(54) Title: ADAPTATIVE FEEDFORWARD AND FEEDBACK CONTROL SYSTEM
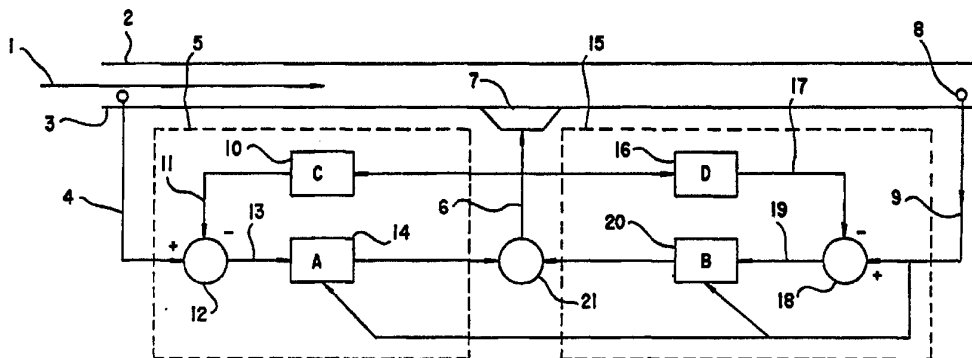
(57) Abstract

An active control system for multiple interacting channels to control a constant noise or vibration consisting of first (3) and second (8) sensor means adapted to provide reference and residual signals respectively, a first and second filter means and first (12) and second (18) subtraction means to provide first and second output control signals and means for combining said output signals.

ADAPTIVE FEEDFORWARD AND FEEDBACK CONTROL SYSTEM

Background

Active control of disturbances, such as sound, vibration or disturbances in signals
is well known. A recent review of the field is contained in 'Active Sound Control' by
P.A. Nelson and S.J. Elliot, Academic Press, 1991. Such systems use an actuator to
generate a control disturbance which is out of phase with the original disturbance and so
tends to cancel it. This technique is first described by Lueg in U.S. Patent No.
2,043,416. Most active control systems use adaptive filtering techniques, in which the
controller characteristic is adjusted according to an algorithm such as the 'filtered-x LMS
algorithm' such as disclosed by D.R. Morgan, IEEE Transactions on Acoustics, Speech
and Signal Processing, Volume ASSF 28, Number 4,1980, and by Widrow and Stearns,
'Adaptive Signal Processing', Prentice Hall, 1985. Two widely used techniques are
feedforward control, as described in Chaplin U.S. Patent No. 4,122,303, and feedback
control as described in Ziegler U.S. Patent No. 4,878,188.

In an active control system, the reference sensor is usually sensitive to the control
disturbance. This provides a feedback mechanism which can cause the system to become
unstable. One known method for compensating for this is to estimate the feedback
component and to subtract it from the sensor signal. Both Chaplin and Ziegler use this
compensation technique.

The adaptive feedforward controller disclosed in Chaplin is shown in Figure 1. In
this configuration the control system is used for canceling noise (1) propagating down a
pipe or duct (2). An upstream (relative to the direction of sound propagation) or
reference sensor (3) provides a reference signal (4) related to the sound at the sensor
position. This signal is input to the control system (5) which in turn generates a control
signal (6). The control signal is supplied to actuator (7) which in turn produces sound to
cancel the original noise. An error or residual sensor (8), downstream of the actuator,
produces a residual signal (9) related to the residual sound at that position. This signal is
used to adjust the characteristic of the control system (5). The control system comprises
a compensation filter (10) which acts on the control signal (6) to produce a compensation
signal (11) which is an estimate of the component of signal (4) due to the actuator.
Hence the characteristic of the filter should correspond to the impulse response of the
physical system from controller output to controller input (including the response of the
actuator (7), the sensor (3) and, for digital systems, any anti-aliasing filter or anti-imaging
filter). The compensation signal (11) is subtracted at (12) from the reference signal (4)
to produce an input signal (13). The input signal is then passed through a cancellation
filter (14) to produce the control signal (6). The filtered-x LMS algorithm is commonly

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used to adjust the characteristic of the cancellation filter (14). The characteristic of compensation filter (10) can be determined by known system identification techniques.

The adaptive feedback controller disclosed by Ziegler is shown in Figure 2. In this configuration the control system is used for canceling noise (1) propagating down a

5   pipe or duct (2). A sensor (8), downstream of the actuator (relative to the direction of sound propagation), provides a signal (9) related to the sound at the sensor position. This signal is input to the control system (15) which in turn generates a control signal (6). The control signal is supplied to actuator (7) which in turn produces sound to cancel the original noise. The same sensor (8) acts as a residual sensor since the signal (9) is related

10   to the residual sound at that position. This signal is used to adjust the characteristic of the control system (15). The control system comprises a compensation filter (16) which acts on the control signal (6) to produce a compensation signal (17) which is an estimate of the component of signal (9) due to the actuator. Hence the characteristic of the filter should correspond to the impulse response of the physical system from controller output

15   to controller input (including the response of the actuator (7), the sensor (8) and, for digital systems, any anti-aliasing filter or anti-imaging filter). The compensation signal (17) is subtracted at (18) from the residual signal (9) to produce an input signal (19). The input signal is then passed through a cancellation filter (20) to produce the control signal (6). The filtered-x LMS algorithm is commonly used to adjust the characteristic of

20   the cancellation filter (20).

The performance of a feedforward control system is limited by noise at the reference sensor which is uncorrelated with the disturbance. This is called the 'coherence limit'. The performance of a feedback control system is limited by the delay in the control loop, which limits performance to narrow-band or low frequency disturbances.

25   Hence for disturbances which are a mixture of broadband and narrow band noise there is an advantage to be gained by using a combination of feedforward and feedback control.

This has been recognized by N.J. Doelman, 'A Unified Strategy for the Active Reduction of Sound and Vibration', Journal of Intelligent Materials Systems and Structures, Volume 2, Number 4 October 1991, pp. 558-580. This system is shown in

30   Figure 3 (also Doelman's Figure 3). The outputs of a feedforward filter (5) and a feedback filter (15) are combined at (21) to produce the control signal (6). Doelman uses recursive filters and derives the optimal filter characteristics for stationary noise signals. However, there is no interaction between the two filters (5) and (15) in his arrangement. This can have serious implications since there is no guarantee that the

35   filters he derives are stable. For an 'off-line' design process the stability of the filters (both in open-loop and in closed loop) can be checked before the filter is implemented, but for adaptive control systems it is not practical to continually check for system

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stability. The risk of instability in the system would make this system unsuitable for practical implementation.

There is, therefore, a need for an adaptive control system which can be adapted easily without the risk of instability.

Summary of the Invention

The current invention relates to a combined feedback and feedforward system for controlling disturbances. The system uses compensation filters to ensure the closed loop stability of the system and provides a computationally efficient way for adapting such a system while maintaining stability.

An object of the invention is to provide a system which can be adapted without any instability.

This and other objects of the invention will become apparent when reference is had to the accompanying drawings in which:

List of Figures

Figure 1 is a diagrammatic view of a known adaptive feedforward control system.

Figure 2 is a diagrammatic view of a known adaptive feedback control system.

Figure 3 is a diagrammatic view of a known combined feedforward and feedback control system.

Figure 4 is a diagrammatic view of a combined feedforward and feedback control system of the invention.

Figure 5 is a diagrammatic view of another embodiment of a combined feedforward and feedback control system of the invention.

Figure 6 is a diagrammatic view of the application of the current invention to a muffler noise control system.

Detailed Description of the Invention

The invention relates to a system for controlling a vibration or noise disturbance. For example, the disturbance may be sound propagating down a pipe duct, or propagating in an open region, or it may be vibration propagating through a structure. The system is a combined feedforward and feedback control system which utilizes compensation filters to ensure stability of the system.

A reference sensor is used to provide a reference signal (uf) related at least in part to the disturbance to be controlled and a residual sensor is used to provide a residual signal (ub) related to the controlled disturbance. A reference compensation signal (Cy) is subtracted from the reference signal to produce a feedforward input signal (xf). The feedforward input signal is filtered by a feedforward cancellation filter (A) to produce a Feedforward output signal (yf). A residual compensation signal (Dy) is subtracted from

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the residual signal to produce a feedback input signal (xb). The feedback input signal is filtered by a feedback cancellation filter (B) to produce a feedback output signal (yb).

The feedforward and feedback output signals are then combined to produce a control signal (y) which is sent to an actuator. The actuator produces a control

5   disturbance which modifies the original disturbance. Usually, but not always, the intention is that the residual disturbance is smaller than the original disturbance.

In the general implementation the cancellation filters are recursive filters, in the simplest implementation they are Finite Impulse Response(FIR) filters. In this case the operation at the n$^{\text{th}}$ time step is described by the equations

10      $$xf(n) = uf(n) - Cy(n) \tag{1}$$

$$xb(n) = ub(n) - Dy(n) \tag{2}$$

$$yf(n) = \sum_{m=0}^{nA-1} A(m).xf(n-m) \tag{3}$$

$$yb(n) = \sum_{m=0}^{nB-1} B(m).xb(n-m) \tag{4}$$

$$y(n) = yf(n) + yb(n) \tag{5}$$

15   where nA is the number of coefficients in the feedforward cancellation filter and nB is the number of coefficients in the feedback cancellation filter. The reference compensation signal is derived from the combined output using

$$Cy(n+1) = \sum_{m=0}^{nC-1} C(m).y(n-m) \tag{6}$$

where the filter C is the reference compensation filter which models the physical feedback

20   from the controller output to the controller reference input, including the response of the actuator, the sensor and any filters. nC is the number of coefficients in this filter. This is in contrast to the scheme of Doelman in which the combined output is not used in the filters.

The residual compensation signal can be derived in one of two methods. Firstly,

25   it can be derived from the combined output using

$$Dy(n+1) = \sum_{m=0}^{nD-1} D(m).y(n-m) \tag{7}$$

where the filter D is the residual compensation filter which models the physical feedback from the controller output to the controller residual input, including the response of the actuator, the sensor and any filters. nD is the number of coefficients in this filter.

30   Alternatively, the residual compensation signal can be derived from the output of the feedback cancellation filter, so that

5

$$Dy(n+1) = \sum_{m=0}^{nD-1} D(m).yb(n-m) \qquad (8)$$

The characteristics of the filters C and D (which may be recursive filters or FIR filters) can be found by standard system identification techniques or by on-line system identification. In the latter case a low level test signal is added to the output control signal and the difference between the actual response and the predicted response is used to adjust the filter characteristics. The LMS algorithm, for example, can be used for this adaption.

The feedback cancellation filter B can be adapted by the filtered-x input algorithm for example. This is the simplest algorithm but many alternative adaption algorithms have been disclosed. The coefficients are updated using

$$Dxb(n) = \sum_{m=0}^{nD-1} D(m).xb(n-m) \qquad (9)$$

$$B_n(m) = (1-\mu_B\lambda_B).B_{n-1}(m) - \mu_B.rb(n).Dxb(n-m), \quad m=0,nB-1 \qquad (10)$$

where μ_B is the adaption step size and λ_B is a leakage parameter. The feedforward filter may also be adapted using the filtered-x LMS algorithm. The filtered-input signal is given by

$$Dxf(n) = \sum_{m=0}^{nD-1} D(m).xf(n-m) \qquad (11)$$

The feedforward cancellation coefficients can be updated using the residual signal, rb, according to

$$A_n(m) = (1-\mu_A\lambda_A).A_{n-1}(m) - \mu_A.rb(n).Dxf(n-m), \quad m=0,nA- \qquad (12)$$

where μ_A is the adaption step size and λ_A is a leakage parameter. This is depicted in Figure 4. Figure 4 is a combination of Figures 1 and 2, except the outputs from the feedforward filter (14) and the feedback filter (20) are combined at (21) to produce the output control signal (6), and the compensation signals (11) and (17) are obtained by filtering the combined output control signal (6) rather than the individual output signals. Both of the filters (14) and (20) are adjusted in response to the residual signal (9). In most adaption algorithms, such as the filtered-x LMS algorithm described above, the input to the cancellation filters is also used in the update calculation.

An alternative to equation (12) is to adapt the feedforward cancellation coefficients using the feedback input signal, xb, according to

$$A_n(m) = (1-\mu_A\lambda_A).A_{n-1}(m) - \mu_A.xb(n).Dxf(n-m), \quad m=0,nA- \qquad (13)$$

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This is depicted in Figure 5. Here the feedback compensation signal (17) is calculated from the output (22) from the feedback cancellation filter (20) rather than the combined output (6). Thus the feedback input signal represents the residual signal resulting from the effect of the feedforward control signal only - it is independent of the output from the
5   feedback controller.

The combined algorithm of this invention can be used for multi-channel systems. The extension of LMS style algorithms to multi-channel control systems is well known. For example, multi-channel feedforward control, using feedback compensation, is described in Nelson & Elliot, Chapter 12. Multi-channel feedback control using feedback
10   compensation is disclosed by Ziegler, 'Multiple Interacting DVE Algorithm', US Patent Application number 07/928,471 herein incorporated by reference. The extension of the current invention from the single channel described above to multiple reference inputs, multiple actuators and multiple residual sensors will be obvious to those skilled in the art.

The basic equations for a system implemented using FIR filters are
15

$$xf_i(n) = uf_i(n) - Cy_i(n), \quad i = 1...nI \tag{14}$$

$$xb_j(n) = ub_j(n) - Dy_j(n), \quad j = 1...nJ \tag{15}$$

$$yf_k(n) = \sum_{i=1}^{nI} \sum_{m=0}^{nA-1} A_{ki}(m).xf_i(n-m), \quad k = 1...nK \tag{16}$$

$$yb_k(n) = \sum_{j=1}^{nJ} \sum_{m=0}^{nB-1} B_{kj}(m).xb_j(n-m), \quad k = 1...nK \tag{17}$$

20   $$y_k(n) = yf_k(n) + yb_k(n), \quad k = 1...nK \tag{18}$$

where nI is the number of reference sensors, nJ is the number of residual sensors and nK is the number of actuators. A_{kj} represents the filter between the jth input and the kth output. Multi-channel versions of B, C and D are similarly defined.

25   The compensation signals are given by

$$Cy_i(n) = \sum_{k=1}^{nK} \sum_{m=0}^{nC-1} C_{ki}(m).y_k(n-m), \quad i = 1...nI \tag{19}$$

and either

$$Dy_j(n) = \sum_{k=1}^{nK} \sum_{m=0}^{nD-1} D_{kj}(m).y_k(n-m), \quad j = 1...nJ \tag{20}$$

or

30   $$Dy_j(n) = \sum_{k=1}^{nK} \sum_{m=0}^{nD-1} D_{kj}(m).yb_k(n-m), \quad j = 1...nJ \tag{21}.$$

The multi-channel LMS algorithm for updating these filters is described by Nelson and Elliot (Chapter 12).

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Example Algorithm

In one embodiment of the controller the filters are implemented as Finite Impulse Response (FIR) filters. The parameters are defined in the table below:

Parameter	Description
freq	sampling frequency
nA	number of coefficients in forward cancellation filter
nB	number of coefficients in backward cancellation filter
nC	number of coefficients in forward compensation filter
nD	number of coefficients in backward compensation filter
gf	forgetting factor for power estimate
gb	forgetting factor for power estimate
fmin	minimum power
bmin	minimum power
leak	leakage parameter
leakmin	minimum leakage
Astep	step size for forward LMS
Bstep	step size for backward LMS
Cstep	step size for LMS adaption of C filter
Dstep	step size for LMS adaption of D filter
grb	forgetting factor for residual power estimate
gl1	smoothing factor for leak adjustment
gl2	memory factor for leak adjustment
gp	forgetting factor for peak detect
level	set level for peak output
invlevel	reciprocal of *level*
gmin	minimum test signal level
testlevel	test signal level relative to residual level
invf	forward normalization factor, (calculated automatically)
invb	backward normalization factor, (calculated automatically)
gain	gain for test signal level, (calculated automatically)
Amu	normalized step size for A filter, (calculated automatically)
Bmu	normalized step size for B filter, (calculated automatically)

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The variables, that is the dynamic data in the processor, are defined in the table below.

Variable Name	Description	Size
A	FIR forward cancellation filter	nA
B	FIR backward cancellation filter	nB
C	FIR reference compensation filter	nC
D	FIR residual compensation filter	nD
uf	reference input signal	1
ub	residual input signal	1
$test$	identification test signal delay line	max(nC+1,nD+1)
$Ctest$	compensation for test signal	1
$Dtest$	compensation for test signal	1
rf	compensated reference signal	1
rb	compensated residual signal	1
Cy	reference compensation signal	1
Dy	residual compensation signal	1
yf	forward control signal	1
yb	backward control signal	1
y	control signal delay line	max(nC,nD)
$output$	output signal	1
xf	forward input signal delay line	max(nA,nD)
xb	backward input signal delay line	max(nA,nD)
Dxf	filtered forward input signal delay line	nA
Dxb	filtered backward input signal delay line	nB
pf	forward power estimate	1
pb	backward power estimate	1
prb	residual power estimate	1
$peak$	peak output level	1

An algorithm for adaptation of the filter coefficients is given below. This
describes the nth step of the algorithm and is repeated every sample time. This particular
example uses a Normalized Least Mean Square (NLMS) algorithm and includes on-line
system identification using a random test signal. The square brackets [...] denote
operations that may not be required, but are desirable. The braces {..} denote operations
that can be done at a reduced rate (i.e. not every sample) or as a background task so as
to reduce the processing load on the processor.

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$$read\ ADCs\ to\ get\ uf(n)\ and\ ub(n) \qquad (22)$$

$$[high\ pass\ filter\ uf\ and\ ub] \qquad (23)$$

Comment: Compensate for test signal

$$rf(n) = uf(n) - Ctest(n) \qquad (24)$$

$$5 \qquad rb(n) = ub(n) - Dtest(n) \qquad (25)$$

Comment: Compensate for output signal

$$xf(n) = rf(n) - Cy(n) \qquad (26)$$

$$xb(n) = rb(n) - Dy(n) \qquad (27)$$

Comment: Complete calculation of output

$$10 \qquad yf(n) = yf(n) + A(0).xf(n) \qquad (28)$$

$$yb(n) = yb(n) + B(0).xb(n) \qquad (29)$$

$$y(n) = yf(n) + yb(n) \qquad (30)$$

$$output(n) = y(n) + test(n) \qquad (31)$$

$$[high\ pass\ filter\ output] \qquad (32)$$

$$15 \qquad output\ to\ DAC \qquad (33)$$

Comment: Calculate mean modulus of inputs signals

$$pf_n = pf_{n-1} + gf.(|Dxf(n-1)| - pf_{n-1}) \qquad (34)$$

$$pb_n = pb_{n-1} + gb.(|Dxb(n-1)| - pb_{n-1}) \qquad (35)$$

$$\{\ invf = 1/(pf_n + fmin)\ \} \qquad (36)$$

$$20 \qquad \{Amu = Astep.invf.invf\} \qquad (37)$$

$$\{\ invb = 1/(pb_n + bmin)\ \} \qquad (38)$$

$$\{Bmu = Bstep.invb.invb\} \qquad (39)$$

Comment: Regulate peak output signal (calculate new leak)

$$peak_n = (1 - gp).peak_{n-1} \qquad (40)$$

$$25 \qquad if\ |y(n)| > peak_n\ then\ peak_n = |y(n)|\ \ end \qquad (41)$$

$$\left.\begin{array}{l} if\ peak_n > level\ \ then \\ \quad leak = leak + gl1.(1 + leak).(peak_n.invlevel - 1) \\ else \\ \quad leak = gl2.leak \\ end \end{array}\right\} \qquad (42)$$

$$\left.\begin{array}{l} if\ leak < leakmin \\ \quad leak = leakmin \\ endif \end{array}\right\} \qquad (43)$$

$$\{Ascale = 1 - leak.Amu\} \qquad (44)$$

$$\{Bscale = 1 - leak.Bmu\} \qquad (45)$$

$$30 \qquad Comment:\ Update\ filters$$

$$factor = Amu.rb(n) \qquad (46)$$

10

$$A_n(m) = Ascale.A_{n-1}(m) - factor.Dxf(n-m), \quad m = 0,nA - \tag{47}$$

$$factor = Bmu.rb(n) \tag{48}$$

$$B_n(m) = Bscale.B_{n-1}(m) - factor.Dxb(n-m), \quad m = 0,nB - \tag{49}$$

$$factor = -Cstep.rf(n) \tag{50}$$

$$C_n(m) = C_{n-1}(m) - factor.test(n-1-m), \quad m = 0,nC-1 \tag{51}$$

$$factor = -Dstep.rb(n) \tag{52}$$

$$D_n(m) = D_{n-1}(m) - factor.test(n-1-m), \quad m = 0,nD - \tag{53}$$

Comment: Calculate filtered inputs

$$Dxf(n) = \sum_{m=0}^{nD-1} D(m).xf(n-m) \tag{54}$$

$$Dxb(n) = \sum_{m=0}^{nD-1} D(m).xb(n-m) \tag{55}$$

Comment: Calculate compensation signals for next iteration

$$Ctest(n+1) = \sum_{m=0}^{nC-1} C(m).test(n-m) \tag{56}$$

$$Dtest(n+1) = \sum_{m=0}^{nD-1} D(m).test(n-m) \tag{57}$$

$$Cy(n+1) = \sum_{m=0}^{nC-1} C(m).y(n-m) \tag{58}$$

$$Dy(n+1) = \sum_{m=0}^{nD-1} D(m).y(n-m) \tag{59}$$

Comment: Calculate partial sums for next iteration

$$yf(n+1) = \sum_{m=1}^{nA-1} A(m).xf(n+1-m) \tag{60}$$

$$yb(n+1) = \sum_{m=1}^{nB-1} B(m).xb(n+1-m) \tag{61}$$

Comment: Calculate mean modulus of residual signal

$$prb_n = prb_{n-1} + grb.(|rb(n)| - prb_{n-1}) \tag{62}$$

Comment: Calculate test signal gain

$$\{ gain = gmin + testlevel.prb_n \} \tag{63}$$

$$get \ new \ test \ signal, \ random(n+1) \tag{64}$$

$$test(n+1) = random(n+1).gain \tag{65}$$

There are a great many applications for the known feedforward adaptive filter. Since all of these use both a reference sensor and a residual sensor, the feedforward controller can be replaced by a combined feedforward and feedback controller of the current invention. These applications are not necessarily restricted to the control of noise or vibration.

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One application area is for reducing noise propagated down ducts or pipes. Here the reference sensor is usually in the pipe upstream (relative to the sound propagation) of the actuator. The actuator is often one or more loudspeakers which can be placed in the pipe or adjacent to the end of the pipe. The main reason for placing the actuator adjacent

5 to the end of the pipe is to remove the actuator from the gases or liquids in the pipe - since these may be hot or corrosive and may be damaging to the actuator. A further advantage is that the feedback from the actuator to the upstream sensor is reduced and may sometimes be neglected. This can simplify the control system by removing the need for the reference compensation filter.

10 The control system has been successfully tested for canceling the noise from an automobile muffler. The general arrangement is shown in Figure 6. The exhaust gases and noise (1) propagate down the exhaust pipe (2) towards the open end. The upstream sensor (3) was a microphone, the actuators were loudspeakers in an enclosure (7) adjacent to the end of the muffler pipe. The residual sensor (8) was a microphone placed

15 adjacent to the end of the pipe. The control system used FIR filters and a sampling rate of 2KHz. The resulting noise reduction was approximately 10dB under transient driving conditions and 20dB during steady driving conditions. This was better than using a feedforward or feedback controller alone. Further details are described in a co-pending patent application.

20 Another application is in an active ear defender. Here the actuator is a loudspeaker adjacent to the ear or within the ear canal. The residual sensor is placed between the loudspeaker and the ear drum and the reference sensor is placed on the outside of the loudspeaker enclosure or at a nearby position. Adaptive feedforward control has been disclosed for use with ear defenders of this type. Combined feedforward and feedback

25 control provides improved performance.

Having described the invention it will be obvious to those of ordinary skill in the art that many changes and modifications can be made without departing from the scope of the appended claims.

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CLAIMS

1. A control system for controlling a continuing base disturbance, said system comprising

    a first sensor means for providing a reference signal related to said disturbance,

    a second sensor means for providing a residual signal also related to said disturbance,

    a first subtraction means for subtracting a first compensation signal from said reference signal to produce a first input signal,

    a first filter means responsive to said first input signal to produce a first output signal,

    a second subtraction means for subtracting a second compensation signal from said residual signal to produce a second input signal,

    a second filter means responsive to said second input signal to produce a second output signal,

    combining means for combining said first and second output signals to produce a control signal, and

    actuator means adapted to respond to said control signal and to produce controlling disturbance to thereby control said base sound or vibration disturbance by continually controlling it.

2. The system of claim 1 wherein said first compensation signal is derived by filtering said control signal.

3. The system of claim 1 wherein said second compensation signal is derived by filtering said control signal.

4. The system of claim 1 wherein said second compensation signal is derived by filtering said second output signal.

5. The system of claim 1 wherein said first filter means is an adaptive filter.

6. The system of claim 5 wherein a characteristic of said first filter means is adapted in response to said residual signal.

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7.   The system of claim 5 wherein a characteristic of said first filter means is adapted
        in response to said second input signal.

8.   The system of claim 1 wherein said second filter means is an adaptive filter.

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9.   The system of claim 8 wherein a characteristic of said second filter means is
        adapted in response to said residual signal.

10.   The system of claim 8 wherein the adaption is based on a Least Mean Square
10   algorithm.

11.   The system of claim 1 wherein said filter means are digital Finite Impulse
        Response filters.

15   12.   The system of claim 1 wherein said filter means are digital Recursive filters.

13.   The system of claim 1, and including means for on-line system identification.

14.   A control system with multiple interacting channels for controlling a continuing
20   disturbance, said system comprising
                first sensor means to provide reference signals related to said disturbance,
                second sensor means to provide residual signals related to said
        disturbance,
                first subtraction means for subtracting first compensation signals from said
25   reference signals to produce first input signals,
                first filter means responsive to said first input signals to produce first
        output signals,
                second subtraction means for subtracting second compensating signals
        from said residual signals to produce second input signals,
30   second filter means responsive to said second input signals to produce
        second output signals,
                combining means for combining said first and second output signals to
        produce control signals, and
                actuator means adapted to respond to said control signals and to produce
35   control disturbances to thereby control said continuing disturbance.

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15.   The system of claim 14 wherein said first compensation signals are derived by filtering said control signals.

16.   The system of claim 14 wherein said second compensation signals are derived by filtering said control signals.

17.   The system of claim 14 wherein said second compensation signals are derived by filtering said second output signals.

18.   The system of claim 14 wherein said first filter means are adaptive filters.

19.   The system of claim 18 wherein a characteristic of each said first filter means is adapted in response to said residual signals.

20.   The system of claim 18 wherein a characteristic of said first filter means is adapted in response to said second input signals

21.   The system of claim 14 wherein a characteristic of each said second filter means is adapted in response to said residual signals.

22.   The system of claim 21 wherein said adaption is based on a Least Mean Square algorithm.

23.   The system of claim 14 wherein said filter means are digital Finite Impulse Response filters.

24.   The system of claim 14 wherein said filter means are digital Recursive filters.

25.   The system of claim 14 and including means for on-line identification.
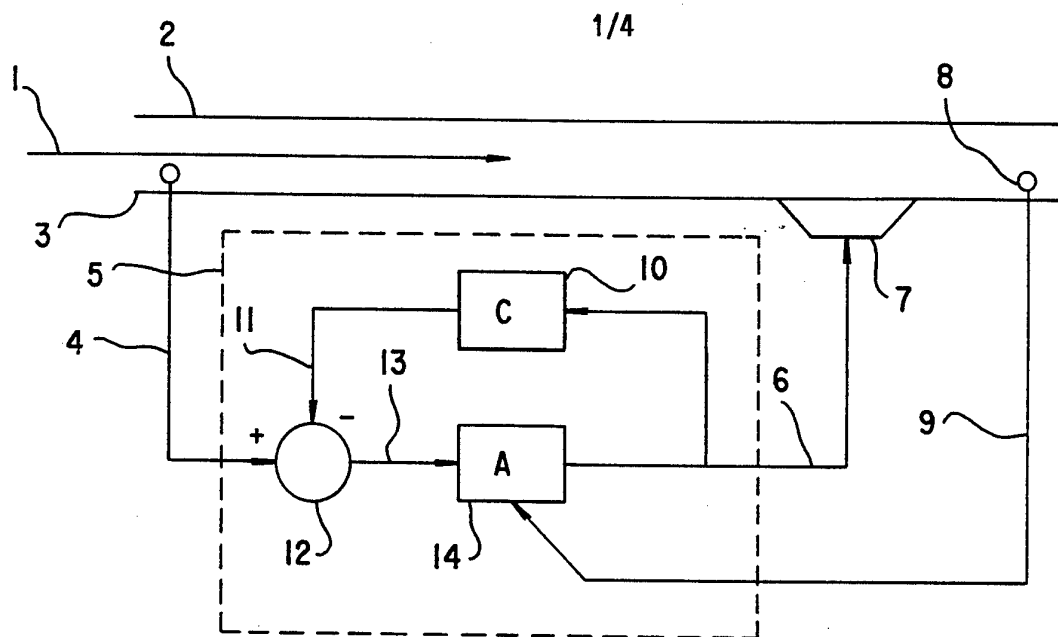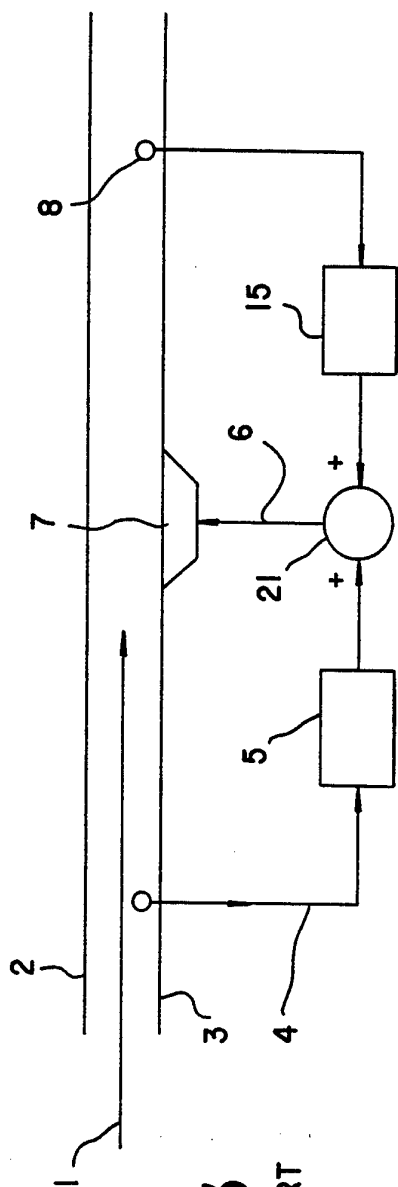
Fig.1
PRIOR ART
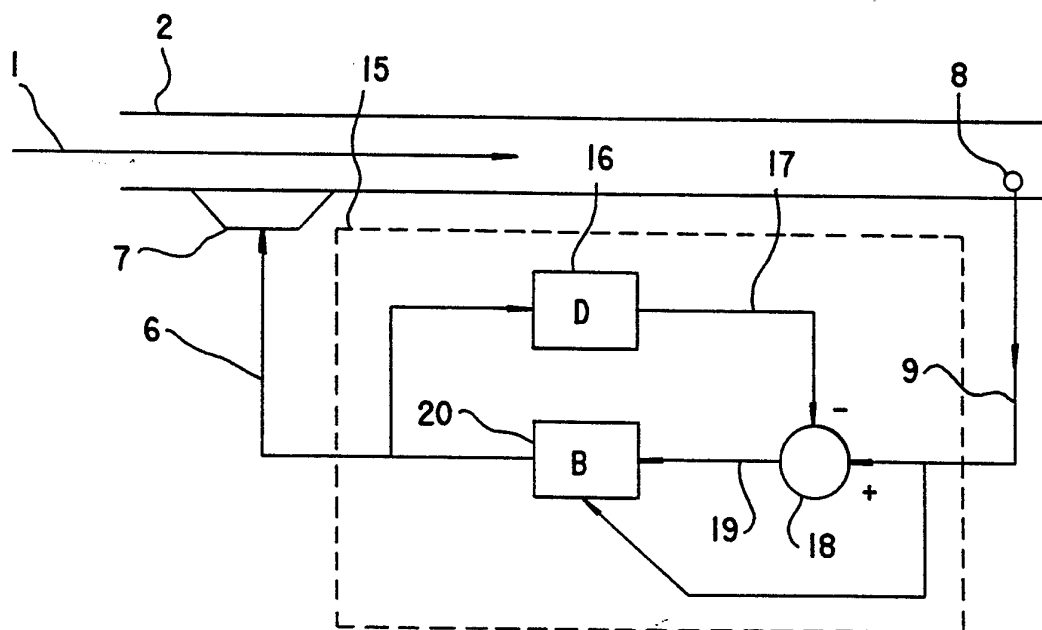
Fig.2
PRIOR ART

Fig.3
PRIOR ART

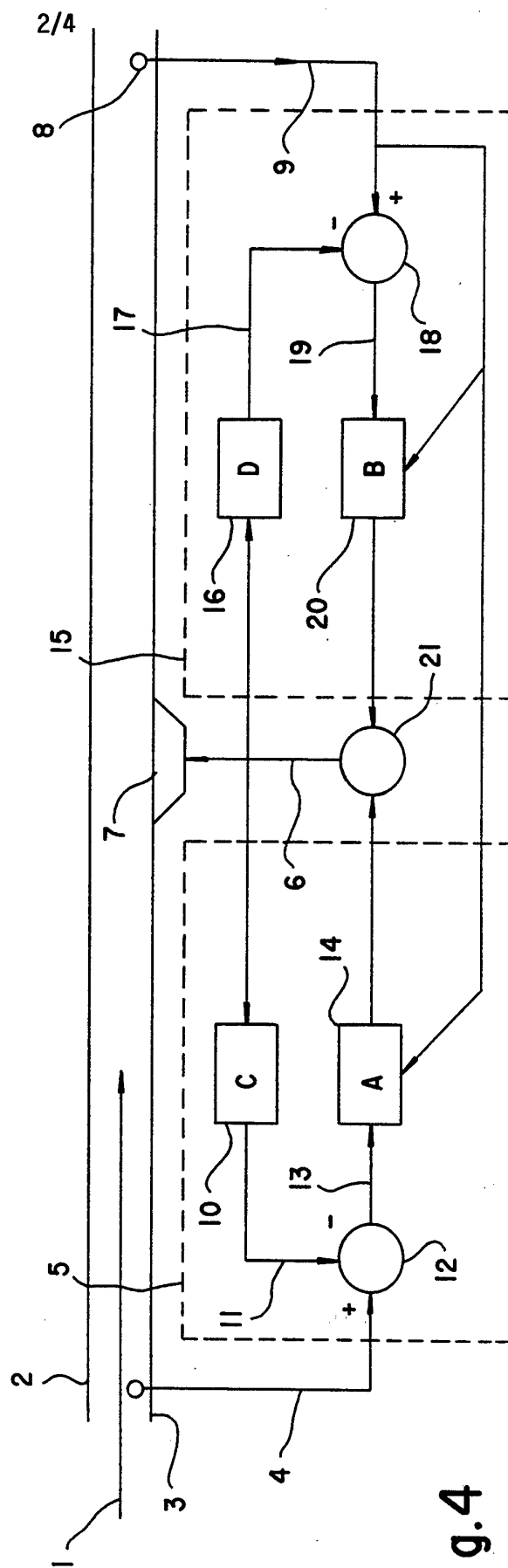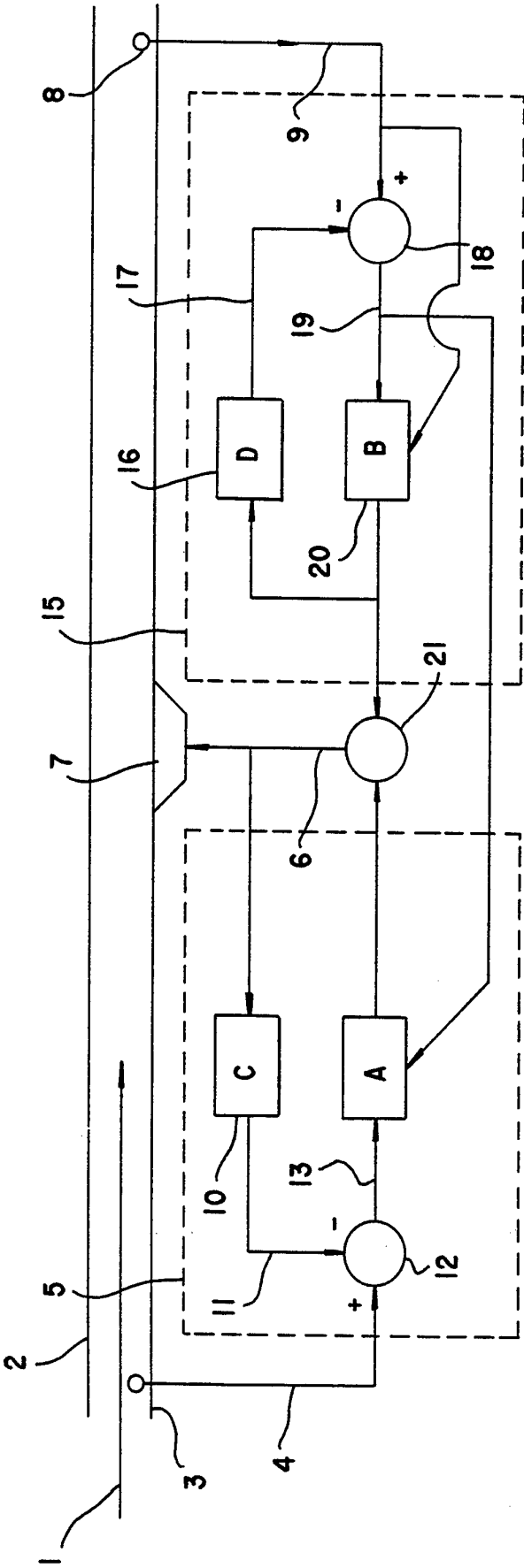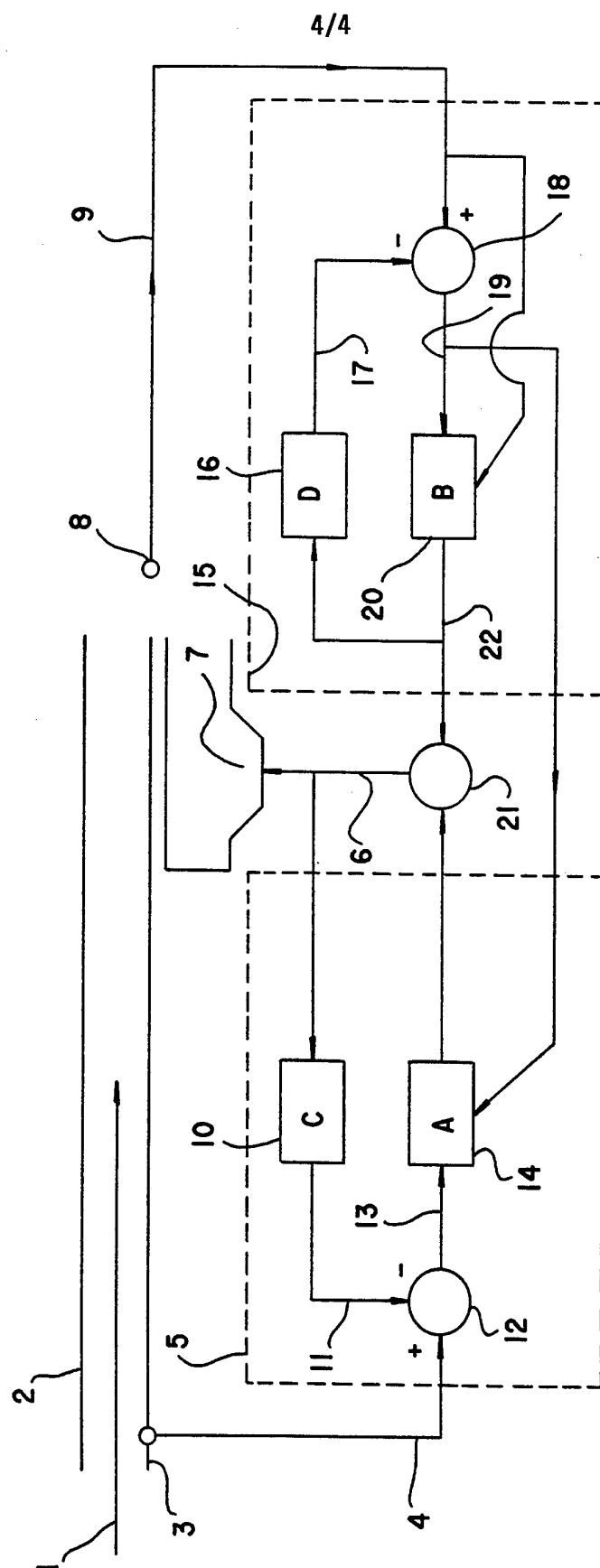Fig.4

Fig.5

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Fig.6

International application No.
PCT/US95/01039

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :H03B 29/00
US CL :381/71

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 381/71,94

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US, A, 5,377,276 (TERAI ET AL) 27 December 1994, Figure 2, column 7, lines 7-42.	1, 12, 14, 24
A	US, A, 5,295,192 (HAMADA ET AL) 15 March 1994, see abstract.	1-25
A	US, A, 5,146,505 (PFAFF ET AL) 08 September 1992, see abstract.	1-25
A	US, A, 4,769,847 (TAGUCHI ET AL) 06 Septemberf 1988, see abstract.	1-25
A	US, A, 4,536,887 (KANEDA ET AL) 20 August 1985, see abstract.	1-25

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

*	Special categories of cited documents:	"T"	later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"A"	document defining the general state of the art which is not considered to be part of particular relevance		
"E"	earlier document published on or after the international filing date	"X"	document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"L"	document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)	"Y"	document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"O"	document referring to an oral disclosure, use, exhibition or other means		
"P"	document published prior to the international filing date but later than the priority date claimed	"&"	document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
14 MARCH 1995	15 MAY 1995
Name and mailing address of the ISA/US Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231	Authorized officer Stephen Brinich
Facsimile No. (703) 305-3230	Telephone No. (703) 305-4390

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