (\text{Eq. 2})$$

and the output admittance, including Batteryfeed 125, prefilter 120 and line loads, can be represented as:

$$Y_{out} = Y_L' + gH \quad (\text{Eq. 3})$$

In Figure 3A, the digital synthesis of an output
30 admittance, is shown by a simplified and generalized circuit usable for numerous applications wherein it is

desirable to digitally synthesize a specific output
admittance in which A/D and D/A converters are used
instead of the specialized telephony forms of A/D and
D/A, or as coder/decoders. The functions of the A/D
5 prefilter 250 is to limit the bandwidth of the input
signal V_{in} on line 252 to prevent spectral folding due
to undersampling. After analog-to-digital conversion
at A/D converter 252A, the A/D post filter 254 "mops
up" to provide overall low pass unity gain character-
10 istic for the A/D path 256. The g Amplifier 200 and
digital H filter 116 are described herein elsewhere.
The D/A prefilter 258 "mops up" to insure the D/A path
260 gain equals gH. The D/A post-filter 262 eliminates
or minimizes the effects of the D/A quantization
15 process occurring in D/A converter 264. Quantizing
effects are defined as the noise due to the D/A error
in approximating a continuous analog output signal.

Where the prefilter and batteryfeed impedances
(admittance) are large (small) compared to the line
20 impedance (admittance), they can be neglected.
Normally, the battery-feed impedance is designed to have
this characteristic to prevent signal loss and the
coupling of unwanted signals into the codec. The pre-
filter can also be designed to have the characteristic
25 by utilizing a high series resistance or the high input
impedance of the gate input of a FET.

Under these conditions $Y_L' = Y_L$ and the output
admittance is:

$$Y_{out} = Y_L + gH \quad (Eq. 4)$$

30 If gH is made equal to Y_L then;

$$\frac{V_o'}{V_{in}} = -1/2 \quad (Eq. 5)$$

and the output impedance of the circuit matches the line impedance.

Correspondingly, the delayed transmitted signal V_{in} attenuated by $-1/2$ will appear at the summing node 270 of Figure 3B. If the digital F filter 272 provides an attenuation of $1/2$ and an absolute delay corresponding to the round-trip delay from the H filter 116 input to the coder post filter 125 output, then the output filter 272 cancels the unwanted returned transmit signal V_{in} at the output of the summing node 270, resulting in an output voltage V_o which does not contain any portion of the transmitted signal, V_{in} . Implementation of the F filter 272 is, under these conditions, simplified since it can be a simple shift-register type delay element. Attenuation of $1/2$ is accomplished by right shifting the binary value of the voltage V_{in} one place and rounding off.

For a more general application of this technique where admittances of the prefilter and batteryfeed type are not negligible but are known, g_H can be changed to eliminate the loading effects of the admittances on the line, i.e.:

$$g_H = Y_L - Y_x \quad (\text{Eq. 6})$$

where Y_x corresponds to their combined admittance.

Under these circumstances, the round trip gain (through coder and decoder paths 108 and 110 respectively) changes to:

$$\frac{V_o}{V_{in}} = -1/2 + \frac{Y_x}{2Y_L} \quad (\text{Eq. 7})$$

and the net admittance seen by the line at terminals 102 and 104 remains equal to Y_L . The effect of the second right hand term in Equation 7 represents the

effect of the modification of the H filter 116 to compensate for Y_x . This results in F filter 272 being modified such that:

$$F = \frac{+gH}{2Y_L} \quad (\text{Eq. 8})$$

- 5 in order to cancel the unwanted transmit signal returned to the receive output at V_o .

In sampled data Z transform notation, the digital H filter is of the form:

$$H(Z) = K_o \frac{Z^2 + K_1 Z + K_2}{Z^2 + ZK_3 + K_4} = K_o \frac{N(Z)}{D(Z)} = \quad (\text{Eq. 9})$$

10
$$K_o \frac{[1+K_1 Z^{-1}+K_2 Z^{-2}]}{[1+K_3 Z^{-1}+K_4 Z^{-2}]} = K_o \frac{N(Z^{-1})}{D(Z^{-1})}$$

where Z is a delay operator, K's are programmable coefficients, and N and D represent numerator and denominator polynomials. The filter has its poles and zeroes restricted to within the unit circle of the Z
15 plane.

Relating the above to the synthesized Y_L , this allows Y_L to be equivalent to any combination of resistors, capacitors and inductors having no more than two non resistive elements. This restriction is
20 not inherent to the technique, having been chosen to simplify the form of the H filter 116 from the viewpoint of a large scale integrated circuit hardware implementation to obtain a simple recursive digital filter having five (5) coefficients. In addition, limiting the number
25 of elements composing Y_L as described, represents a good practical approximation to the function. For applications other than telephone line circuits, the

definition of the H filter 116 may be extended according to the complexity of the admittances involved, based on the techniques described herein. For example, where Y_x has a non-resistive component, gH , which corresponds to $Y_L - Y_x$, can be designed to have third order polynomials in its numerator and denominator. This would allow Y_L as synthesized to meet the previous constraint of no more than two non-resistive elements.

Figure 5 shows the recursive auto equalizer 130. The need to establish the coefficients of the H filter 116 such that gH equals Y_L (or $Y_L - Y_x$) is satisfied by the equalizer 130 function. The "equalizer system 130", under control of the maintenance system routinely checks that the digital filter coefficients are correct in addition to establishing them initially. The equalizer operates in an off-line manner (no user call in progress) and can be time shared by N lines. Once the filter coefficients are established, these coefficients need only to be checked periodically, under control of the maintenance system, since any given line characteristic does not normally vary from day to day. This allows the equalizer to be time shared among a number of line circuits, thus amortizing its cost over a number of lines. Figure 5 shows the conditions under which the recursive equalizer operates. Switches 106 and 107 of Figure 2, when open under the control of the MULDIS during checking of the control coefficients, disconnect the F filter 272 and the feedback on line 112 from the coder path 108. The H filter 116 is then set in the bypass mode (H filter is effectively short circuited across input/output terminal, i.e., $gH = 1$).

Figure 2 can be described as follows:

The code path 108 includes coder prefilter 133, a sigma delta coder 135 and a decimating and low pass filter 137. The filter 137 output and F filter 272 output are summed at summing node 139 to derive the

intermediate receive output on line 141, which is filtered at receive audio filter 143. The output of filter 143 is the 4-wire receive signal of its line circuit, on line 145. The 4-wire transmit signal on line 147 is filtered by transmit audio filter 149. The output of filter 149, the intermediate transmit input, is coupled to H filter 116 via summing node 151, whereat the intermediate transmit signal is summed with the feedback on line 112, when switch 106 is closed, under control of multiplexor/distributor 130. Prior to decoding at decoder 114, the filtered transient signal is filtered by an interpolating filter 153. Programmable signal generators circuit 155 is shown for completeness, and reference is made to our Patent No. 1583635 (R.Treiber 1 Div A) for details of circuit 155. The muldis 130 includes means for distributing control coefficients for 1... N line circuits, shown generally at 157, and which includes a digital audio equalizer 159 and a digital reference generator 161.

A telephone subset at the subscriber end of the line is taken off-hook for example when it is undergoing maintenance. At the intermediate transmit input point, a reference generator 161 supplies a reference signal having uniform (equal) energy components within the frequency band of normal operation and zero energy outside the band. The output signal $O(Z)$ will be correct when, with reference to Figure 5,

$$E(Z) = 0 \text{ where } Z = \text{delay operator, and} \quad (\text{Eq. 10})$$

$$E(Z) = O(Z) = R(Z) Z^{-L} \quad (\text{Eq. 11})$$

The Z^{-L} term compensates for the known absolute sampling delays through the loop. Under these conditions:

$$\frac{N(Z^{-1})}{D(Z^{-1})} K_o = Y_L(Z) \quad (\text{Eq. 12})$$

which is the desired result and the resulting recursive filter coefficients K_0, K_1, K_3, K_4 have been correctly established and can be loaded into H filter 116 for normal operation. At equalizer 130 start-up, the H filter 116 is loaded with trial coefficients or the last previous coefficient values, which allows the equalizer to converge rapidly. The equalizer effectively solves a set of simultaneous partial differential equations which minimizes the mean square G between $R[nT-LT]$ and $O(nT)$ as a function of the coefficient K_k . This is represented as:

$$\frac{\partial G}{\partial K_k} = \overbrace{\sum_n [O(nT) - R(nT-LT)]}^{\text{error term}} \overbrace{\frac{\partial O(nT)}{\partial K_k}}^{\text{partial}} \quad (\text{Eq. 13})$$

according to established theory as for example Lucky & Rudin at BSTJ Nov. 1967 and Weiner - Time Series Analysis - MIT Press published 1964, Appendix B.

The outputs, $P_0(nT), P_1(nT)$, etc., represent the partial derivatives of $O(nT)$ with respect to the filter coefficients. These outputs, when cross-multiplied with $E(nT)$, summed, and periodically "sliced" by a slicer, which truncates and rounds off to the sum term, give the C_k outputs which allow the $N(z^{-1})$ and $D(z^{-1})$ polynomials to be updated with new coefficients according to the equation:

$$K_{k,\text{new}} = K_{k,\text{old}} - C_k \Delta \quad (\text{Eq. 14})$$

where Δ is an incremental value or factor for step adjustment. This slicing technique, and means for implementing the solution to Equation 14, is known in the auto equalizer field for non-recursive structures. The circuit of Figure 5, however, uniquely implements an automatic equalizer function for a recursive

48

equalizer structure. This circuit provides the P_k terms which take into account the interaction of the coefficients K_k , which heretofore has been considered one of the limiting factors in recursive equalizer structures. This may be one of the reasons that non-recursive equalizer structures have prevailed in the prior art, namely the inherent simplicity of obtaining the partial functions for the auto equalizers operating on the Mean Square Error Criteria. Such nonrecursive structure of the prior art requires 30-60 complex coefficients, whereas the present recursive structure requires only five (5) coefficients, and thus reduces circuit complexity and the amount of hardware.

In the block diagram of Figure 5, showing the recursive equalizer operation, the following equations apply:-

$$P_0(Z) = A(Z) + \frac{N(Z^{-1})}{D(Z^{-1})} \quad (\text{Eq. 15})$$

$$P_1(Z) = A(Z) + \frac{K_0 Z^{-1}}{D(Z^{-1})} \quad (\text{Eq. 16})$$

$$P_2(Z) = Z^{-1} P_1(Z) \quad (\text{Eq. 17})$$

$$P_3(Z) = \frac{-K_0 Z^{-1}}{D(Z^{-1})} P_0(Z) \quad (\text{Eq. 18})$$

$$P_4(Z) = Z^{-1} P_3(Z) \quad (\text{Eq. 19})$$

$$O(Z) = K_0 P_0(Z) \quad (\text{Eq. 20})$$

$$E(Z) = O(Z) - R(Z) \quad (\text{Eq. 21})$$

$$C_{k \text{ new}} = \text{slice } \sum_{k=1}^N P_k(nT) \cdot E(nT) \quad (\text{Eq. 22})$$

$$K_{k \text{ new}} = K_{k \text{ old}} - C_{k \text{ new}} \Delta \quad (\text{Eq. 23})$$

Using these equations iteratively, the H filter coefficients are continually updated to a point where
 5 the C_k 's are negligible compared to some values δ_k 's:

$$C_k \leq \delta_k \quad (\text{Eq. 24})$$

The values for δ_k are dependent upon noise and other factors and are predetermined empirically. When the criteria of Equation 24 is met, the equalizer has
 10 completed its task and can be reassigned to another line.

The values obtained for the K_k 's are loaded into the H filter. Where $Y_x = 0$, the F filter transfer characteristic is simply $1/2$, and there is an absolute delay corresponding to the round trip sampling delays
 15 through the digital filters, Z^{-L} in sampled data notation.

Where $Y_x \neq 0$, the equalization process yields;

$$\frac{N(Z^{-1})}{D(Z^{-1})} K_O = Y_L(Z) + Y_x(Z) \quad (\text{Eq. 25})$$

To obtain the required value for gH , the known quantity Y_x , multiplied by two must be subtracted:

$$20 \quad gH(Z) = \frac{N(Z^{-1})}{D(Z^{-1})} K_O - 2Y_x(Z) = Y_L(Z) - Y_x(Z) \quad (\text{Eq. 26})$$

In this case, the number of partial functions P_k must be increased to accommodate third order polynomials for

$N(Z^{-1})$ and $D(Z^{-1})$ so as to accommodate a first order polynomial type admittance for $Y_x(Z)$. The F filter 272, under these conditions, becomes:

$$F = \frac{gH}{2Y_L} = \frac{Y_L - Y_x}{2Y_L} \quad (\text{Eq. 27})$$

5 This value for the F filter 272 may be computed from the equalization results obtained from determining gH and the known value of Y_x . Alternatively, the equalizer may be used to determine the value of F directly. This operation is accomplished
10 as follows:

The coefficients for the H filter 116 are loaded, the switch 106 in the feedback path 112 of Figure 2 is closed by a switch control signal from MULDIS 157; the F filter 272 path switch 102 is opened, and the equalizer
15 130 process is run. This sequence of operation will yield:

$$\frac{N(Z^{-1})}{D(Z^{-1})} K_o = \frac{2Y_L}{Y_L - Y_x} \quad (\text{Eq. 28})$$

and thus:

$$F = \frac{D(Z^{-1})}{N(Z^{-1})} \frac{1}{K_o} \quad (\text{Eq. 29})$$

20 This provides coefficients for a recursive filter of the same type as the H filter 116, having third order polynomials for $N(Z^{-1})$ and $D(Z^{-1})$, for a Y_x of

the first order polynomial type. Actual circuit implementation of the H and F filters can be accomplished using state-of-the-art techniques.

Figure 6 shows generalized H and F filters 116 and 272 for processing $N(Z^{-1})$ and $D(Z^{-1})$ polynomials having orders of $k/2$ according to equation 30.

$$\frac{Y(Z)}{X(Z)} = \frac{K_0 [1 + K_1 Z^{-1} + K_2 Z^{-2} \dots K_{k/2} Z^{-k/2}]}{[1 + K_{\frac{k}{2}+1} Z^{-1} \dots K_k Z^{-k/2}]} \quad (\text{Eq. 30})$$

Coefficients and data are stored in semiconductor RAM memory organized into "push down" stacks 300 and 302 and recirculating stacks 304 and 306 to facilitate fetching and storing of information. At each sample time T , data outputs of RAM memory stacks 300-306 are fed to a multiplier/summer circuit 308 which computes the required output Y_n on line 310 by sequentially multiplying and accumulating the results according to Equation 31.

$$Y_n = K_0 \left[X_n + K_1 X_{n-1} + K_2 X_{n-2} \dots K_{\frac{k}{2}} X_{n-\frac{k}{2}} \right] - \left[K_{\frac{k}{2}+1} Y_{n-1} + K_{\frac{k}{2}+2} Y_{n-2} \dots K_k Y_{n-\frac{k}{2}} \right] \quad (\text{Eq. 31})$$

The first term computed is $K_0 X_n$ with the switches S_1 at 312 and S_2 at 314 in position 1. S_1 is then placed in position 3 and the x terms are computed. After that operation, S_1 and S_2 are placed in position 2 and the y terms are computed. Thus $k+1$ multiply/add operations are involved. This can be accomplished easily within the sampling period, sharing the same

memory and multiplier 308 and accumulator 316 for both the H and F filters. Thus for H and F filters having (six coefficients) $K=6$, fourteen (14) multiply/add operations are involved, allowing approximately one (1) 5 microsecond for each such operation, and a sampling period T of 14 microseconds can be accommodated. Higher order polynomials may be accommodated by adding parallelism in the arithmetic and memory operations. Other forms of recursive filters are possible, and the 10 filter structure of Figure 6 is exemplary only.

Various implementations of the equalizer structure can be realized. A preferred embodiment uses semiconductor RAM's, arithmetic unit and control logic to form a digital signal processing structure to implement 15 the principles of the equations. Furthermore, although the mean square error (M.S.E.) algorithm has been shown as the basis for the equalizer operation, other algorithms may be used to determine the coefficients of the H and F filters. For instance, the algorithm described 20 is based upon determining values of C_k over a period of a number of samples equal to NT. Each time the C_k 's are computed, the coefficients K_k 's are updated every N samples, thus

$$C_k(mNT) = \text{slice} \sum_{n=(m-1)NT+1}^{mNT} F_k(nT)E(nT) \quad (\text{Eq. 32})$$

25 and the new value of the coefficients are computed based on the gradient vector components C_k of Equation 33:

$$K_k(m) = K_k(m-1) - C_k(m) \quad (\text{Eq. 33})$$

By simplifying the computation of C_k by approximating:

$$C_k(nT) = \text{sign } P_k(nT) \cdot \text{sign } E(nT) \quad (\text{Eq. 34})$$

the coefficients K_k may be updated at every sample time T , allowing for a more rapid convergence to the
5 final values and reducing the amount of hardware needed. This algorithm is a simplification which approximates minimum means square error performance in a recursive filter and is advantageous, particularly
10 where emphasis is on fast convergence and reduced hardware.

The digital filters used to develop the partial outputs P_k may be implemented in like manner as for the F and H filters.

Figure 9 is a block diagram of a specific embodiment of the equalizer 159. A time shared arithmetic
15 logic unit (ALU) 500 performs sequential arithmetic and logic operations on information stored in the equalizer memory 502 under the control of control logic signals from control logic 504 and which signals
20 are synchronized with the control logic clock 506. The clock 506 frequency is, in turn, synchronized with the data sampling clock signal, and is a multiple thereof. First and second RAM memories 508 and 510,
which together comprise the equalizer memory 502, are
25 controlled by a series of control words, each of which also controls the operation of the arithmetic logic unit 500.

External control words on line 512 are shifted by shift register 514 to control gates 514A to provide
30 control words on line 516, under the control of logic 504. Constants may be loaded into memory 502 under external control, the memory contents may be examined

externally and the equalization process may be initiated externally. A logic signal output EC is provided when equalization is complete.

Figure 10 illustrates the organization of the first memory 508, which comprises four push-down stacks 520, 522, 524 and 526, of which the functional details of stack 520 are shown. Stacks 522, 524 and 526 function in like manner, as does memory stack 520. Each push-down memory stack 520 - 526 functions so that each new input word on memory bus 528 and line 530 replaces the last previous word in the stack and the last word in the stack is outputted from the stack. This is a first in - last out (FILO) operation. Each data location in each stack can be directly accessed or read. Register stack 520 stores words $A(n)$, $A(n-1)$... stack 522 stores $P_0(n)$, $P_0(n-1)$..., stack 524 stores $P_1(n)$, $P_1(n-1)$..., stack 526 stores $P_3(n)$, $P_3(n-1)$ Decoder 532 decodes the control words on line 516, from which the individual control signals to each stack. The output of each read operation is stored in memory register 534, and which output is an input to the Arithmetic Logic Unit 500.

Figure 11 illustrates the organization of the second memory 510, which includes memory sections 540, 542, 544 and 546. Memory section 540 stores constants K_k , including K_0 to K_4 . Memory section 542 stores data words SK , including S_0 to S_4 . Memory section 544 stores data words $R(n)$ from reference generator 161, which are shifted in via shift register 548. Memory section 546 stores $E(n)$. All inputs to all memory sections may go separately into memory or be read out by appropriate control words which are decoded by decoder 550. Any addressed data word in any of memory sections 540 - 546 causes the addressed word to be coupled to memory register 552 via gating

logic 554. Memory section 544 is organized as a push-down memory stack wherein any stored data word $R(n)$, $R(n-1) \dots R(1)$ can be directly addressed. The Arithmetic Logic Unit 500 has, as inputs thereto, the outputs of registers 534 and 552 from memories 508 and 510, respectively. The ALU performs its arithmetic processing on these inputs, after which processing, the results are placed in its accumulator 556. These results are then coupled out for storage in memory by control logic 504.

The actual arithmetic and logic operations performed by ALU 500 will now be summarized.

	<u>Function</u>	<u>Operation</u>
	Multiply	$C(R1) \cdot C(R2) \rightarrow ACC$
15	Multiply/Add	$C(R1) \cdot C(R2) + C(ACC) \rightarrow ACC$
	Subt/Add	$C(\pm R1) \pm C(\pm R2) + C(ACC) \rightarrow ACC$
	Increment	$C(M2) + LSB \rightarrow M2$
	Decrement	$C(M2) - LSB \rightarrow M2$
20	Complement ACC, if Neg	If sign ACC is — $2^N - C(ACC) \rightarrow ACC$
	Zero ACC	"0" Value $\rightarrow ACC$
	Initialize	Set up to begin equalization process

To implement the multiply function, the contents of memory registers 534 and 552 are multiplied and stored in accumulator 556.

To implement the multiply/add function, the contents of memory registers 534 and 552 are multiplied, and the result is added to the contents of accumulator 556.

To implement the subtract/add function, the contents of either or both registers 534 and 552, with appropriate sign changes under control of a field code, are added to the contents of accumulator 556.

54

To implement the increment operation or the decrement operation, the contents of the particular memory location is incremented or decremented if the sign of the accumulator at 560 is respectively negative or positive, also according to the field code.

To implement the "complement accumulator if negative" function, the sign of the contents of the accumulator is changed to positive when it is negative.

To implement the "zero accumulate" function, the numeric value zero is stored in the accumulator.

To implement the initialize function, which allows for external loading of constants if an external write control signal is present, the accumulator is cleared and accumulator flags at 562 are reset.

The control word structure from control logic 504, which is inputted via line 514 to memories 534 and 552 consists, for example, of a 6-bit memory M1 field, a 7-bit memory M-2 field and a 5-bit ALU 500 field. Each control word is then 18-bits. The foregoing is shown below.

Memory M1 Field	M1 1	Push 2	Address 1	Select 2	Subtotal 6-bits
Memory M2 Field	M2 1	Read/Write 1	Address 3	Select 2	Subtotal 7-bits
ALU 500 Field	Operation 5				Subtotal 5-bits
Control Word	Memory M1 Instruc- tion Field 6	Memory M2 Instruc- tion Field 7	ALU Instruc- tion Field 5	Total 18-bits	

Referring now to Figure 12, a flow chart of the equalizer operation is illustrated. The sequence is as follows:

Step 1 Initialization

From an external start signal on line 570, an Equalization Complete (EC Flag) signal on line 572 is reset and memory locations, accumulator 556 and
5 related registers are cleared. If an External Write signal is present on line 574, the control logic 504 allows the initial K_k 's and S_k 's to be loaded externally via shift register 576. When no External Write signal is present, internally stored values of K_k 's and S_k 's
10 are supplied via control logic 504.

Step 2 Compute $P_o(n)$

The values $A(n)$, $R(n)$ are obtained externally at the sampling time nT . $R(n)$ and $A(n)$ are pushed into their respective memories 520 and 548, respectively,
15 $A(n-2)$ and K_2 are fetched into memory 1 register 534 and memory 2 register 552, respectively. Their product is computed by the ALU 500 and held in the accumulator 556. Similarly, $A(n-1) \cdot K_1$ is then computed and added to the contents of the accumulator 556. Similarly,
20 $A(n)$ is then added to the contents of the accumulator. The product terms associated with the feedback P_o terms are subtracted from the accumulator according to the equation:

$$\text{new } P_o(n) = A(n) + K_1 A(n-1) + K_2 A(n-2) - K_3 P_o(n) - K_4 P_o(n-1)$$

25 (Eq. 35)

This operation corresponds to the $\frac{N(z^{-1})}{D(z^{-1})}$ function

shown in Figure 5. The result $P_o(n)$ is pushed into the P_o memory 522.

Step 3 Compute $P_3(n)$

The $P_3(n)$ term corresponding to the output of the $\frac{-K_0 z^{-1}}{D(z^{-1})}$ filter illustrated in Figure 5 is computed in a similar manner according to:

$$\text{new } P_3(n) = -K_0 P_0(n-1) - P_3(n) K_3 - P_3(n-1) K_4 \quad (\text{Eq. 36})$$

5 the result is pushed on the P_3 memory stack 526.

Step 4 Compute $P_1(n)$

Similarly, $P_1(n)$ is computed and pushed on the P_1 memory stack according to:

$$\text{new } P_1(n) = K_0 A(n-1) - P_1(n) \cdot K_3 - P_1(n-1) \cdot K_4 \quad (\text{Eq. 37})$$

10 which corresponds to the filter function of $\frac{K_0 z^{-1}}{D(z^{-1})}$ illustrated in Figure 5.

Step 5 Compute $E(n)$

The error term is computed according to:

$$E(n) = P_0(n) K_0 + R(n-1) \quad (\text{Eq. 38})$$

15 $E(n)$ is saved, i.e. it is stored in its prescribed location. This operation corresponds to the summing node function illustrated in Figure 5.

Steps 6 & 7 Compute C_k's, Updating K_k's

The C_k's are computed based on the simplified approach described previously. The operation involves incrementing or decrementing K_k based on the sign of the gradient vector term C_k, i.e.,

$$C_k(n) = P_k(n) E(n) \begin{array}{l} \text{increment } K_k \text{ if sign negative} \\ \text{decrement } K_k \text{ if sign positive} \end{array} \quad (\text{Eq. 39})$$

Steps 8, 9 and 10 Testing for Equalization Complete

The absolute value of C_k is computed by changing the sign of C_k if it is negative. The corresponding value of δ_k is subtracted from the absolute value of C_k. If the result in the accumulator is positive, the Test Flag is set = 1. If it is negative, the Test Flag Flip-Flop remains unchanged, i.e.,

$$\begin{array}{l} |C_k| - \delta_k = + \text{ SET TEST FLAG (TF)} \\ = - \text{ NO CHANGE TO TF} \end{array}$$

This step is performed for each gradient vector term C_k, i.e., C₀ through C₄. At the end of this procedure, the Test Flag is examined. If the Test Flag is = 0, indicating that no C_k term exceeded the corresponding value of δ_k, then equalization is complete. However, if the Test Flag = 1, indicating that one or more values of |C_k| exceeded its corresponding value of δ_k, then equalization is not complete and the cycle must be repeated. The control logic 504 will return to Step 2 to await the next sampling signal at the interval of T units. When Test Flag = 0, the Equalization Complete (EC) Signal is outputted for external recognition, and the procedure terminates allowing the external system to read the values of K₀ through K₄ from the memory for external use. The Equalizer may also be reassigned at

this point to some other line.

From the viewpoint of execution times, the Equalizer must complete Steps 2 to 9 in a time less than or equal to the sampling time T. For a sampling time T corresponding to present state-of-the-art sampling of speech telephony signals, 125 microseconds is adequate, corresponding to a sampling frequency of 4 Kh_z.

Based on a maximum of 50 control word executions corresponding to Step 2 to Step 9, each control word must be executed in approximately two microseconds. Thus, for a worst case control word consisting of fetching two words from memory 508 and memory 510, multiplying them together and adding them to the ACC, the performance requirements can be assigned as follows:

Memory Access	.5 microseconds
Multiply	1.0 microseconds
Add	<u>.5</u> microseconds
Total	2.0 microseconds

The significance of these requirements are parallel register-to-register transfers and arithmetic operations. For 13-bit arithmetic, which satisfies the telephony requirements, these performance requirements are attainable in today's LSI technology, using the Equalizer structure described herein.

Present state-of-the-art off-the-shelf general purpose microcomputers of the 8- to 16-bit variety would not be able to meet the above performance requirements using standard programming techniques. The novel features heretofor described of the herein disclosed equalizer structure enable it to attain these performance requirements.

A brief recapitulation of said novel equalizer

features includes, inter alia;

- (1) Multiple memories which can be addressed simultaneously;
- (2) Special memory organization facilitating the operations required (push-down stacks which are directly addressable);
- (3) Parallel multiply/add arithmetic capability;
- (4) Microcoded control words which simultaneously control the memories and the arithmetic logic unit;
- (5) Control words directly related to the specific operation required, e.g., MULTIPLY/ADD, COMPLEMENT ACC IF NEG, INCREMENT, DECREMENT.

An alternative embodiment of the equalizer hereinbefore described is implementable utilizing a general purpose type signal processor having special arithmetic and memory processing capabilities. A simplified block diagram of such general purpose processor is illustrated by Figure 7.

The circuits described are all implementable in digital LSI. By the addition of the feedback and feed-forward loops and the F and H filters and by replacing the standard decoder output amplifier with the transconductance amplifier, a complete line circuit is possible on a single LSI chip. The F and H filters are simple recursive filters, which make it possible to include the F and H filters on a codec and/or codec and filter chip. Thus, the present invention eliminates the prior art analog 2/4 wire hybrid and discrete terminating and balancing networks replacing them with the programmable digital LSI circuits described. This results in lower costs of manufacture, installation and maintenance, in addition to improved performance.

Although non-recursive filters can be used for the H and F filters, their costs would exceed those of the recursive filters described. Similarly, a non-recursive equalizer, based on known techniques, could have been

used for either non-recursive or recursive F and H filters, but would be inferior as aforescribed. Conversion of the non-recursive filter structures provided by a non-recursive equalizer could be used
5 to convert to a recursive structure of the type described utilizing the Fletcher-Powell algorithm as described in Deczky, Synthesis of Recursive Digital Filters, IEE Trans. Audio Electro Acoust. Vol. AU-20, pp. 257-263, Oct. 1972. Again, such technique requires
10 substantially greater hardware than the all-recursive structure of the present invention.

Memory 330 of Figure 7 includes allocated portions at 336 for storing C_k coefficients, at 338 for storing K_k coefficients, at 340 for storing correlator sums,
15 at 342 for storing values of D_k and δ_k , at 344 for storing intermediate results $A(n)$, $P_k(n)$, $\dots$, $P_{k(n-\frac{k}{2})}$, and at 346 for storing a control program for accessing the stored data in accordance with addresses provided by control logic 332. Computation is accomplished via
20 arithmetic unit 334.

Figure 8 illustrates a generalized digital two-to-four wire converter for full duplex signals on line 400. The coder has a unity gain and includes analog-to-digital prefilter 402, analog-to-digital converter 403 and
25 analog-to-digital postfilter 404. Operationally, the circuit of Figure 8 functions in like manner to that of Figure 3B in that line impedance matching occurs similarly. The decoder loop, including digital-to-analog converter 406 and associated prefilter 408 and
30 postfilter 410 converts digital transmit signals on line 412 to analog signals on line 400. Transconductance amplifier 414 provides an infinite output admittance. Automatic equalizer system 130 provides updated filter coefficients and control to H filter 416 and F filter
35 418, with unwanted signal return in the receive signal

eliminated at summing node 420. Coder feedback and the transmit signal are combined at summing node 421. Both coder feedback and F filter 418 inclusion in the circuit are determined by switches 422 and 424, respectively, under equalizer 130 control.

The multiplexer/distributor (MULDIS) 157 described generally with reference to Figure 2 is illustrated in greater detail by Figure 13, the MULDIS 157 enables the equalizer to be shared by a plurality of lines 1 through N. In essence, the MULDIS 157 makes a determination of which line is to be connected to the equalizer. The MULDIS 157 multiplexes the A(n) signals from the plurality of line circuits and distributes the filter coefficients, the switch and filter mode control signals, and the reference generator 161 output to the line circuit selected under control of a maintenance system.

Signals from the digital reference generator 161, R(n) are coupled to lines 1 to N via logic gating 600, which ANDs R(n) with ADD 1...ADD N from the maintenance system at logic 602. The appropriate address is shifted through shift register 604, decoded by decoder 606 and coupled to logic 600 as illustrated.

The clock and switch control/mode control signals from the maintenance system are coupled to lines 1 through N through logic gating 608. These signals are coupled to flip-flops 610 and 612, the outputs of which flip-flops are AND'ed with the appropriate ADD 1 through ADD N signal from decoder 606.

The A(n) words from lines 1 through N are selected at logic gating 614 and coupled to the equalizer as the output of OR gate 616. Each A(n) input from each of lines 1 through N is AND'ed with signals ADD 1 through ADD N from decoder 606 at the AND gates in logic 614.

The filter coefficients to the F and H filters for each of lines 1 through N, such as F filter 272

and H filter 116 of line circuit 1 of N are coupled from the equalizer to the appropriate line 1 of N via logic gating 618. The filter coefficients are AND'ed at logic 618 with the ADD 1 through ADD N signals from decoder 606 to select the correct line.

Referring now to Figure 14, a block diagram of the major interface connections between the equalizer, the maintenance system, the MULDIS and the line circuits. The maintenance system 650 may comprise a conventional data source, such as a computer and its associated memory. The illustrated data and control signals, described hereinelsewhere, provided the requisite timing and handshake between the equalizer, the maintenance system and the line circuits via the MULDIS 157.



CLAIMS:

1. A circuit for automatically digitally synthesizing an output matching line terminating impedance including:

- 5 (a) analog to digital conversion means for converting analog communication signals occurring at circuit output terminals to digital signals;
- (b) means for summing said digital signals with input digital communication signals providing a composite digital signal input having negative feedback;
- 10 (c) digital filter means having said composite digital signal input coupled thereto for directly establishing a predetermined output impedance characteristic, said digital filter having alterable filter coefficients, and having a digital signal output;
- 15 (d) digital to analog conversion means for converting said digital signal output to an analog voltage signal; and
- (e) means for converting said analog voltage signal to an analog current having a high output impedance characteristic compared to the matched line and to
20 the synthesized impedance such that said analog current is coupled to said circuit output terminals and to said analog to digital conversion means.

2. A circuit as claimed in Claim 1, wherein said
25 means for converting said analog voltage signal to an analog current includes transconductance amplifier means having an output current defined by a specific current to voltage transfer ratio g mhos.

3. A circuit as claimed in Claim 1, wherein said
30 predetermined output impedance characteristic of said circuit is derived by varying said alterable filter coefficients in accordance with a control data input.

4. A circuit as claimed in Claim 3, wherein the
35 digital filter means is a recursive filter, the circuit including automatic equalizer means for directly



- 34 -

deriving recursive digital filter coefficients based on the minimization of error between the equalizer output and a reference.

5. A circuit as claimed in Claim 3, wherein
5 said digital filter is a recursive digital filter.

6. A circuit as claimed in Claim 5, wherein said recursive digital filter includes five variable filter coefficients.

7. A circuit as claimed in Claim 1, for a line
10 which includes incidental shunting impedances between it and said circuit, and wherein the admittance of said line is represented by Y_L , the admittance of said shunting impedances is represented by Y_x and wherein said predetermined output impedance characteristic of
15 said digital filter is equal to $Y_L - Y_x$.

8. A circuit as claimed in Claim 3, including analog-to-digital prefilter means for limiting the input bandwidth of said analog communication signals to a predetermined frequency range; and analog-to-digital
20 postfilter means for establishing a desired feedback gain characteristic.

9. A circuit as claimed in Claim 3 or 8 including digital-to-analog prefilter means for establishing a desired digital-to-analog path gain characteristic; and
25 digital-to-analog postfilter means for filtering the output of said digital-to-analog conversion means to reduce quantizing effects.

10. A circuit for digitally synthesizing a line matching impedance, the circuit including analog-to-
30 digital conversion means for converting negative feedback input analog signals to negative feedback digital signals, means for summing said negative feedback digital signals with additional digital signals to derive a combined signal output, recursive digital filter means



- 35 -

having said combined signal output coupled thereto,
and having alterable filter coefficients, means for
programmably altering said filter coefficients such
that the processing of said combined signal output
5 by said digital filter establishes a controllable
line matching impedance characteristic of said circuit
in dependance upon said digital filter transfer
characteristics, digital-to-analog conversion means
for converting the output of said digital filter to an
10 analog voltage signal, and transconductance amplifier
means for converting said analog voltage signal to an
analog current having a high output impedance
characteristic relative to the impedance at the output
of said digital-to-analog conversion means, said
15 analog current also being coupled to said analog-to-
digital conversion means.

11. A circuit as claimed in Claim 10, and
including control logic means for deriving a coefficient
control data signal, said data signal being coupled to
20 said digital filter to vary said filter coefficients in
accordance with said control signal, said control logic
including means for generating a reference such that
said digital filter coefficients are varied to minimize
the error between the digital filter output and said
25 reference.

12. A circuit as claimed in Claim 11, wherein
said control logic includes means for deriving a digital
representation of a line impedance, and means for loading
said digital representation into memory for storage.

30 13. A circuit as claimed in Claim 12, including
means for initializing the digital filter coefficients
with said stored digital representation, such that fast



68

- 36 -

equalization is provided by holding the previous digital filter tap coefficients as the starting coefficients for a subsequent equalization operation.

5 14. A circuit as claimed in Claim 11, configured on an LSI chip.

15. A circuit for automatically digitally synthesizing an output matching line terminating impedance, substantially as described with reference to the accompanying drawings.
