

AN ACCELERATION FEEDBACK SYSTEM

BY HANS KLARSKOV MORTENSEN

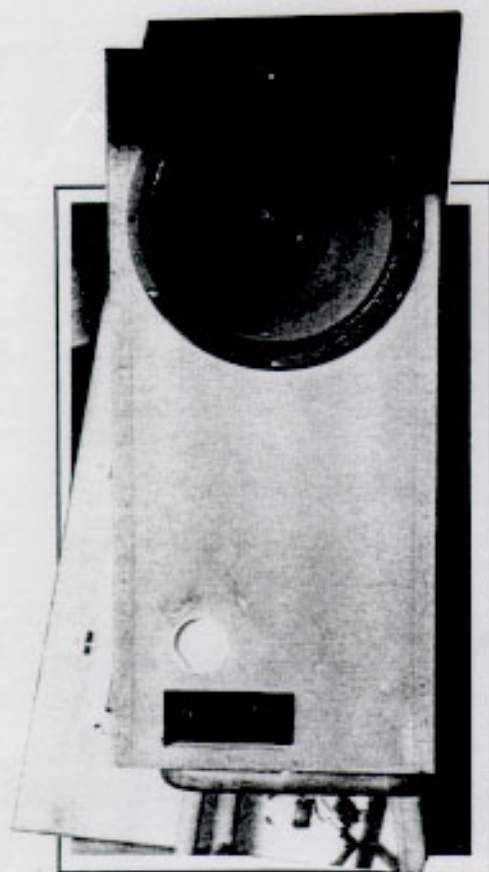


PHOTO 1: The subwoofer with a 12 inch driver.

This article differs somewhat from the standard do-it-yourself (DIY) articles, which usually offer complete construction information. Sometimes I am bored by all the "do this, don't do that"—all of which is necessary for a construction story. I decided to try another type of DIY article, including notes, ideas, formulas and considerations.

INTRODUCTION. Since the invention of the dynamic loudspeaker, we have

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Mr. Mortensen lives in Denmark, is a full-time high school teacher with a Masters Degree in Danish and English, and has published some works of literary criticism (in Danish). He has been an active, amateur jazz musician, but during the last couple of years he has devoted most of his spare time to high fidelity design, contributing to the Danish magazine, Hi-Fi & Electronics.

concentrated on improving this ingenious and imperfect device. I am quite sure if the inventor could listen to the new breed of his device, he would hardly believe his own ears. No doubt we will see still improved versions of the dynamic driver. However, recent years have shown a return of interest in circuitry designed to compensate for various speaker shortcomings. This is partly made possible by the vast improvement of relatively inexpensive, good quality audio circuits which have appeared during the last two decades. Some years ago KEF introduced their K-UBE in connection with some of their high-end products, and recently

even B&O has introduced an active equalized loudspeaker system.

In this article we will look at two ways to improve the problematic low-frequency response of dynamic speaker systems.

CHEATING NATURE. The human mind has come up with many ways of "tricking" nature in an attempt to make small speakers reproduce frequencies which are, in theory, impossible for their size. In the following, I will use "low frequency" to signify the register from approximately 120 to 20Hz.

In terms of quality we need linearity

$$F_0 = \frac{1}{2\pi R_1 \sqrt{C_1 C_2}}$$

$$Q_0 = \frac{1}{2R_1 R_2} \sqrt{\frac{C_1}{C_2}}$$

$$\frac{R_2}{R_1} = \frac{1}{Q_0} \sqrt{\frac{C_1}{C_2}} - 2$$

$$F_1 = \frac{1}{\pi R_1 C_1}$$

$$F_2 = \frac{1}{2\pi (R_2 + R_3) C_2}$$

$$A_{dc} = \frac{R_3}{2R_1}$$

$$A_{\infty} = \frac{R_3}{R_2 + R_3} < 1$$

$$A_{dc} = \frac{R_3}{R_2 + R_3}$$

$$A_{\infty} = \frac{2R_1}{R_2 + R_3}$$

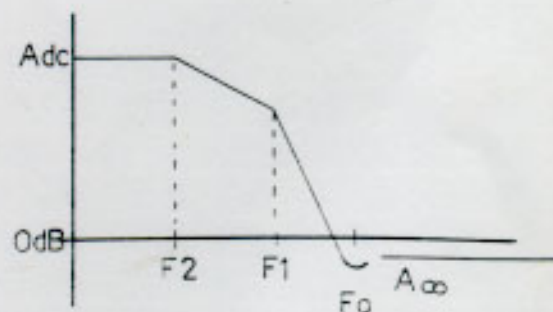
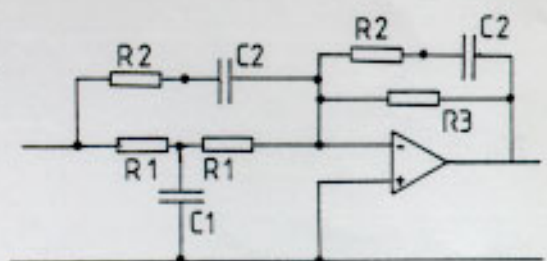


FIGURE 1: The Q compensation circuit developed by Siegfried Linkwitz along with the design formulas. Reprinted by permission from *Electronics and Wireless World*.

For C7=0.1uF and C8=0.047uF and C9=0.33uF you get following resistor values for the Fc and Qs as indicated:

Fc	25.0	Qc	0.4	R13c	77605.6	R14c	100322.6	R15c	7720.5
Fc	50.0	Qc	0.4	R13c	39842.8	R14c	54161.8	R15c	3860.3
Fc	75.0	Qc	0.4	R13c	26561.9	R14c	36107.9	R15c	2573.5
Fc	100.0	Qc	0.4	R13c	19921.4	R14c	27080.9	R15c	1930.1
Fc	125.0	Qc	0.4	R13c	15937.1	R14c	21664.7	R15c	1544.1
Fc	150.0	Qc	0.4	R13c	13280.9	R14c	18053.9	R15c	1286.8
Fc	175.0	Qc	0.4	R13c	11383.7	R14c	15474.8	R15c	1102.9
Fc	200.0	Qc	0.4	R13c	9960.7	R14c	13540.4	R15c	965.1
Fc	25.0	Qc	0.6	R13c	119528.4	R14c	72215.7	R15c	11580.0
Fc	50.0	Qc	0.6	R13c	59764.2	R14c	36107.9	R15c	5790.4
Fc	75.0	Qc	0.6	R13c	39842.8	R14c	24071.9	R15c	3860.3
Fc	100.0	Qc	0.6	R13c	29882.1	R14c	18053.9	R15c	2895.2
Fc	125.0	Qc	0.6	R13c	23905.7	R14c	14443.1	R15c	2316.2
Fc	150.0	Qc	0.6	R13c	19921.4	R14c	12036.0	R15c	1930.1
Fc	175.0	Qc	0.6	R13c	17075.5	R14c	10316.5	R15c	1654.4
Fc	200.0	Qc	0.6	R13c	14941.1	R14c	9027.0	R15c	1447.6
Fc	25.0	Qc	0.8	R13c	159371.2	R14c	54161.8	R15c	15441.0
Fc	50.0	Qc	0.8	R13c	79685.6	R14c	27080.9	R15c	7720.5
Fc	75.0	Qc	0.8	R13c	53123.7	R14c	18053.9	R15c	5147.0
Fc	100.0	Qc	0.8	R13c	39842.8	R14c	13540.4	R15c	3860.3
Fc	125.0	Qc	0.8	R13c	31874.2	R14c	10832.4	R15c	3088.2
Fc	150.0	Qc	0.8	R13c	26561.9	R14c	9027.0	R15c	2573.5
Fc	175.0	Qc	0.8	R13c	22767.3	R14c	7737.4	R15c	2205.9
Fc	200.0	Qc	0.8	R13c	19921.4	R14c	6770.2	R15c	1930.1
Fc	25.0	Qc	1.0	R13c	199214.0	R14c	45329.4	R15c	19301.3
Fc	50.0	Qc	1.0	R13c	99607.0	R14c	21664.7	R15c	9650.6
Fc	75.0	Qc	1.0	R13c	66404.7	R14c	14443.1	R15c	6433.8
Fc	100.0	Qc	1.0	R13c	49803.3	R14c	10832.4	R15c	4825.3
Fc	125.0	Qc	1.0	R13c	39842.8	R14c	8665.9	R15c	3860.3
Fc	150.0	Qc	1.0	R13c	33202.3	R14c	7221.6	R15c	3216.9
Fc	175.0	Qc	1.0	R13c	28459.1	R14c	6189.9	R15c	2757.3
Fc	200.0	Qc	1.0	R13c	24901.0	R14c	5416.2	R15c	2412.7
Fc	25.0	Qc	1.2	R13c	239056.8	R14c	36107.9	R15c	23161.6
Fc	50.0	Qc	1.2	R13c	119528.4	R14c	18053.9	R15c	11580.0
Fc	75.0	Qc	1.2	R13c	79685.6	R14c	12036.0	R15c	7720.5
Fc	100.0	Qc	1.2	R13c	59764.2	R14c	9027.0	R15c	5790.4
Fc	125.0	Qc	1.2	R13c	47811.4	R14c	7221.6	R15c	4632.3
Fc	150.0	Qc	1.2	R13c	39842.8	R14c	6018.0	R15c	3860.3
Fc	175.0	Qc	1.2	R13c	34151.0	R14c	5158.3	R15c	3308.8
Fc	200.0	Qc	1.2	R13c	29882.1	R14c	4513.5	R15c	2895.2
Fc	25.0	Qc	1.4	R13c	278899.6	R14c	30949.4	R15c	27021.8
Fc	50.0	Qc	1.4	R13c	139449.8	R14c	15474.8	R15c	13510.9
Fc	75.0	Qc	1.4	R13c	92966.5	R14c	10316.5	R15c	9007.3
Fc	100.0	Qc	1.4	R13c	69724.9	R14c	7737.4	R15c	6755.5
Fc	125.0	Qc	1.4	R13c	55779.9	R14c	6189.9	R15c	5404.4
Fc	150.0	Qc	1.4	R13c	46483.3	R14c	5158.3	R15c	4503.4
Fc	175.0	Qc	1.4	R13c	39842.8	R14c	4421.4	R15c	3860.3
Fc	200.0	Qc	1.4	R13c	34862.5	R14c	3860.7	R15c	3377.7
Fc	25.0	Qc	1.6	R13c	318742.5	R14c	27080.9	R15c	30882.1
Fc	50.0	Qc	1.6	R13c	159371.2	R14c	13540.4	R15c	15441.0
Fc	75.0	Qc	1.6	R13c	106247.5	R14c	9027.0	R15c	10294.0
Fc	100.0	Qc	1.6	R13c	79685.6	R14c	6770.2	R15c	7720.5
Fc	125.0	Qc	1.6	R13c	63748.5	R14c	5416.2	R15c	6174.4
Fc	150.0	Qc	1.6	R13c	53123.7	R14c	4513.5	R15c	5147.0
Fc	175.0	Qc	1.6	R13c	45534.6	R14c	3860.7	R15c	4411.7
Fc	200.0	Qc	1.6	R13c	39842.8	R14c	3385.1	R15c	3860.3

FIGURE 2: Low-pass filter component values. Also included is the program which calculates component values for the low-pass filter, written in GW-Basic. It should be fairly easy to rewrite in another Basic dialect.

and low distortion and in terms of quantity we need SPL levels of at least 90dB. Fulfilling these demands is very difficult indeed. Some of the more conventional ways comprise—in order of physical dimension—horns, transmission lines and reflex systems, all of which are very well described.

There is another way: electronic compensation of an "ordinary" dynamic driver for Q and resonance frequency (F_0). Electronic compensation is preferable to passive compensation for several reasons. The most obvious is that you can realize the low register with quite small speakers. Another is that you have a large selection of usable drivers, thus reducing the total cost of the system. Let's look at two different sorts of electronic compensation circuits.

SEMIACTIVE Q COMPENSATION. The circuit in Fig. 1 was developed by Siegfried Linkwitz and described in *Electronics and Wireless World*¹ along with the design formulas. If you know the Q and resonance frequency of the speaker in question it is only a matter of using the formulas to get the right component

values. Of course you must compromise somewhat since the driver's cone excursion is limited, and every time the F_0 is lowered by a factor of two the cone excursion is increased four times. But if you have a fairly good driver you should have no problems at reasonable levels.

As I mentioned, you must know the Q and F_0 of the complete speaker system in order to use the semiactive compensation circuit. Measure the Q as described below on the actual terminal of the closed-box speaker system you want to compensate. Include the influence of possible crossover components and their influence on Q. Here is how you do it:

1. Measure the DC resistance of the speaker with an ohmmeter.

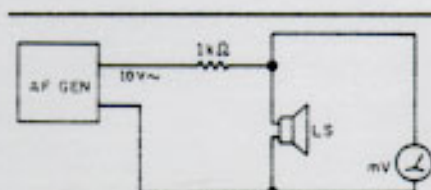


FIGURE 3: Measuring setup for the audio frequency generator.

2. You need a calibrated audio frequency generator which can deliver 10V RMS (you may need a power amplifier if your generator cannot supply this voltage). See Fig. 3.

Now you can read the impedance (Z) of your speaker directly since 10mV RMS corresponds to a Z of 1Ω. First, find the frequency where Z is at maximum; this is F_0 . Then calculate R_{AV} from

$$R_{AV} = \sqrt{R_{MAX} \cdot R_{DC}}$$

3. Next step is finding F1 and F2, the frequency where $Z = R_{AV}$ on each side of the peak. Now calculate Q:

$$Q = \sqrt{\frac{R_{DC}}{R_{MAX}}} \cdot \frac{F_0}{F_2 - F_1}$$

(As a check, $F_0 = \sqrt{F_1 \cdot F_2}$)

You are now ready to calculate the necessary component values. See formulas in Fig. 1.

The mathematics of the design formula are not complicated, but a bit tedious, since you'll have to do the calculations a number of times to get not only com-

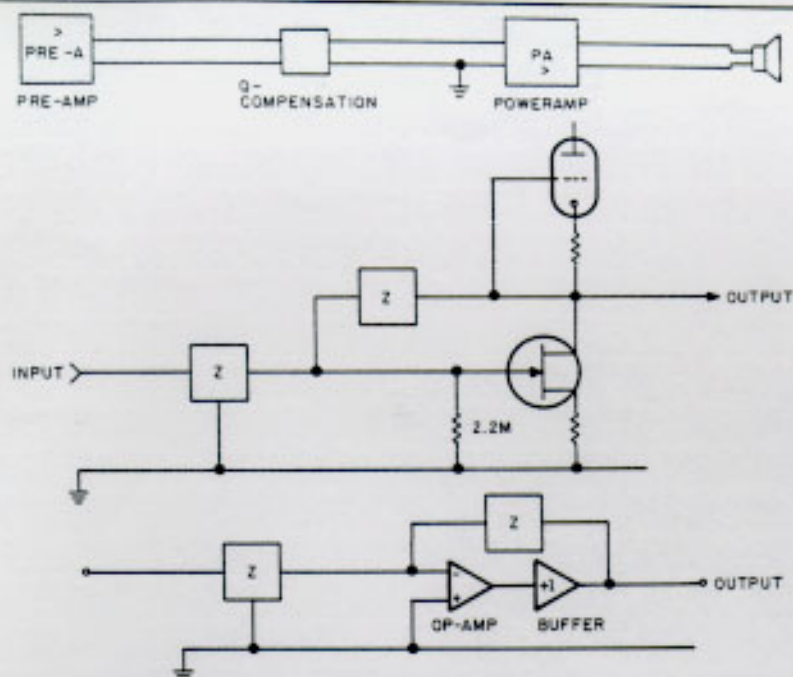


FIGURE 4: Practical details in connection with the use of the Q compensation network.

ponent values which are obtainable but also an F1 which is not too low. Mostly for the fun of it, I wrote a Turbo-Prolog program which you can obtain on disk by writing to Old Colony Sound (see parts list).

Using this circuit in a two- or three-amp system with an electronic crossover network is straightforward: The amplifying device is just a very good op amp (Fig. 4). Using the system in a single-amp system requires a very good amplifying device. No op amp I tried alone was satisfactory. The loss of dynamics and detail was obvious indeed. I got a far better result with the one in Fig. 5—an FET-tube-shunt-regulated push-pull amplifier.

The circuit is based on the well known valve, SRPP, with which Anzai and Jean Hiraga have experimented.² The input and output impedance of this circuit are so high and low, respectively, that the error you introduce by disregarding them is irrelevant, provided you use an R1 between 47 and 150kΩ and a C1 between 0.22 and 0.56μF. In my system this didn't affect the tonal balance or the sense of dynamics outside the low register at all.

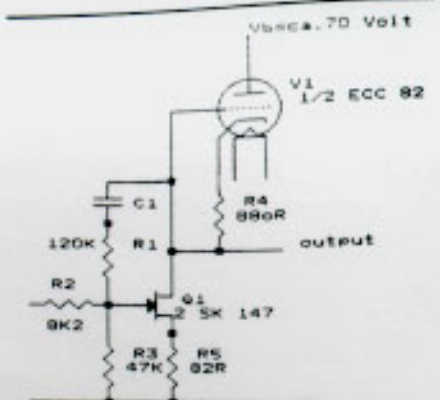
I have tried the Q compensation circuit with two systems. The first one was a three-way single-amp system, where I compensated the KEF B200G woofer. The result was quite astonishing: a far better transient response and a much richer low register. In this case I used the FET-tube-shunt-regulated push-pull amplifying device.

The second system was my old ESS Amt 1. This speaker is extremely ambi-

tious and in many respects still very good, mainly because of the excellent Heil driver. It is also well built with a strong cabinet, and the crossover network is stuffed with lots of heavy and expensive copper wiring.

The slave-woofer system, also ambitious, sounds horrible, mainly because of a strong peak around 50–60Hz and a very poor transient response. To make things even more interesting it's almost impossible to fit the speaker into ordinary living rooms. For quite a while I had been looking for a replacement, but the upper midrange and treble of the Heil are very difficult to live without, whereas it is nearly impossible to live with the horrible bass reproduction.

So I removed the passive slave radiator and closed the hole with a piece of chip-board, adding more damping material to



Open loop gain:
191x = 45.6 dB
Z_{in} = ca. 9000
S-N: 120 uV / -78 dB ref. 1 Vrms

Closed loop
Gain 14.6x = 23 dB
THD: 0.03%, v. 2.5 Volt rms 1 KHz.
SN: 70uV -63.1 dB ref 1Vrms

FIGURE 5: A modernized version of the SRPP circuit which will work at about 70V. Open-loop condition: R1, 2 and C1 removed, otherwise no changes.

lower the Q and improve the woofer's transient response. The resulting Q turned out to be 0.8 with a resonant frequency around 45Hz. Then I calculated a Q compensation circuit. This time I used an op amp and the superb home-spun video buffer described in *Audio Amateur* 2/88, p. 24. For further details see Fig. 6.

And the result? A remarkable improvement of the woofer's transient response and a much cleaner bass reproduction. The only snag is a heavy loss of sensitivity, which is, however, easily compensated for with the controls on the speakers. With this update I feel a better speaker would cost at least three times the original price.

However good this idea is, it still has one problem: The increased cone movement in the low register will cause greatly increased distortion. Even though

Continued on page 15

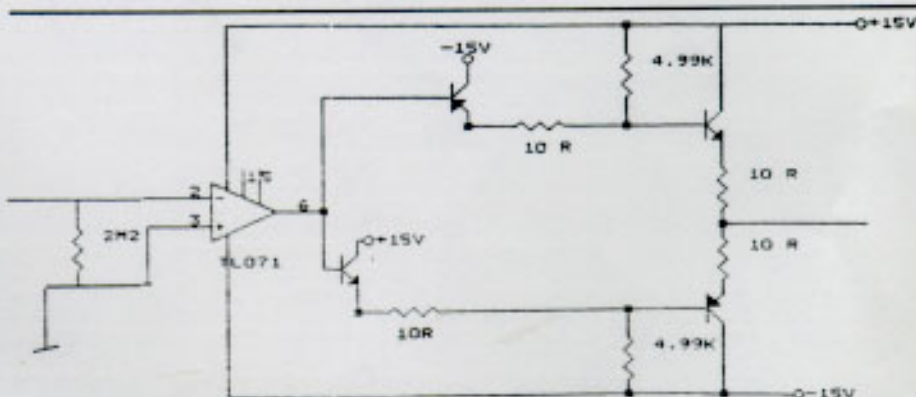


FIGURE 6: Another amplifying device for the Q compensation network. The buffer compensates for the op amp's poor performance under near-unity gain conditions.

Continued from page 12

a couple percent distortion at 50Hz is not important it is nevertheless undesirable, and 10% at 30Hz is decidedly unwellcome. This distortion may be eliminated in two ways: very expensive drivers in fairly large cabinets or via electronics. We'll look at the latter.

ACCELERATION FEEDBACK. Nowadays the most well known feedback system is probably Infinity's servo-system in (among others) their IRS5. Other manufacturers have experimented with servo/feedback systems as well. Several years ago Philips marketed a "Motional Feedback" loudspeaker system [Fig. 7]. The bass unit of this active system was unique in that it had a small piezo accelerometer and an FET impedance converter mounted on the cone. The accelerometer's output was then returned to the input of the power amp and thus created a feedback loop. The feedback circuit [Fig. 7] is marked by quite heavy phase and filter compensations around R647, 652, 651 and C521, 523 and 525, and the feedback path of the low-frequency power amp is also heavily compensated. Another author³ gave the reason for some of these compensations:

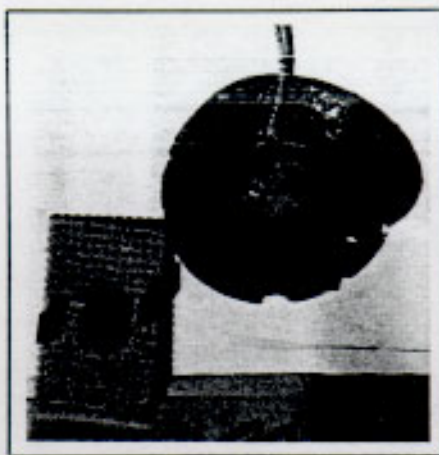


PHOTO 2: Left, the impedance converter. Right: the piezo transducer glued onto the inside of the dust cap.

The piezo is far from linear. Part of this nonlinearity, however, was due to the limited input impedance of the Philips FET.

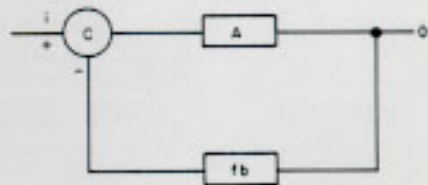
On the basis of the Philips speaker, Mr. D. De Greef and J. Vandewege⁴ [Fig. 8] succeeded in designing a single-amp system with "motional feedback" compensation of the bass unit. Looking at this design from the low-frequency viewpoint it is interesting to notice that the piezo is hardly compensated at all. In

fact, the De Greef and Vandewege circuit is remarkably simple.

The idea reappeared in a new version by R. Conell.⁵ Conell's idea was to modify an ordinary driver by adding a simple piezo accelerometer and a vastly improved impedance converter. This is a very good idea, and it makes it possible for us to experiment with many drivers other than those from Philips.

Cone movements can be monitored in other ways. Elbert Hendricks has described a version using a coil whose inductance is like that of the speakers in a bridge configuration.⁶ The function of the bridge is to separate the drive signal from the driver's "error" signal. Mr. Hendricks' circuit does, however, require quite a lot of adjustment and modification to suit drivers other than the ones he used.

The basic principle of the motional feedback system can be seen in the block diagram below:



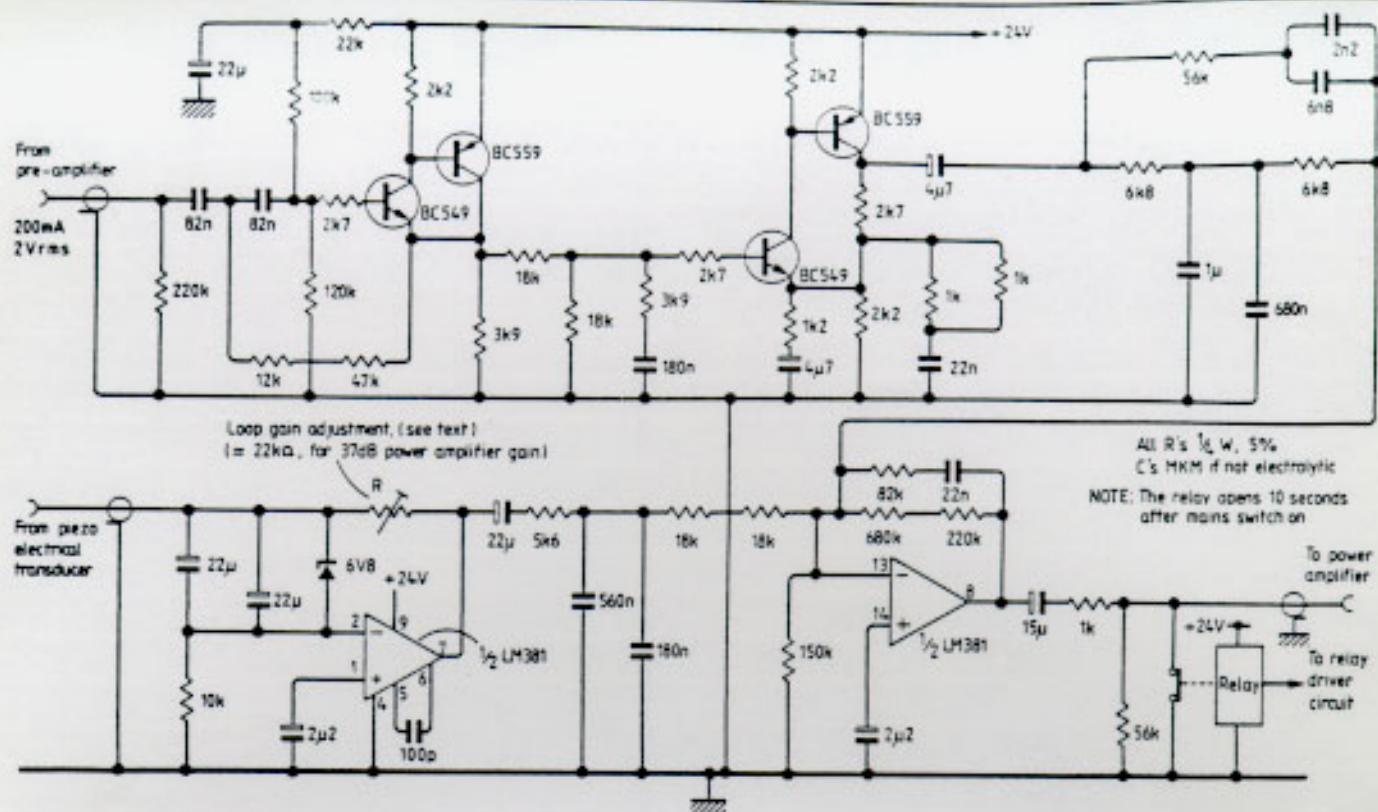
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A represents the power amplifier and fb the feedback path. At point C the feedback signal is deducted from the incoming signal so the actual output at point O is the difference between the incoming signal and feedback signal. If $fb \cdot A$ is much higher than unity then O will be equal to $1/fb \cdot i$. This fact means that the resulting sound pressure level (SPL) is independent of the driver, provided that the entire system is stable.

In practice we cannot do as well, but with luck a modest driver can be persuaded to give this performance: in a small cabinet (I used approximately 40

liters), I obtained a ± 3 dB response from 15–250Hz, with a few percent distortion. The price is modest: an extra amplifier and a feedback network.

DRIVER CHOICE. In choosing a driver for this system you need not look at its specifications. They will quite likely change significantly when the unit is modified. Three parameters you should consider critical are: the unit's power rating, cone excursion limit and driver Q.

The first parameter is easily taken care of: 50W is enough for most applications in ordinary living rooms.

The second is more problematic: If you want high SPL at 20Hz out of an 8 inch speaker you will need large cone movements. One of my prototypes used two 8 inch speakers with 1cm maximum cone excursion limits. This system does reproduce 20Hz notes, but not very loudly. Three or four 8 inch drivers per system should perform quite well. You can also increase the size of the drivers. A second prototype uses a 12 inch unit with maximum 6-7mm cone movement. This one gives somewhat richer sound, and the actual levels are a bit higher than the 8 inch units. When planning a system along these lines remember that the cone amplitude is constant above the resonance frequency. It increases with a factor of four every time the frequency is lowered by a factor of two. Therefore a low resonance frequency is preferable.

The third parameter, Q , affects the transient response. You should aim at a low total Q . A total Q of 0.5 is good, but somewhat higher quality factors will work.

Keep in mind that loud 20Hz notes demand large cones and large amplitudes. Infinity's biggest system uses five 12 inch drivers.

DRIVER MODIFICATION. The first problem: You need a small piezo disk (Photo 3). I found mine inside inexpen-

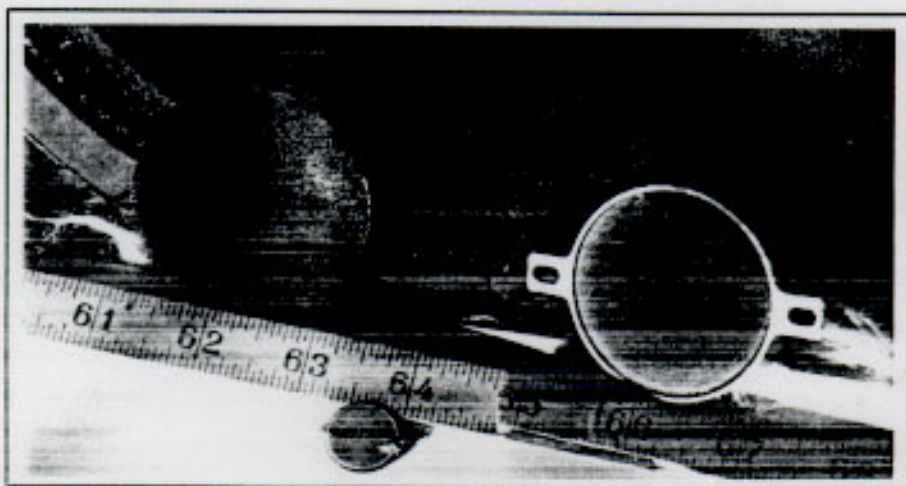


PHOTO 3: From the left: the dust cap, the piezo disk and the piezo speaker.

sive piezo loudspeakers, see example in the same photo. I saved the small piezo disk and threw the rest away. Unfortunately you cannot use the ever-beeping computer types as they are too stiff. Conell used a similar type of piezo, bought in Great Britain; I bought my two different looking piezo drivers in Denmark, but their performance seemed uniform. Since they are used to record frequencies far below any of their numerous resonances they perform well. This doesn't prove you'll be able to get one with the same performance in the US, but it is likely.

Once you have found the piezo disk you proceed to the driver. It's best to mount the accelerometer inside the dust cap which is normally glued to the cone with thermoplastic glue. This can be lifted with a hot knife. If you cannot do it this way you'll have to cut off the dust cap; gluing it back in place is not difficult.

With the dust cap removed you are ready to mount the accelerometer whose edges must be free of the dust cap so you'll need a small block of wood or acrylic (3 x 3 x 2mm) to lift the piezo disk from the surroundings. The dust cap must be stiff; coat it with a thin layer of

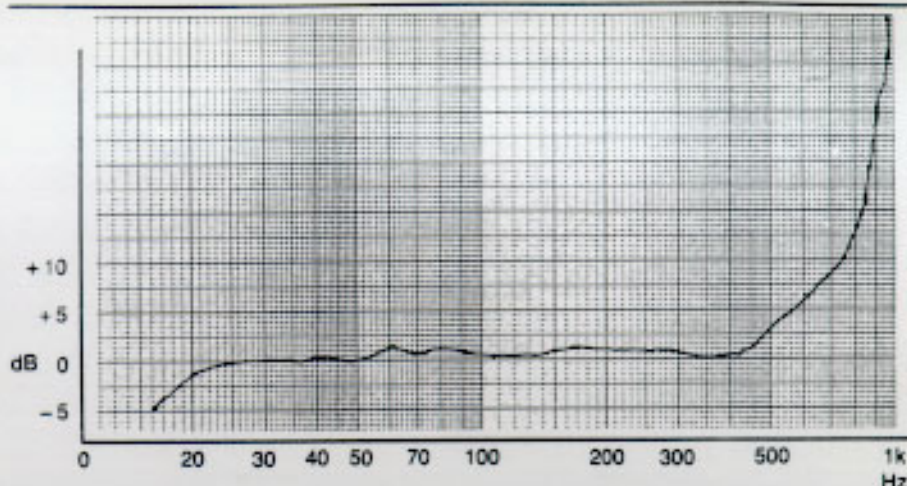


FIGURE 9: Typical transducer response. The curve was made keeping the output from the speaker constant and by plotting the transducer response picked up at point B in Fig. 10. The power amplifier for the speaker was fed directly from the audio frequency generator, and the signal picked up with a Bruel & Kjaer condenser microphone.

epoxy resin, the same as you use to glue the piezo in place.

When the epoxy has hardened, solder some thin flexible wires to the piezo's terminals. You can either twist the leads together firmly and fix them inside the dust cap, or use shielded wire. The latter is preferred because it minimizes the risk of hum pickup.

Now punch a small hole in the cone

for the leads from the piezo and replace the dust cap with thermoplastic glue.

This modification should be done carefully—anything loose will rattle horribly.

While the glue is drying proceed to the impedance converter. It can be built on a small piece of vero-board or the like. Drill one or two holes in the driver chassis, but be careful not to glue the converter onto the chassis. This also must

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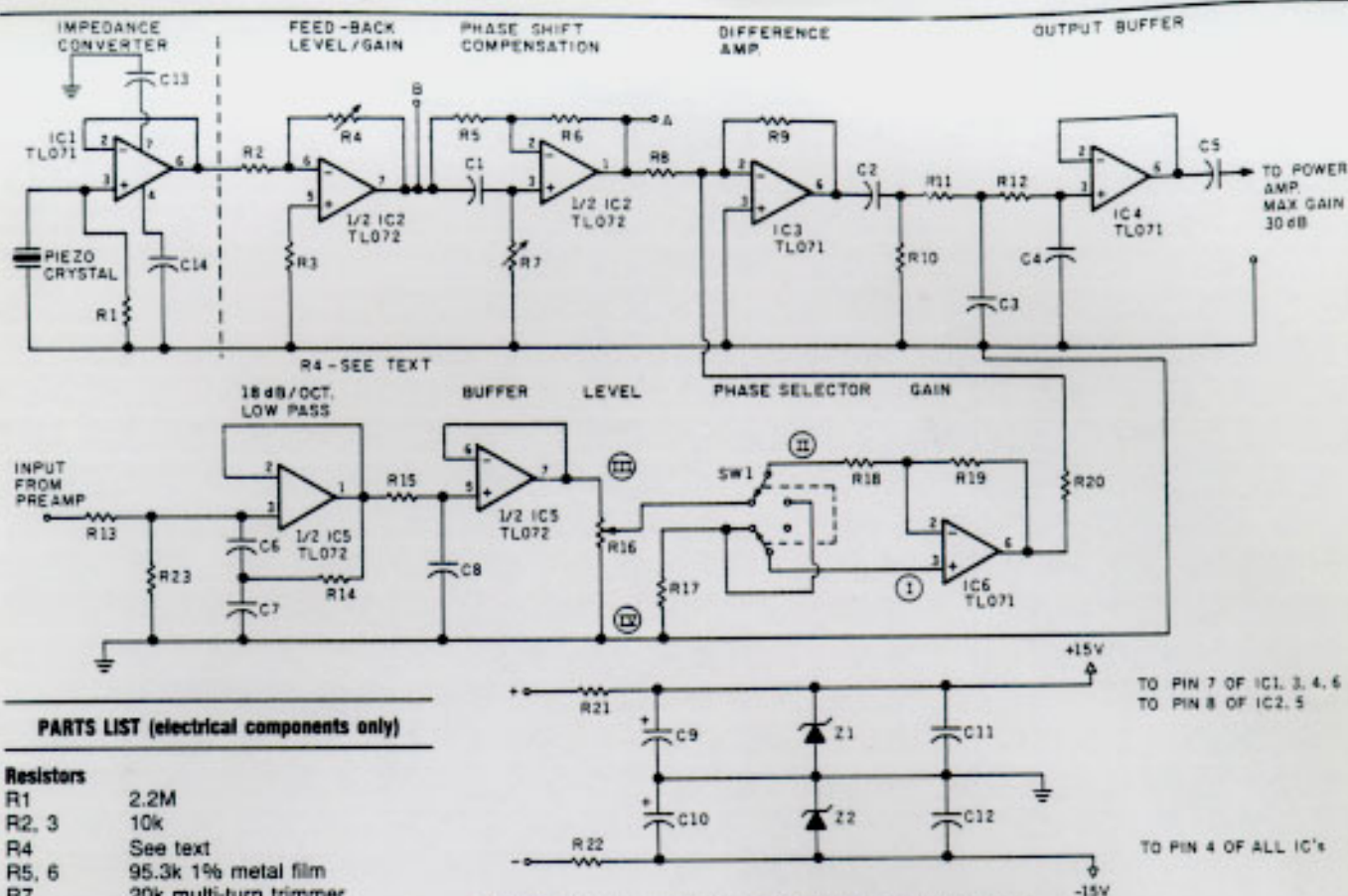


FIGURE 10: Schematic of acceleration feedback system.

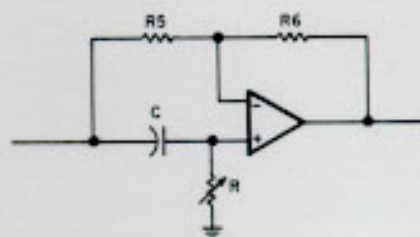
be done with care. As a result of this modification the resonance frequency of the 8 inch driver went down 5Hz, whereas the stiff suspension of the 12 inch unit kept its resonance frequency almost unchanged.

FEEDBACK SIGNAL. Our aim is to get as good a replica of the cone movements as possible, which means the amplitude of the output from the piezo must follow the amplitude of the cone movements. I have modified two sizes: an 8 inch and a 12 inch. In Fig. 9 you see a typical response curve of the relative output from the transducer. I kept the near-field SPL constant while recording the output of the transducer. It appears that the linearity in the region of interest is very good, so good in fact that I didn't find any further compensation necessary.

So far so good, but amplitude is not everything in audio. Phase is also important. In my case a rough estimate of the phase response suggested some 90° phase shift around 70Hz. The other versions along these lines have included quite complex phase-shift compensations, but since you are going to use your own drivers the components I could suggest wouldn't work. Though manipulating phase shift in feedback circuitry

is risky because of the danger of oscillation, we do need some sort of adjustable phase-shift network.

The phase-shift circuit is shown below:



The phase shift is given by:

$$\theta = 180^\circ - 2 \cdot \tan^{-1}(f/F)$$

where θ = the resulting phase shift in degrees at frequency f and where

$$F = 1/(R \cdot C \cdot 2 \cdot \pi)$$

On the complete schematic, Fig. 10, you'll find it around IC2. I will return to the practical adjustment procedure later.

In Fig. 10 you will notice the differential point at the junction of R8 and R31 in combination with IC3. The filter network around R11, 12 and C3, 4 is necessary to ensure stability. The buffer IC4

prevents the power amplifier's input impedance from interfering with this network.

CROSSOVER CONSIDERATIONS.

Many a small closed-box speaker has a rolloff around 12dB/octave. Adding a simple first-order filter will give a rolloff of 18dB, which indicates that the subsystem should be equipped with a third-order 18dB/octave filter. You can see this around IC5. At the end of the article you'll find component lists for various crossover frequencies and Qs. This should make it possible for you to match the subsystem quite closely with your side systems. The extra first-order filter for the side system is best implemented by adding an extra capacitor in series with the input of the power amplifier. It may be calculated from:

$$C = \frac{1}{2\pi \cdot Z_{IN} \cdot F}$$

where C is the capacitor, Z_{IN} is the input impedance of your power amp and F is the required crossover frequency. (The units used are ohms, hertz and farads.)

Since it is difficult to invert the true phase of this subsystem I found it necessary to provide a simple 180° phase selector. This is seen around S1 and IC6. Apart from the influence of the series resistance of R17's wiper, the gain of this stage is unity, both in and out of phase (the difference between the inverted and noninverted gain is maximum 1dB).

THE BOX. I'll leave the box dimensions to you. Remember, the bigger the box the higher the SPL. I have only experimented with boxes of approximately 40 liters.

The box must be absolutely airtight. Any leak will increase distortion and lower the maximum SPL. Fill the box with damping material to reduce the total Q. A total around 0.5 is ideal, but I haven't experimented with it. The resonance frequency is irrelevant (see block diagram under Feedback Signal).

THE POWER AMP. Any good power amplifier will do as long as it has a gain of not more than 30dB and a frequency response down to 3-5Hz. The Hendricks system uses a power amp with no more than 19dB gain. Higher gain and higher cutoff frequency will reduce the feedback system's stability margin. I have used a simple 60W bipolar device. Slew rate and distortion are not important in this case. LSI integrated power amp ICs should work very well.

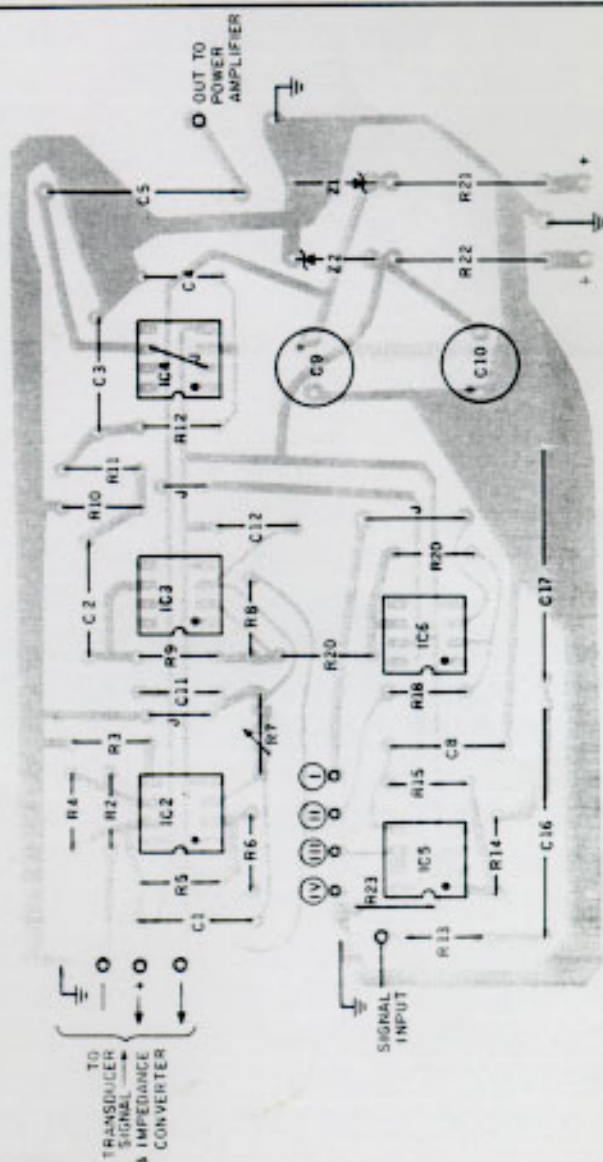
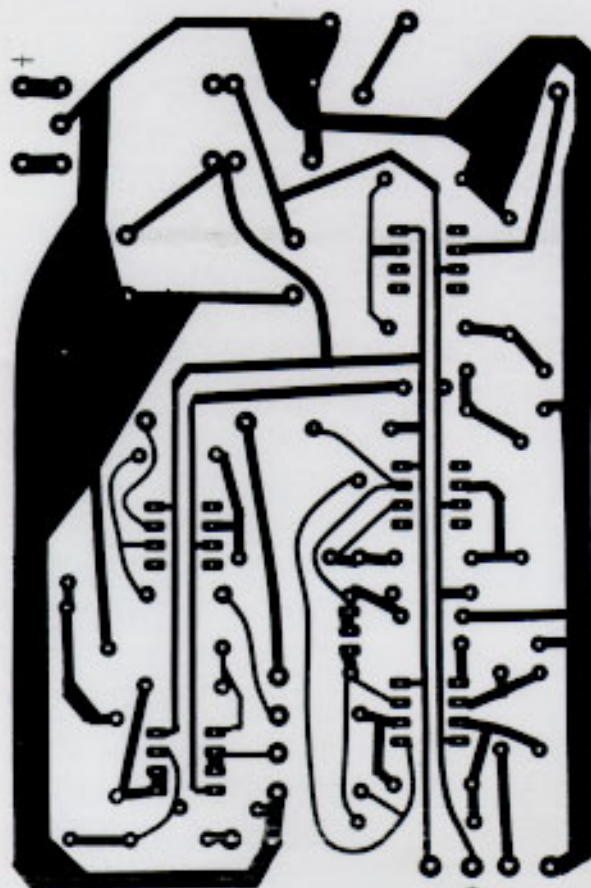


FIGURE 11a: Circuit pattern for the acceleration feedback system.

FIGURE 11b: Stuffing guide.

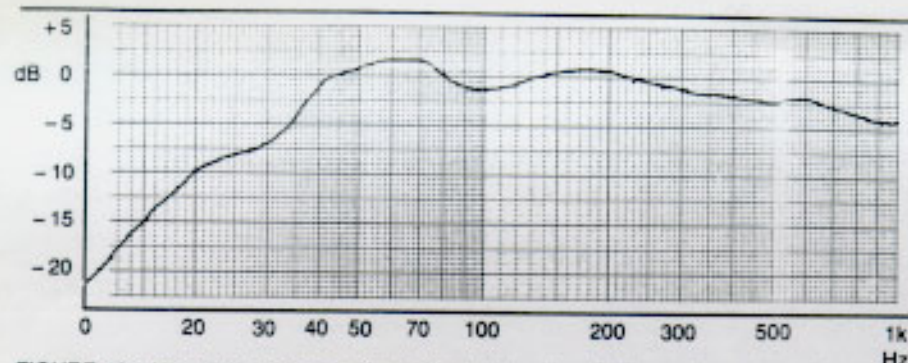


FIGURE 12a: Near-field response of one 8 inch driver in a 40 liter cabinet without feedback.

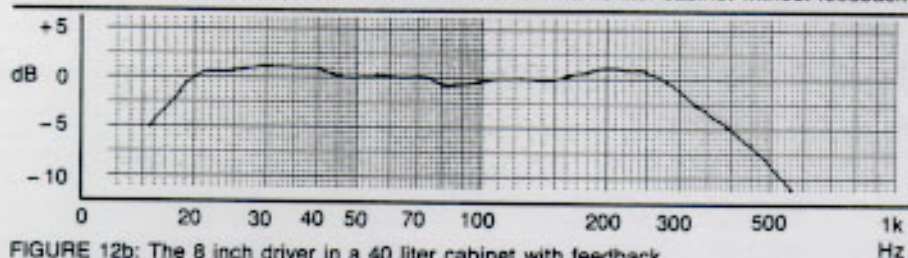


FIGURE 12b: The 8 inch driver in a 40 liter cabinet with feedback.

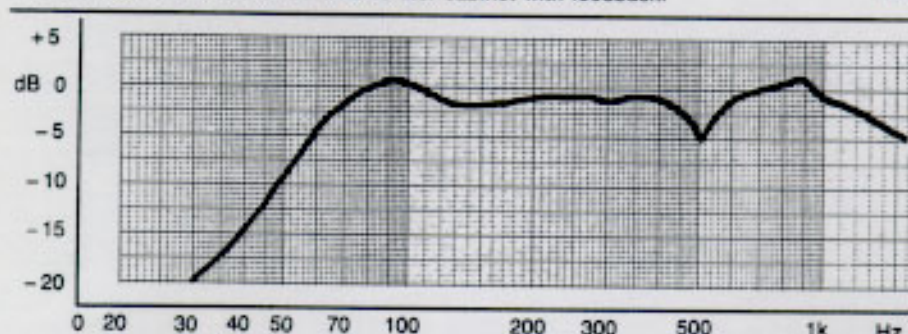


FIGURE 12c: Near-field response of one 12 inch driver in a 35 liter closed box, without feedback.

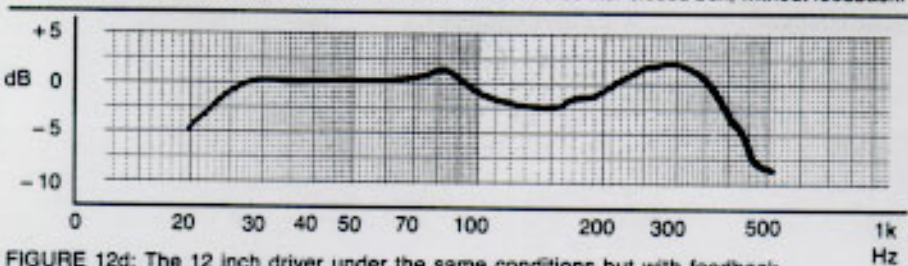


FIGURE 12d: The 12 inch driver under the same conditions but with feedback.

feedback by turning R4. This will increase the gain of IC2 and the signal from the transducer will be allowed into the feedback path. Increase the feedback level until you sense the first small sign of oscillation, and try to keep the oscillation so that it is just barely audible. When you apply feedback, the output from the speaker should decrease. It is now possible to compensate for the phase shift which is the root cause of this oscillation by adjusting R5. When oscillation disappears, increase the feedback level still more, until you sense beginning oscillation again. Remove the oscillation by adjusting R5. Continue in this way until you cannot get rid of any more oscillation.

My system has been stable up to 17dB feedback, but it is not a good idea to take it to the limit. I suggest you experiment with feedback around 7-12dB, as higher feedback levels tend to produce a sub-sonic peak (around 10-15Hz).

You are not through yet. The whole system will need some playing to "burn in" the suspensions of the driver, and the temperature changes of the amplifier need some time to settle. Connect the units to your system and use it extensively for a week or two. After this period you should readjust the drives and replace the trimmer with the value which gives the feedback level you want.

SONIC MERITS. In Figs. 12a-d you see the results of my systems, both before and after the modification. These speak for themselves. Unfortunately I do not have a line writer, so I have plotted the curves instead, but there were no peaks which I haven't plotted.

Measurement is one thing, sound another. Two observations may be of interest: It is striking how little recorded information is present below 40Hz and equally striking as to how much is subjective. Adding bass units of this type is like standing the music on its feet and placing it firmly on the floor of the room. Traditionally, the transient response of feedback systems is their weak feature, but to my ears this properly adjusted system makes it very easy to distinguish, for example, a low bass guitar note from the bass drum. And it is this fundamental you hear, not the second harmonic which we are so used to from pop recordings. But organ music is the genre where this system really comes into its own.

The fly in the ointment? You'll be surprised at the amount of VLF garbage

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POWER SUPPLY. The feedback circuit consumes about 25mA. On the board I have provided space for two 15V/1W zeners and two electrolytic capacitors along with further high frequency decoupling. It goes without saying that these capacitors are not critical at all. The only thing to notice is the inclusion of high frequency decoupling of the impedance converter around IC1. Leaving these out will cause severe problems.

The idea is to get power for the feedback circuit from some other unit. Zener diodes greatly simplify the power supply and you can get the current you need from any source in the audio chain. Resistors R21 and R22 are given by Ohm's law:

$$R = (\text{input voltage} - 15) / 0.025 \text{ [ohms]}$$

ADJUSTMENT. Complete the circuit board but instead of R4 use a 220kΩ trimmer. Connect the phase and level controls with a three-lead shielded wire. Notice that R18 should be connected between the switch and potentiometer. This arrangement saves one long lead. Connect the feedback circuit to the amplifier. Make sure that R4 is as close to 0Ω as possible.

Turn on the power. If you are lucky and everything is done correctly nothing should happen. If you get a very low frequency oscillation reverse the connections to the speaker (or to the piezo, but that is difficult since you'll have to take the speaker out of the cabinet).

Next, apply a small signal of 70-90Hz. Turn the level control till you hear the signal softly. Now try to apply a little

FEEDBACK SYSTEM

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your records and turntables pour out. Some records are not playable with this system unless you include a steep LF-garbage filter. But that is a very different story. 

ACKNOWLEDGMENTS. I thank *Electronics and Wireless World* for permission to quote *Figs. 1* and *8* and Philips for permission to reprint the schematic. I owe my wife, children and friends thanks for their support and patience. Continuous low-frequency sine waves at high SPL are a demanding experience.

REFERENCES

1. Siegfried Linkwitz, "Loudspeaker System Design," *Electronics and Wireless World*, May and June 1978.
2. Jean Hiraga, "Le Preamplificateur SRPP," *Selection de l'Audiophile*, Tome 1: L'Electronique, Paris, 1985.
3. D. De Greef and J. Vandewege, "Acceleration Feedback Loudspeaker," *Electronics and Wireless World*, September 1981.
4. *Electronics and Wireless World*, September 1981. Reprinted here with permission from *Electronics and Wireless World*.
5. R. Conell, "Feedback in Loudspeakers," *Elektor, Electronics*, April 1987.
6. Elbert Hendricks, "Et EMK-MFB Bashottaler System," *High Fidelity*, No. 11 and 12, 1979, Copenhagen, 1979.