THE K-AMP[T.M.] HEARING AID: AN ATTEMPT TO PRESENT HIGH FIDELITY FOR THE HEARING IMPAIRED.

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Abstract

I would like to start by acknowledging that much of what I say today as if I had thought of it myself is based on the thinking and research of others. A few of those others have been especially important to me personally in shaping my own thinking, and I would like to acknowledge them: Elmer Carlson, Hugh Knowles, Edgar Villchur, Dave Pascoe, Margo Skinner (who has my vote as the worlds best experimentalist), Tom Tillman, Sam Lybarger, Jesper Barfod and Hy Goldberg.

My talk has five headings:
I. Why Nobody Wants to be Seen Wearing a Hearing Aid.
II. High Fidelity Hearing Aids are Possible.
III. What Does High Fidelity Mean to the Hearing Impaired?
IV. What's Wrong with the K-AMP hearing aid?
V. Noise Complaints.
VI. Learning to Hear Again in Noise.

I. Why Nobody Wants to be Seen Wearing a Hearing Aid.

Why does nobody want to be seen wearing a hearing aid? Why is there a stigma associated with wearing a hearing aid? Edgar Villchur has speculated that it is because hearing aids didn't work. They left their wearer with an obvious handicap: inability to hear in noise. This is not the case with eyeglasses. If you see someone wearing eyeglasses, you presume that they can see perfectly well. You don't rush up to them with large print editions. But if you see someone wearing a hearing aid, your past experience tells you that you will have to raise your voice in order for you to communicate with that person if there is any noise present.

Why is this? Well, it is increasingly clear that a series of seemingly sensible engineering and marketing decisions resulted in hearing aids that made only a fraction of the speech cues audible for many listeners. To explain how that happened, let's start with some facts.

A. The Facts

Normal-hearing people tested in noise at a -5 dB signal-to-noise ratio can get 50% correct on word lists. Catching 50% of the words, listeners can understand about 95% of conversational sentences, which means that at a party they can - with an occasional "what" - carry on a conversation perfectly well.

A second fact is we are all vocally lazy. When we speak in noise, we don't use enough effort to bring our voices well above the noise level. Pearson and his colleagues (1976) studied this phenomenon by putting a microphone and tape recorder on people and sending them out into homes, offices, hospitals, trains, cars, airplanes, and department stores. They found that once the background noise level exceeds about 60 dB SPL, almost all normal conversations take place at a signal-to-noise ratio between -5 dB and +5 dB. A few conversations even dropped below -5 dB signal-to-noise ratio, as shown in Fig. 1.

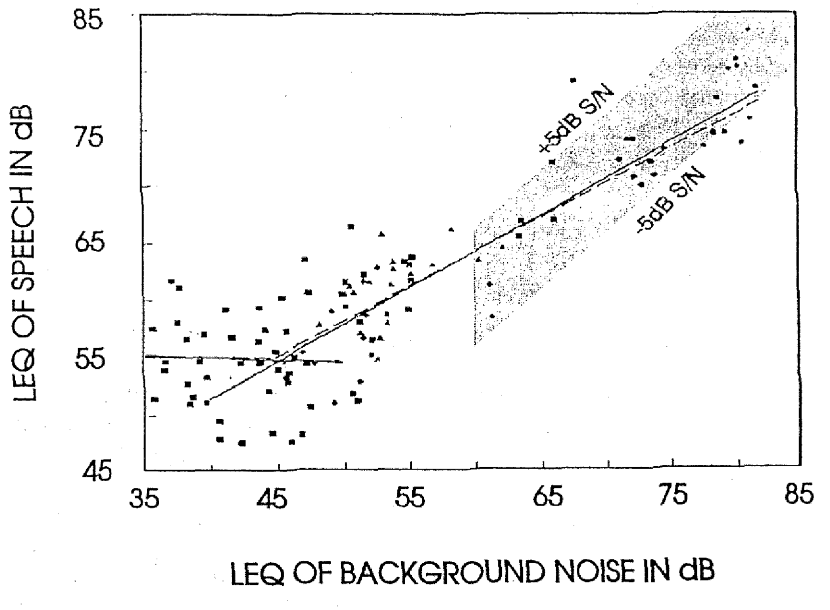


Fig. 1
Signal-to-noise ratio for conversational speech in homes, hospitals, department stores, trains and aircraft. (After Pearsons et al 1976)

So in daily life, once the noise goes up, we're routinely presented with the problem of trying to understand somebody whose vocal output is 5 dB below the sum of all the other people talking (or the other noise sources).

Surprisingly enough, normal-hearing individuals can cope with this task. Once you have a hearing loss, however, you're no longer able to carry on as normal. Back in the 1960's at Northwestern University where I now teach, Tillman, Carhart and Olsen performed a series of studies on this question. As reported in their 1970 summary paper, they found that the typical presbyacoustic required a 14 dB greater signal-to-noise ratio than their subjects with normal hearing. So while normal people can get by at a -5 dB signal-to-noise ratio, meaning they can get 50% of words and 95% of sentences correct, the hearing-impaired require a +9 dB signal-to-noise ratio. This places their abilities outside the range of normal conversation in noise. They simply can not participate, unless the others raise their voices well above the background noise.
Worse yet, Tillman et al found that when they added the best hearing aid they could find back in the 1960's, it made things worse. So much worse that their hearing-impaired subjects required a +27 dB signal-to-noise ratio before they could get 50% of word lists correct. Figure 2 summarizes the situation.

Was the hearing aid really to blame? Yes. In fact, when Tillman et al tested that same hearing aid on people with normal hearing, their normal-hearing subjects now required about a +7 dB signal-to-noise ratio for 50% correct. In other words, even subjects with normal hearing couldn't hear in noise with that hearing aid.

B. The Reasons

Why did hearing aids perform that way? Because they made only a fraction of the speech cues audible and clear.

Narrowband hearing aids were commonplace until recently. Figure 3 shows the estimated insertion response, calculated from the 2cc coupler curve that appeared in their paper, for the hearing aid that Tillman, Carhart and Olsen used in their studies. This hearing aid passed only a narrow band of speech cues on to the listener.

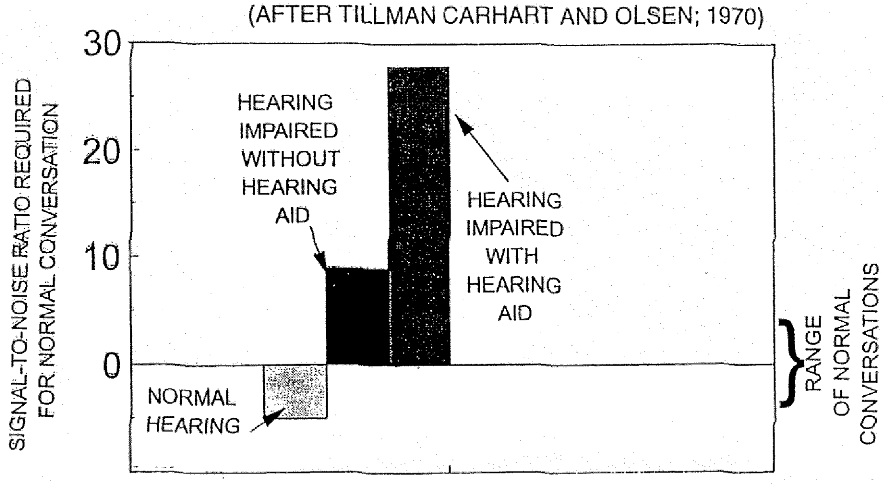


Fig. 2
Effect of 1960's hearing aid on S/N required for normal conversation.

Until recently, most hearing aids used so-called linear circuits, meaning nothing but peak clipping was used to reduce gain to prevent either discomfort or damaged eardrums, so that users were forced to turn the volume control down to avoid distortion or discomfort. This further reduced the audibility of speech cues.

And many hearing aids were designed so that the input stage would overload once the input SPL reached 85-90 dB SPL, so that even turning the volume control down wouldn't reduce distortion in high-level noise. Either at a sports event or a symphony concert, the hearing aid would routinely distort regardless of where the volume control was set.

Why were hearing aids made that way? The apparent reason is that there was a lot of good market evidence that said users did not want them any other way. Manufacturers tried extending the high-frequency response of hearing aids and users rejected them as being harsh and unpleasant. Manufacturers tried extending the low-frequency response of hearing aids, and users rejected them as being muddy and unintelligible. Manufacturers tried *automatic gain control* circuits and compression circuits and again users rejected them, saying they

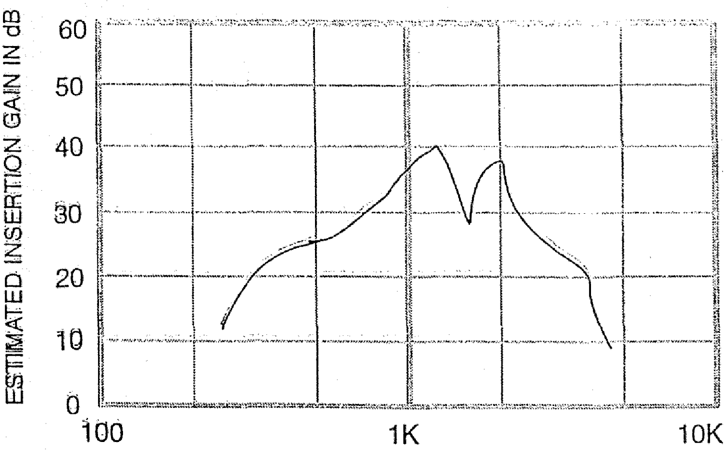


Fig 3.
Estimated insertion response of 1960's hearing aid used by Tillman, Carhart & Olsen (1970).

preferred linear hearing aids. (You can still hear that occasionally on the lecture circuit.)

You will get those answers every time if you start out with a low-fidelity hearing aid. Those are the right answers for low fidelity hearing aids. In fact, they make low fidelity hearing aids tolerable. The problem is you simply can not hear in noise with that type of hearing aid.

C. Amplifier distortion

Amplifier distortion reduces intelligibility in two ways. It reduces distortion directly because the harmonic distortion products and intermodulation distortion products mask the quieter speech cues. (Both types of distortion generate energy at frequencies not present in the input signal, i.e., they effectively generate noise. Severe distortion simply fills up the auditory space with this noise and buries many of the quieter, typically high-frequency, speech cues.)

Amplifier distortion reduces intelligibility indirectly, more subtly and insidiously, by generating all the wrong answers about what hearing aids should be like! It caused us to believe things that were absolutely false unless you were going to make low-fidelity high-distortion hearing aids.

I think you may have avoided some of this second problem here in Europe by concentrating on Behind-The-Ear (BTE) hearing aids where the larger batteries and larger-size Class B amplifiers were practical.
Once you go to In-The-Ear hearing aids, howevever, the marketing department checks what people in the marketplace are saying, and the users are saying "I'd rather have it small." Unfortunately, the users rarely hear a comparison between small-and-distorted vs. slightly-bigger-and-clear, so most companies went to class A amplifiers for their in-the-ear products. Class A amplifiers were the only things that would fit in the smaller in-the-ear and canal hearing aids. Once you make that class A decision, you are faced with another problem, which is battery life. In a canal aid using a 10A cell, for example, you need to bias the Class A amplifier at only a few tenths milliampere in order to obtain acceptable battery life. (A week is the generally accepted minimum). In order to prolong the battery life, you are forced to starve the Class A output amplifier of bias current.

At first glance, you would think your maximum output SPL would drop too low to be practical with such a "starved Class A" output, but there is a trick. You can keep the low-frequency undistorted output up as high as you need by asking Knowles to wind a receiver coil with very fine wire and a lot of turns, choosing one of the curves in Figure 4. You can even mostly maintain your SSPL 90 specifications with that tiny a battery drain. The hearing aid will sound plenty loud enough, because loud sounds will create overload distortion which makes things sound louder. The hearing aid may even sound louder because it distorts so badly. So using a high-impedance coil, you've achieved a low battery drain and no one will complain that your hearing aid isn't loud enough.

So starved Class A circuits became the rule. Unfortunately, these sacrifice nearly all of the undistorted high frequency output capabilities of the hearing aid. (Compare the lowest curve in Figure 4 to the higher ones). Thus, instead of having plenty of high frequency output capability to handle the high-frequency emphasis in most hearing aids, typically you find that you have nothing but intermodulation distortion for high-intensity, high-frequency speech sounds and noise.

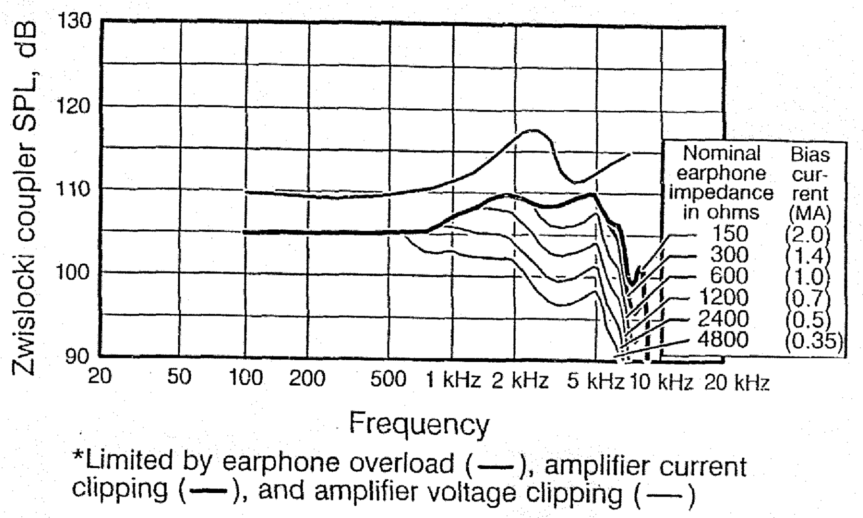


Fig 4.
Maximum undistorted output of Knowles BP-series earphone with well-damped conventional earmold. (After Killion, 1980).

D. The wrong answers explained

Why did we have all of these low-fidelity answers and low fidelity aids? Let me first just play a recorded reminder of what hearing aids used to sound like. (The listeners heard pre-recorded examples of a low-fidelity hearing aid at a symphony concert and at a night club. No one would want to wear such an aid personally, of course, but an embarrassing number of them have been sold.) You can <u>still</u> buy hearing aids that sound like that, and save about $100 an aid to boot.

How is it that we got the answers that no one wanted any more highs? I believe distortion, subtly, fathered this wrong answer. It was a replay of the 1940's, when high-fidelity console radios and phonographs first became available. Published studies showed that if you actually did listening tests, the man on the street wanted you to roll off the high frequencies starting at about 5 kHz. The engineers might like to listen to a full 15 kHz response, but to sell those consoles to the public you had to roll them off at 5 kHz.

Harry Olson, chief engineer at RCA in the U.S., became irritated enough with this answer that he set up a live listening test in which he had an orchestra behind a gauze screen so you couldn't see what was happening, and arranged organ louvers as shown in Figure 5 (Olson, 1957) to sealed off the orchestra stage. With the louvers closed, the sound of the orchestra would have been blocked off except Olson drilled holes through each louver to form an acoustic 5 kHz low pass filter. (The mass of air in the properly-sized holes with the compliance of air in the space between the louvers gave the acoustic filter whose equivalent analog and frequency response are shown in Figure 5). Olson set up live listening tests with both a live orchestra and live voice and with real listeners in the room, and he found two-thirds to three-fourths of the subjects preferred the full-frequency-response reproduction.

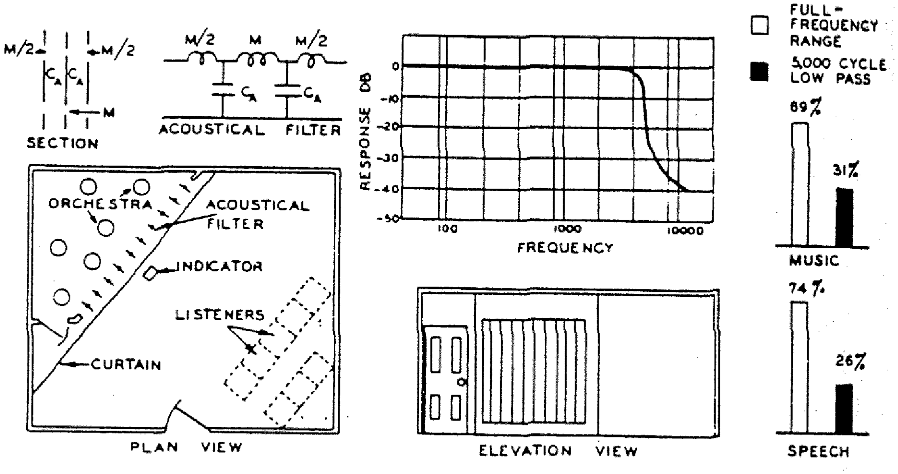


Fig 5.
Setup for and results of Olson listening-test experiment.

Olson went on with a series of experiments to demonstrate that the real problem was not bandwidth but amplifier distortion. The results are shown in Figure 6.

The reason listeners preferred the 5 kHz roll off in previous studies was that you needed to roll off the response above 5 kHz if you were going to allow 8-10% amplifier distortion. With that kind of amplifier distortion, the sound was objectionable unless you rolled it off above 5 kHz. If you cleaned up the amplifier and got the distortion below 1.5% distortion, on the other hand, then you could go the full 15 kHz. In fact, people actually preferred the full frequency response as long as distortion was low.

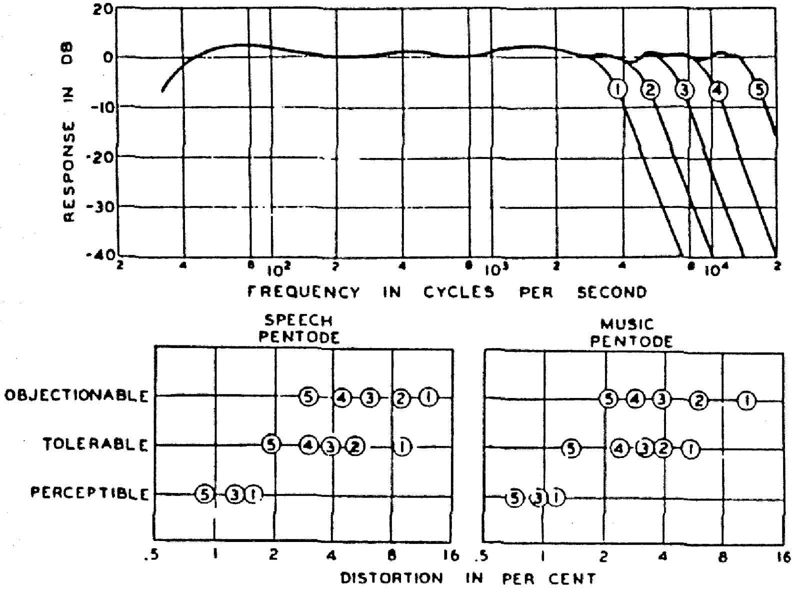


Fig 6.
Explanation of results of Olson's predecessors: amplifier distortion.

I think exactly the same thing happened with hearing aids. As long as clipping is the means of limiting the output, as has been true for most hearing aids purchased for several decades, it means that every time things got reasonable loud, you heard nothing but distortion products at high frequencies.

Once you have a lot of distortion, the higher-frequency distortion products themselves become more intense than the high-frequency portions of speech and music (which are typically weaker than the low-frequency portions). If you extend the bandwidth, all you hear in that extended bandwidth is the noise of the higher-frequency distortion products. The lower-level speech and music signals in the extended bandwidth are not audible because they are covered up by the more intense distortion products. So you can extend the bandwidth of the hearing aid, but the user reasonably says, "That's doesn't sound like an improvement to me, I'd rather listen to it the other way."

Figure 7 shows the insertion response of two hearing aids. The wide-band hearing aid actually has a better amplifier, a cleaner amplifier than the narrow-band hearing aid. Everyone who hears the comparison, however, agrees that they would rather listen to the narrow-band hearing aid. The listener would much rather have those distortions products filtered out. (Recorded comparison at a symphony concert.) Neither of the two hearing aids sounds great in overload, but the narrow-band aid is easier to listen to. Obviously under these circumstances the right thing to do is to roll off the highs! There are no useful high-frequency speech cues in a wideband hearing aid when distortion covers them up.

What about the belief that we should always roll off the lows? I think the same explanation holds as for rolling off the highs: distortion. Back in the 40's it was well known that if you were going to use clipping as a means of limiting the output, you needed to use high frequency emphasis before you clipped if you want to retain intelligibility. Licklider, Pollack and others (1948), showed in their experiments that you could use infinite peak clipping (nothing but square waves coming out) as long as you first rolled off all the low-frequency sounds. Even the Harvard Report (Davis et al, 1946) said; "(With low-frequency emphasis) the scores fell rapidly toward zero when the input to the limiting stage was increased above the clipping threshold. Intelligibility was best maintained with 6 dB/octave upward tilt."

In starved Class A hearing aids, clipping distortion was and is the rule. The way they are designed, they are distorted whenever it gets loud. You <u>must</u> roll off the lows or else destroy intelligibility.

So we had very good evidence, accurate evidence, that said if you wanted any-one to <u>buy</u> your hearing aid you had to roll off the highs and you had to roll off the lows. And it was a result of using linear circuits and, worse yet, starved Class A circuits.

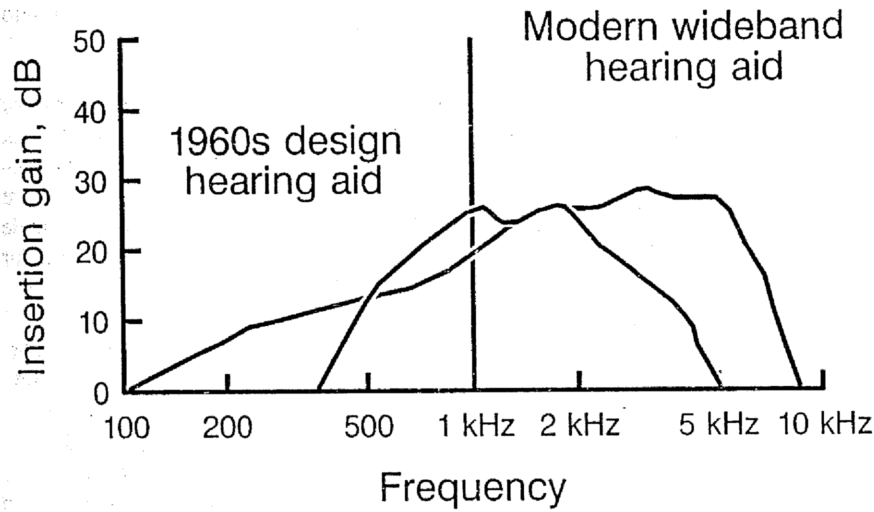


Fig 7.
Frequency response of two hearing aids. The narrowband response survives clipping distortion much better. (From Killion, 1984).

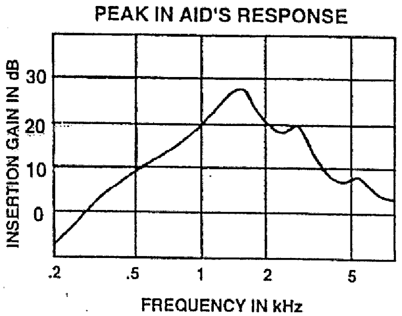
A technical aside: It is important to note that there is nothing wrong with Class A amplifiers per se. If you use enough bias current in a class A amplifier you can make it as clean as you want, just as clean as a Class B or Class D amplifier. With the right receiver coil, you can also have plenty of undistorted high-frequency output. It is not Class A that is the problem, but the fact that a good Class A amplifier takes so much current that the battery dies quickly. Thus most designers are forced to starve the Class A amplifiers to obtain reasonable battery life, resulting in the exaggerated high-frequency distortion problems we've been discussing.

> A well-designed Class B amplifier takes only about twice the battery drain of a Class D amplifier, so a much more reasonable compromise can be made. The main problem with Class B amplifiers historically was that they were usually too big to fit into the smaller hearing aids.

There is one other problem that is also a little subtle, and that is the loss of audible speech cues caused by a peak in the frequency response. A large peak will cause the hearing aid to fail what I call the DHTTDGTLWI (Dit-ta-dug-Attle-we) test, which stands for "Doesn't Have To Turn Down Gain To Live With It." I want to play a simulation of what it sounds like, a recording made using the electronic hearing-loss-simulation equipment described by Villchur (1977).

The hearing loss simulated is of someone with a severe impairment. That individual's loudness growth function hasn't reached normal until it rises into the discomfort region, as seen in Figure 8. The loss may be unusually severe, but the real-ear frequency response shown in Figure 8 is typical of almost every ITE hearing aid produced for about two decades. Those aids all used the BK series receivers that I helped design back in 1964. We didn't know how bad the real-ear high frequency roll off was back then, because we didn't have probe microphone measures. We knew you had to roll off the lows however, so we rolled off the lows. We rarely used damping back then, so the standard ITE insertion response showed a large peak at the 1700 Hz resonance of the BK-series receiver, with a roll off above and below that frequency. You can measure hearing aid after hearing aid made during that period, and this is roughly the real-ear frequency response you'll see, with perhaps some variation in low-frequency slope.

Well, here's what it sounds like to have a severe impairment and have to listen through that frequency response. (Demonstration plays) What you heard was that every time a vowel format frequency coincides with that 1700 Hz peak frequency, you get whacked on the ear. It hurt, and your automatic reaction was almost certainly to turn the gain down. And that's what users did. They turned down the gain, and then all that remained audible was the small region of speech information right around 1700 Hz. So once again, you had a hearing aid with which you could not hear in noise.



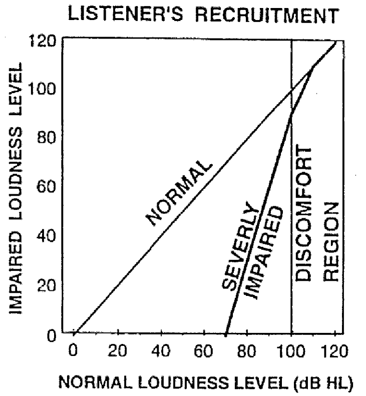


Fig 8.
Villchur's electronic simulation of severely impaired listener subjected to typical 1960/70's ITE hearing aid.

Fortunately, we now have solid evidence as to just what happens to someone with a hearing loss using the sort of narrowband hearing aid we have been describing. Skinner and Miller (1983) did a series of experiments where they looked at the effect of bandwidth on word recognition scores in quiet and noise. Their listeners had moderate to severe loss. (There was plenty of evidence for normal listeners that you need wide bandwidth, but for many years we thought that perhaps it would be more merciful to the people with moderate-to-severe hearing loss to limit the bandwidth and not cause confusion. I can remember Hugh Knowles making basically that argument back in the 1960's.) What Skinner & Miller found was that all of their listeners did best with the widest audible bandwidth. (Note added in formalizing the manuscript: Villchur (1973) reported an example of a subject who had no useful hearing above 3 kHz, and who did better with the high-frequency response of her experimental hearing aid rolled off. During audiometric testing, if asked, such subjects may report that high-frequency tones sound like a buzz or hum. [We often forget to ask.] Obviously these subjects need to be excluded from the generalizations above. Fortunately, they are rare.)

Figure 9 illustrates Skinner & Miller's results for listener #6, who could get 48% correct in noise with the proper wideband frequency response. Getting 48% of words typically means 95% of sentences; Listener #6 could carry on a more or less normal conversation in that noise with the full bandwidth response. When tested with a bandwidth comparable to the bandwidth of hearing aids such as the one in Figure 8, Listener #6 got only 4% correct in noise.

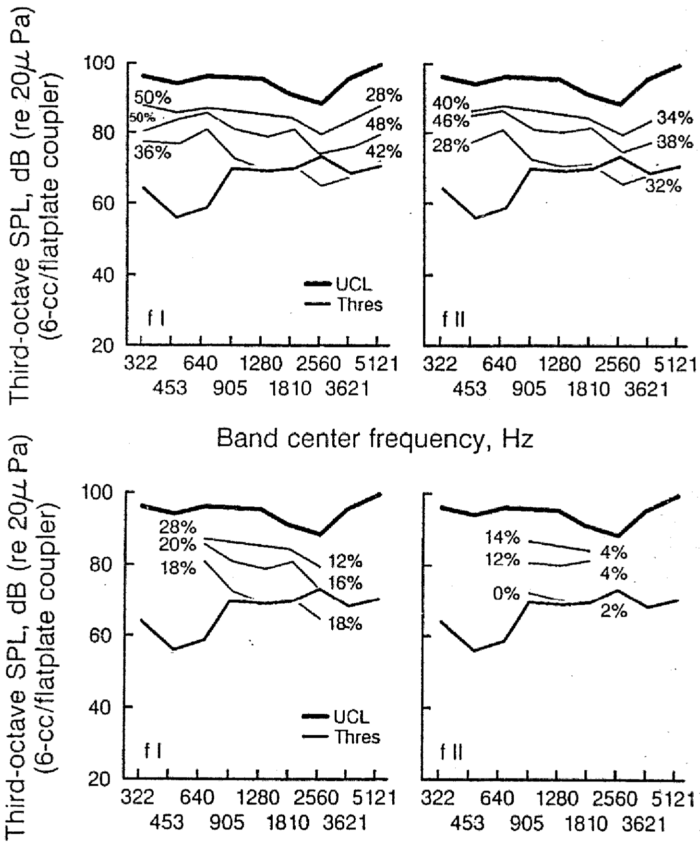


Fig 9.
Effect of bandwidth on the ability of Listener 6 to understand speech in noise. Word recognition scores given in percent for speech in quiet (left numbers) and noise (right numbers). (After Skinner & Miller 1983)

Skinner & Miller's listener #1 shown in Figure 10 did somewhat better, dropping from 66% correct with a full bandwidth response to 26% correct with a narrow bandwidth. Even so, listener #1 would be unable to carry on a conversation in noise with the typical hearing aid of the last two decades. Taking away some of the redundant speech cues leaves these listeners with insufficient cues to understand speech in noise.

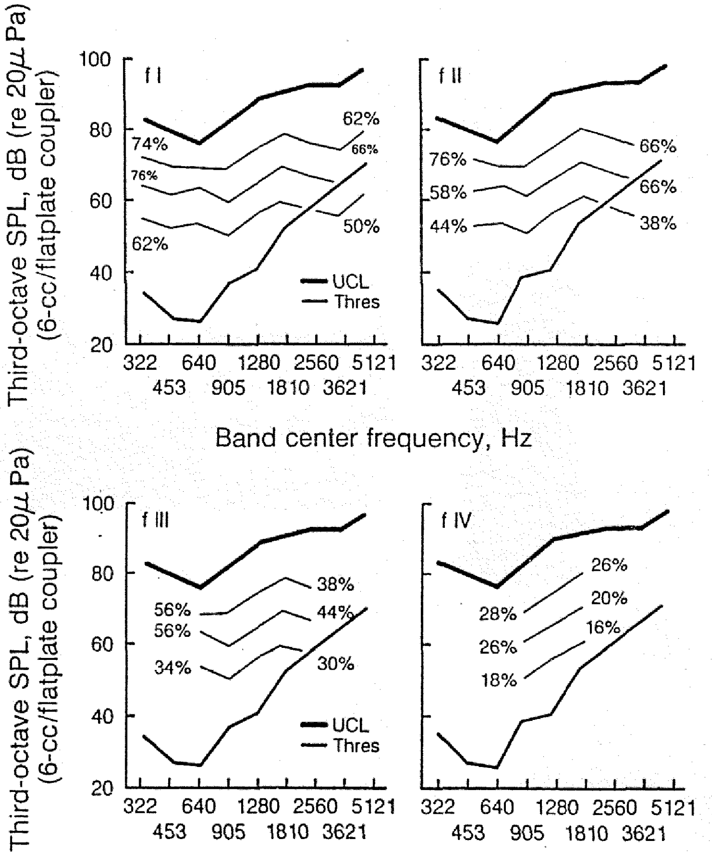


Fig 10.
Effect of bandwidth on the ability of Listener 1 to understand speech in noise. Labels same as Fig 9. (After Skinner & Miller 1983)

With all of this hindsight, it seems obvious why hearing aids didn't work in noise. That in turn is almost certainly the reason your family doctor - whose mother had a hearing aid that didn't work in noise - told everybody with "nerve deafness" not to get a hearing aid because it wouldn't help in noise. He was right, it didn't.

But you can get totally different answers to all of these questions if you start out with high fidelity hearing aid designs and use appropriate automatic adjustments to the gain and frequency response in order to keep all speech cues audible.

II. High fidelity is possible in hearing aids.

For years, no one thought high fidelity was possible. Ironically enough, high fidelity has been possible in hearing aids for some 15 years. Twenty years ago, broadcast and recording studios were already using a flat-frequency-response version of one of the Knowles hearing aid microphones I helped design, Hearing aid receivers capable of a 16 kHz bandwidth have been available for almost that long: The Knowles BP-series receivers were introduced in the 1970's.

The capability of those receivers is so good that we (Etymotic Research) have just introduced to the hi-fi market a pair of stereo earphones we call the ER-4, earphones which I believe are the highest-fidelity earphones available, with the most accurate real-ear response around. But these earphones use the same basic Knowles ED-series receiver mechanism used in the K-Amp hearing aid designs.

My Ph.D. thesis back in 1979 was basically a demonstration that you could make a high-fidelity hearing aid, at least as judged by someone with normal hearing (Killion, 1979). Figure 11 shows the 16 kHz frequency response of an experimental in-the-ear aid used in that study. I prepared KEMAR recordings of loud speakers, ear phones and experimental hearing aids. These were judged by one of three groups of listeners: a "man on the street" panel consisting of 12 men and 12 women (average people, no engineers), a panel of 5 golden ears (people like Julian Hirsch, who does stereo equipment rating for the magazine Stereo Review in the U.S.), and 6 trained listeners (people who have spent hours and hours listening to some type of psycho-acoustic experiments).

All three panels gave basically the same answers, as shown in Figure 12. Indeed, so did an unreported panel consisting of 9 of my relatives who were

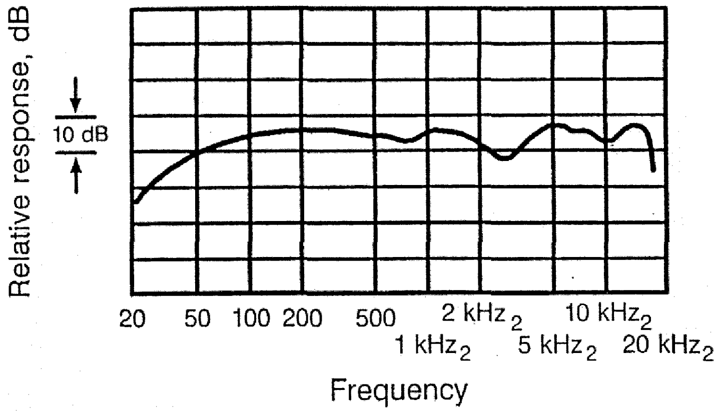


Fig 11.
Experimental high-fidelity hearing aid with 16 kHz bandwidth. (From Killion, 1979)

forced to "take the listening test" as they visited throughout one summer. So the answer was pretty solid: any reasonable group of people with normal hearing would judge the experimental high fidelity hearing aids to be slightly better (but not statistically significantly so) than the expensive pair of high-fidelity stereo loudspeakers which had been chosen because they were the most popular recording-studio monitor speaker used in Chicago at the time.

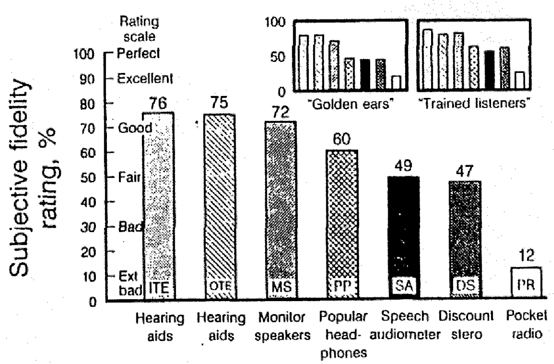
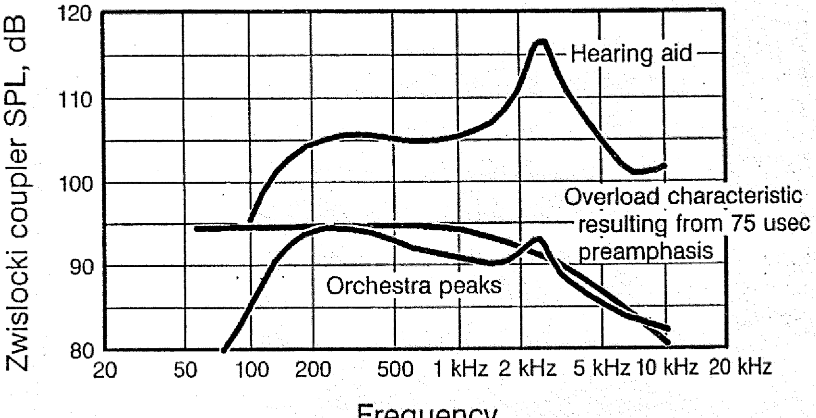


Fig 12.
Fidelity ratings from three listening panels: Man-on-the-street (12 male, 12 female), Golden Ears, and Trained Listeners. (From Killion, 1979)

There was one practical problem in rushing into production with this hearing aid design. I had used 5 mA of battery drain in my class A output stage in order to provide the output capabilities needed to handle a live symphony orchestra concert. Figure 13 indicates why that much current drain was needed if a hearing aid is to handle a live orchestra concert without distortion. I can verify from my own measurements that the Chicago Symphony Orchestra can produce 104 to 106 dB SPL readings on a sound level meter located at a seventh-row, first-balcony seat in Orchestra Hall. This means that the instantaneous peaks run something like 115-118 dB. Now if you want the hearing aid to never, never distort, the hearing aid has to be capable of those outputs; which means you can't even think of using a starved class A output amplifier.



Orchestra peaks correspond to instantaneous peaks of 105 dB SPL or peak readings of 95 dB SPL on sound level meter

Fig 13
Undistorted SPL output requirements for high fidelity hearing aid.

You have to feed a class A amplifier very well if you want it to handle those outputs. "Very well" in this case means 5 mA of battery drain. But 5 mA from a 10A cell in a canal aid would drain it completely in 10 hours: Install a fresh battery in the morning and it would need to be replaced at dinner time. You would get a lot of complaints if you tried to sell hearing aids that ate batteries like that.

The breakthrough solution to that problem was a new amplifier technology. Elmer Carlson got me interested in Class D amplifiers back in 1964, and I built a fist-size 7 watt amplifier that year. That first amplifier would obviously not fit in a hearing aid, and the hearing-aid version became a project that was pursued and canceled several times. Finally, after I left Knowles Electronics to form my own company, Hugh Knowles gave me one more chance to see if we could make a practical low-battery-drain high-fidelity Class D hearing aid amplifier. With a lot of help from our friends, we finally succeeded. The resulting CMOS integrated-circuit chip is now in all of the Knowles class D receivers. Figure 14 is a photomicrograph of that chip (whose actual size would drop nicely in the "u" in the word actual.) Please notice the mustache at the top of the chip.

This new chip makes it possible to have a hearing aid output amplifier which will idle at .17 mA bias, and yet provide the undistorted output of a class A amplifier biased at several mA's. Pleasantly enough, the bonus is that if you use the Class D receiver with a low-current preamplifier such as the K-AMP circuit, the entire hearing aid only drains about .35 mA. Thus, with less battery drain than the starved Class A output amplifiers consumed by themselves, you can have a complete hearing aid which gives 3-4 weeks of 13A battery life and which never overloads - even if you go to a Chicago Symphony concert. You can "sit in" on piano with a Dixie Land band - with the drummer only a few feet away - and the hearing aid simply does not overload. (I'm assuming that the volume control has been set to produce unity gain for loud sounds.)

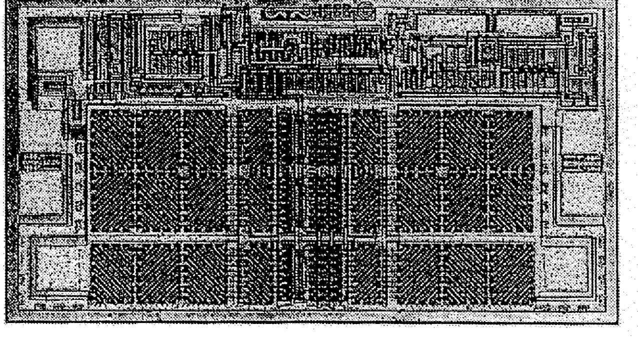


Fig 14
Enlarged view of Knowles Class D amplifier chip.

The performance advantage of the new Class D receiver was obvious. The big question was cost. Adding the Class D amplifier roughly doubled the cost of the receiver, which meant adding about $50 to the wholesale price of the hearing aid. The obvious question everyone asked was: "Is anyone willing to pay for that?" One way to answer that question was to do a research study. Catherine Palmer (1993) did just such a study under Knowles sponsorship. She compared the sound of two hearing aids, each set to 15 dB gain, one using a starved class A amplifier (biased at 0.4 mA), and the other using a class D receiver. She made KEMAR-based recordings using three different source materials (orchestra, piano trio, and male voice), and using a range of input SPL's. She played them back in a randomized form to the listener-judges, giving them roughly the following instructions: Assume you need a hearing aid; assume you are ready to buy a hearing aid and have enough money to buy a hearing aid, and that hearing aids cost at most $700. What would you pay for a hearing aid that sounded like -. The listeners circled, on a ruler going from $0 to $700, what they would pay. She also used a quality scale, asking the listeners to rate, on a 0 to 100% scale, the overall quality of the sound they had heard.

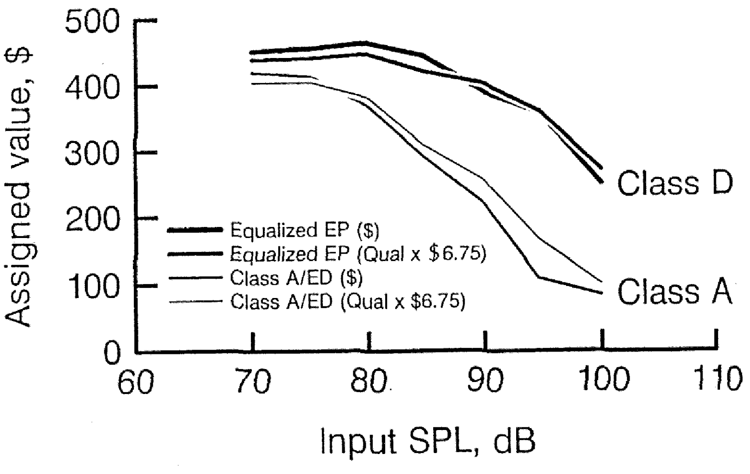


Fig 15
Value judgments for hearing aids using Class D vs. "Starved Class A" power amplifiers.

Figure 15 shows Palmer's results. The laboratory answer to the question of whether or not anybody was willing to pay substantially more for the Class D hearing aid was clearly yes. Listeners indicated they would pay $100 to $200 more over a wide range of input SPL's. Just as interesting, and as far as I know for the first time in the history of the world, Palmer was able to quantify what quality is worth. She found that if she multiplied each quality point by $6.75, she obtained transformed quality-judgement curves nearly identical to the dollar curves. So we now know what quality is worth; it's worth $6.75 a percentage point. That is a pretty high and exciting number. It justifies some improvements in hearing aids that we used to think were not justifiable.

III. High fidelity for the hearing impaired.

Where are we? We have wide bandwidth, we have a smooth frequency response, we have experimental hearing aids that people with normal hearing judge as high fidelity, we have a low battery drain, and we have no audible distortion. We're up to the crucial question that Hugh Knowles asked me some 15 years ago: what does high fidelity mean for someone with a hearing impairment? We still haven't solved the real problem: providing a high-fidelity hearing aid for someone with a hearing impairment. What do you do now?

A. Recruitment.

There is some reason to think we can answer that question. If we consider the loudness growth curves shown in Figure 16, we see that someone with a mild-to-moderate cochlear hearing loss shows normal loudness perception for high-intensity sounds. These cochlear losses are usually due to a loss of outer hair cell function. This is nice because at higher levels we see almost complete recruitment; at higher levels the outer hair cells probably don't affect the motion of the basilar membrane anyway. Complete recruitment means normal loudness perception, and it typically covers a large region of moderate- and high-level input sounds. Not only normal loudness is obtained, but in many cases normal psycho-acoustic tuning curves, normal difference limens for frequency, etc. You can do all sorts of tests, and as long as you have only a mild loss, perhaps a mild-moderate loss, you can get normal function over quite a region of high-level sounds.

I personally believe that one bad definition in psychoacoustics has set back hearing aid design enormously. We learned in school that recruitment was an abnormal growth of loudness. This may be true in a very narrow sense, but it provides exactly the wrong perspective on the problem to be solved. Recruitment is not an abnormal growth of anything, but an abnormal loss of sensitivity for quiet sounds. (Indeed, we now have solid physiological evidence that such is exactly the case: the loss of outer hair cells in the cochlea produces a loss of sensitivity for quiet sounds). If you sit mired in the region of sensitivity loss and look at the problem, the obvious solution is to provide gain. And then having done that you go home. But you've left the hearing aid wearer with that gain for all input levels, because the only thing you see to solve is the loss of sensitivity.

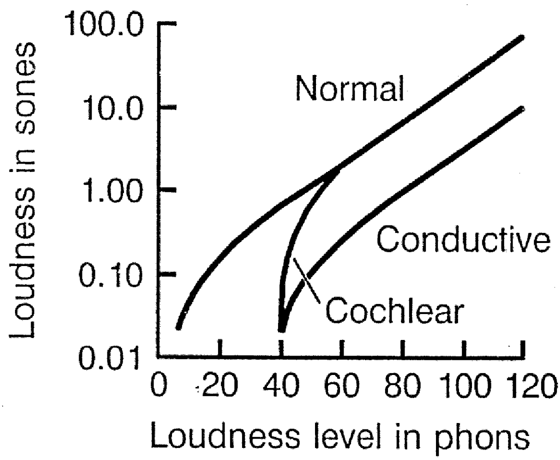


Fig 16.
Loudness function for 1000 Hz tone. (After Steinberg & Gardner, 1937).

If you view the problem from the high-level region where recruitment is complete, you see that you don't want any gain - you don't even want a hearing aid - for much of this range of input. This is a region in which you have normal hearing, and you see that nothing is growing abnormally as long as you don't wear a hearing aid that gives you gain you don't need. Viewed from the input sound levels where the individual mostly lives, the problem is an abnormal loss of sensitivity for quiet sound. It's the equivalent of night blindness: You don't have any trouble when it's bright, you have just lost the ability to see in the dark.

Well if you look at the problem from the standpoint of high-level-inputs, for those inputs you want the hearing aid to do nothing. The hearing aid should not stand in the way of your normal hearing, and it certainly shouldn't provide amplification for things that are already loud. Down in the low-level-input region, on the other hand, the person needs some gain or otherwise he or she wouldn't have come in to your office. They came because they can't hear quiet sounds.

B. K-AMP Processing.

The presumed ideal answer to the overall problem is the hearing aid characteristic shown in Figure 17 (For identical reasons, Barfod (1976) recommended an essentially similar characteristic.)

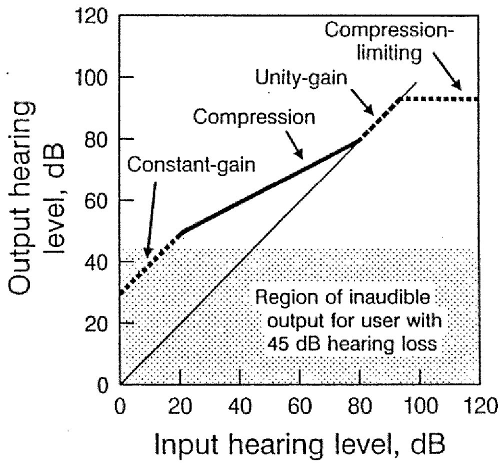


Fig 17.
Presumed ideal input-output function for a hearing aid intended for an individual who has relatively normal high-level hearing.

For high-level sounds it gives unity gain. It's as close as we can come to having the hearing aid sit on your shoulders, out of your way for loud sounds, and then when a quiet sound comes along having it jump into your ears to give you the gain you need. For high-level inputs (80 dB HL and above) the hearing aid should be as transparent as we can make it. For very-low-level inputs (0-20 dB HL), it should produce the gain you need for quiet sounds, to solve the problem that brought you in to get help.

The only remaining question is, how do you get from the high-level region to the very-low-level region. You certainly don't want to suddenly switch gain at some level. (That was one of the problems with the Intellitec circuit. If you've ever listened to it, it would suddenly switch 15 dB of gain in and out in a given channel.) No, for the least obtrusive operation, the highest subjective fidelity, you want to change gain as gradually as you can. Which means wide dynamic range compression (sometimes called logarithmic compression). This makes it possible to gradually change from 25 dB of gain for quiet sounds to 0 dB of gain for loud sounds, spreading the gain change out over a wide range (over 50 dB in the case of the K-AMP circuit).

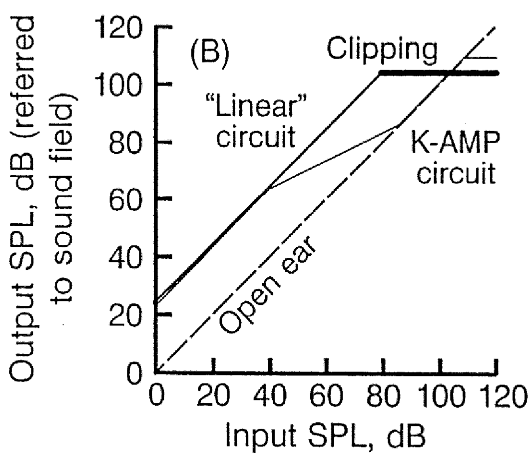


Fig 18.
Input-output curves illustrating two approaches to providing a 25 dB threshold improvement.

Figure 18 shows a comparison between the input-output characteristics of the K-AMP circuit and a linear circuit. The linear circuit operates perfectly well for very quiet sounds, but it provides progressively too much gain for louder sounds until - mercifully - it starts clipping at about 80 dB SPL input. But clipping means it's distorting. So, a linear circuit works fine in quiet, but the rest of the time it's pretty bad: Too much gain for moderately loud sounds and you can't hear anything in noise above 80 dB.

Everyone knows that several research studies confirm that linear is just as good as compression ... if the user can adjust the volume and tone controls fast enough! In a given situation, it is nearly always possible to adjust the volume and tone control so that you score as well with a linear aid as with a compression aid on a word list, especially if the word list is a standard constant-presentation-level list (which Villchur calls a "precompressed" word list).

In the real world, however, you would have to devote your entire energy to that task in order to keep the volume and tone controls at optimum with normally-varying input conditions. No one does that, so you forced to make a choice if you're dispensing linear hearing aids: You can either use clipping to keep the thing from hurting someone's eardrums or becoming uncomfortably loud, or you can use so much power that the hearing aid never clips. The latter hearing aid is the type that Darrell Rose loves to talk about, one that will reduce your IQ 10 points in 3 minutes. Those are your choices with linear aids. You're either going to clip and destroy intelligibility, or you're going to have so much power that you're going to be in trouble with discomfort with loud sounds.

Figure 19 shows the comparison of linear and K-AMP circuits in terms of gain. As someone said, "I sell gain, not output." Both have the required 25 dB of gain for quiet sounds. The difference between the linear and K-AMP circuits is that the linear circuit only has the right amount of gain for quiet sounds, too much gain for louder sounds and then it mercifully goes into clipping for loudest sounds. The K-AMP circuit has the same gain for quiet sounds but gradually turns down the gain to unity or zero dB gain for loud sounds, and doesn't clip until inputs of 110 or 115 dB SPL. Unless you work near jet aircraft, it never clips.

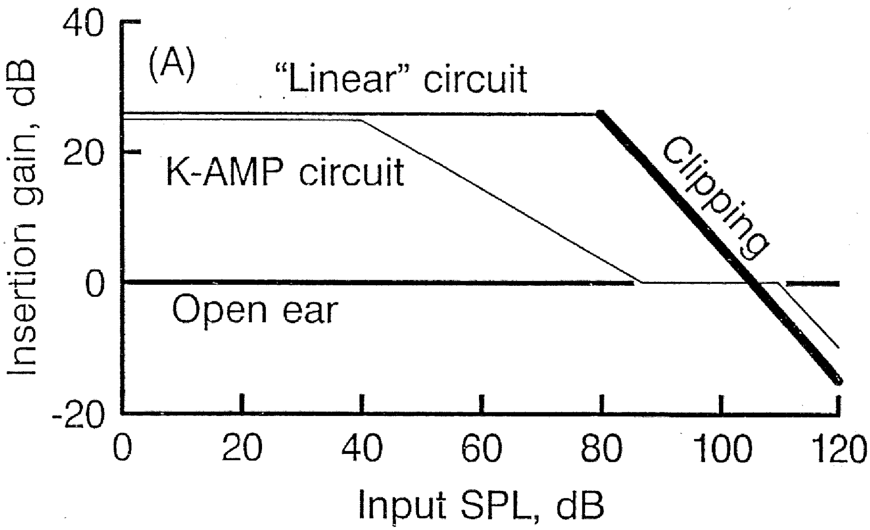


Fig 19.
Gain-vs.-input curves illustrating two approaches to providing a 25 dB threshold improvement.

Figure 20 shows the 2cc coupler curves at various input levels for the K-AMP circuit with the FFR (Fixed Frequency Response) option. Here you have about 25 dB of gain for 40 dB SPL and lower. As you increase the input, the gain keeps dropping (although the frequency response doesn't change) until for 90 dB SPL input you have a 2cc coupler curve that corresponds to about 0 dB real-ear insertion response (about -3 1/2 dB gain at low frequencies on a 2cc coupler).

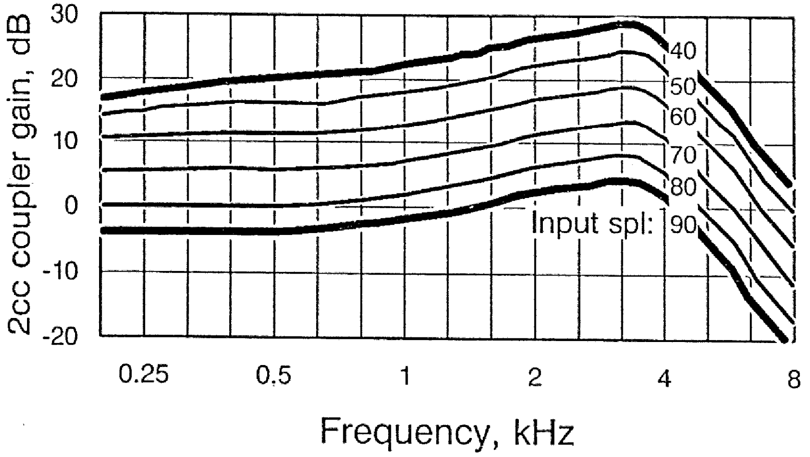


Fig 20.
2cc coupler curves of "FFR" (Fixed Frequency Response) version of K-AMP hearing aid. Volume control adjusted to produce 0 dB REIR for loud sounds.

This FFR characteristic works very well for the flat losses shown in Figure 21 and - surprisingly enough - it also works very well for reverse slope losses such as indicated in Figure 21. Several patients with reverse slope losses have said that their K-AMP hearing aids were by far the best thing they've ever had. (It's possible to wire the K-Amp circuit for a "BILL" response to handle this a little better, but we haven't gotten around to trying it because everyone with this loss has been so happy with the FFR response.)

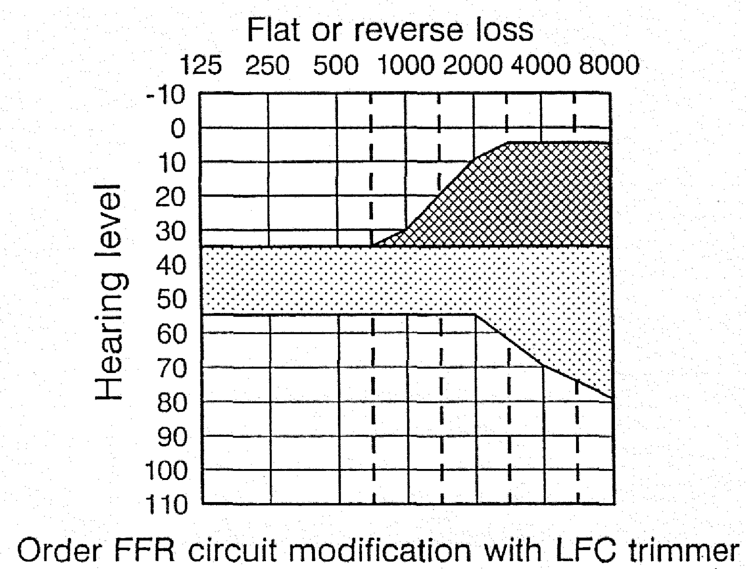


Fig. 21.
Low-risk fittings for K-AMP hearing aids: flat and reserve-slope losses having high probability of a successful fitting without special effort.

C. High-frequency loss.

The FFR characteristic left us without a solution for the more common problem, which is someone with a high-frequency hearing loss. The loss shown in Figure 22 is frequency dependent. At low frequencies we see very little loss, little need for amplification. But we see a substantial loss at high frequencies.

Now if you believe Barfod's data, the individual with the audiogram of Figure 22 has normal or near-normal hearing at all frequencies once you make things loud enough. This means that a flat frequency response for very loud sounds makes sense, but some high-frequency emphasis is required for quiet sounds.

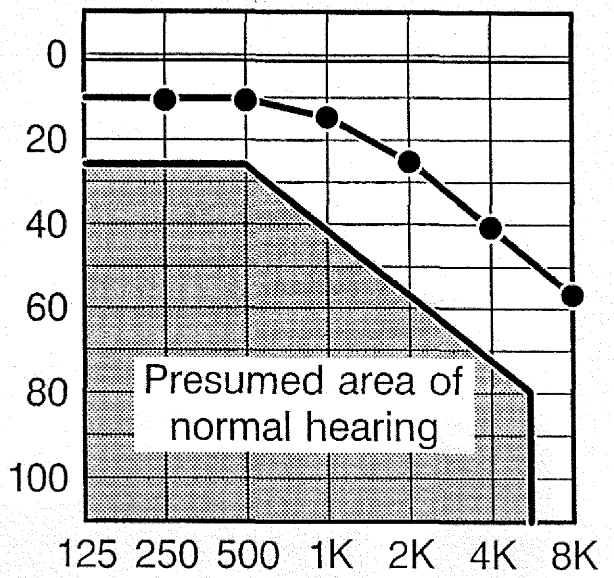


Fig 22.
High-frequency loss with region of normal loudness calculated from Barfod (1976) data.

Fortunately, Margo Skinner (1980) studied exactly that question. She had subjects with normal or near-normal low-frequency hearing but with substantial high frequency loss, and she tested these subjects with the 5 different frequency responses shown in Figure 23 using a wide range of sound pressure levels. She did an enormous amount of testing (she complained that she only had about 25 hours of word responses from each subject!), and finally obtained definitive answers. Figure 24 shows data for one subject tested with two of Skinner's frequency responses. (The rest of the curves pretty much fill in between.) Skinner found, not surprisingly, that for quiet sounds you got a much higher word recognition score if you gave a lot of high frequency emphasis. But, by the time you got up to 55 dB input with that amount of high-frequency gain, the subjects said ouch, they could not take it any more, it was just too loud.

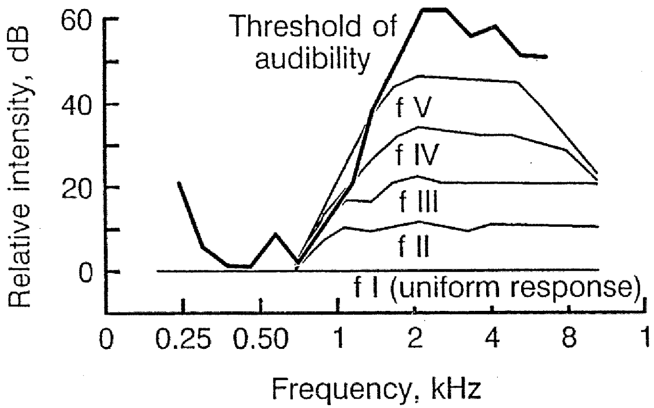


Fig 23.
Frequency responses used in Skinner (1980) study.

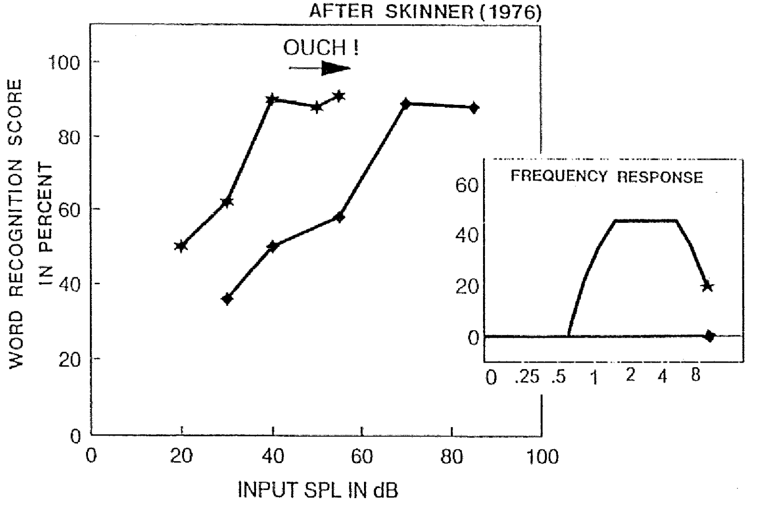


Fig 24.
One subject's word-recognition scores when tested with the two frequency responses shown inset. After Skinner (1980).

On the other hand, for very loud sounds, Skinner found a flat frequency response gave about as good word recognition scores. And a flat frequency response for loud sounds makes some sense, because all the speech cues still fit in the auditory area. A flat response for loud sounds generally sounds much better, also, especially on music.

D. K-AMP LEVEL-DEPENDENT FREQUENCY RESPONSE

So quiet sounds need high-frequency emphasis while loud sounds need a flat frequency response. What is obviously needed is a level-dependent frequency response modification to that FFR curve that we had, one which will provide more gain and treble boost for quiet sounds than for loud sounds, and will in fact give a flat frequency response and no gain at all for loud sounds. Such a set of curves is shown in Figure 25, illustrating the level-dependent gain characteristic of the K-AMP circuit.

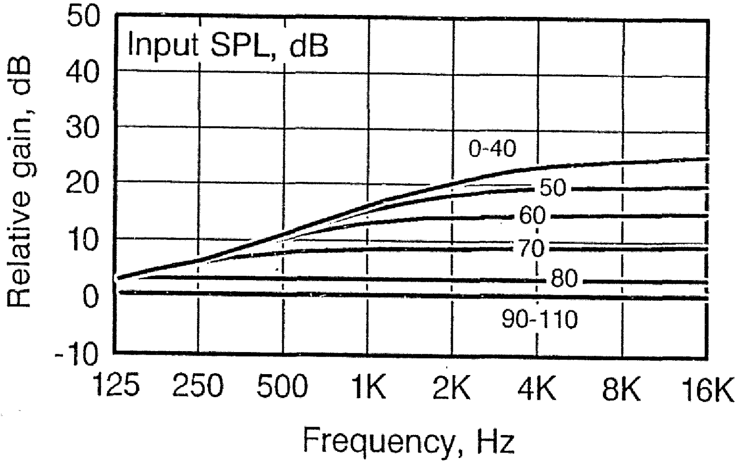


Fig 25.
Relative gain and frequency response of the K-AMP circuit for different input levels.

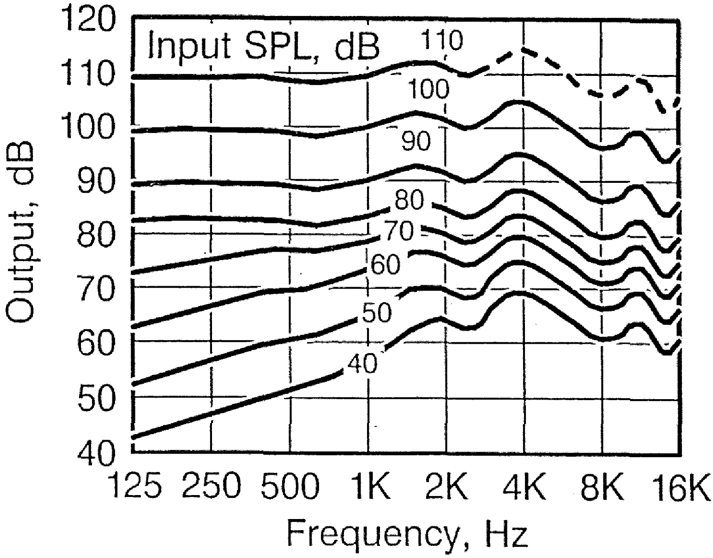


Fig 26.
Real-ear output curves for complete K-AMP hearing aid with 40 to 110 dB SPL inputs.

Figure 26 shows the real-ear output curves for a complete hearing aid using that K-AMP circuit. The volume control has been set to about 1/3, and an input of 90 dB SPL produces a 90 dB SPL output (give or take a little because the real-ear insertion response isn't perfectly flat). And 100 dB in gives 100 dB out; 110 dB in gives 110 dB out. (If speech and music had a flat spectrum, a 110 dB input would cause overload distortion at high frequencies. As it turns out, the spectrum of speech and music rolls off at high frequencies, so an overall input of 110 dB doesn't cause audible distortion. When the Chicago Symphony hits 105 dB, a lot of the energy is due to the kettle drum; the frequency band above 1500 Hz band typically has less than 95 dB SPL in it.)

In Figure 26 note that when you get down to quiet sounds, 40 dB in does not produce 40 dB out. Instead, 40 dB in receives gain and a treble boost, coming out something like 70 dB at high frequencies. So you have the high-frequency gain you need for quiet sounds with a flat, zero-gain operation for loud sounds.

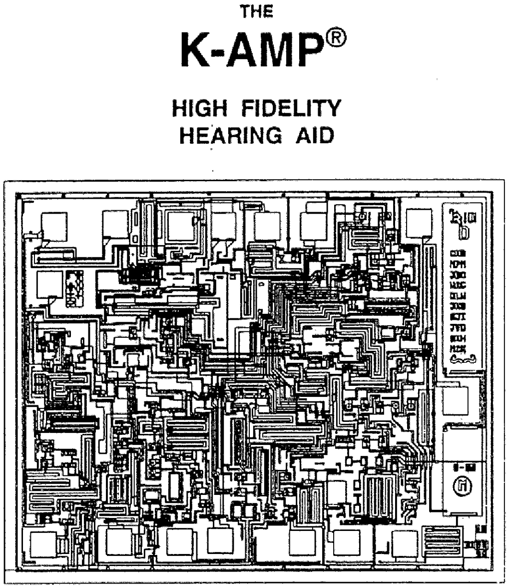


Fig 27.
Poster showing enlarged view of K-AMP integrated circuit chip.

Figure 27 shows the integrated circuit chip that goes into the K-AMP hearing aid. You'll note that it also has a mustache on it, along with the initials of the 9 other technical people who made this chip possible. (Posters of this chip are available incidentally, and if you write or phone Etymotic Research we will send you a couple. Our fantasy is that every dispensing office in the world will soon have this poster prominently displayed.)

The K-AMP chip can't work by itself, but needs to be mounted to a ceramic "hybrid" circuit that contains the necessary capacitors and has solder pads so that it can be wired into a hearing aid. Figure 28 shows the three K-AMP hybrids that have been produced to date. The first one, the 19D, was designed by one of our fellows, Jonathan Stewart; the second one, the 24D, by Andy Joder and Chris Conger of Rexton (who correctly believed they could make a much smaller one), and the even smaller third one, the 28D, by Tony Becker of TEC. The 28D uses a buried capacitor substrate (BCS) construction where all of the capacitors are contained in one monolithic block of ceramic.

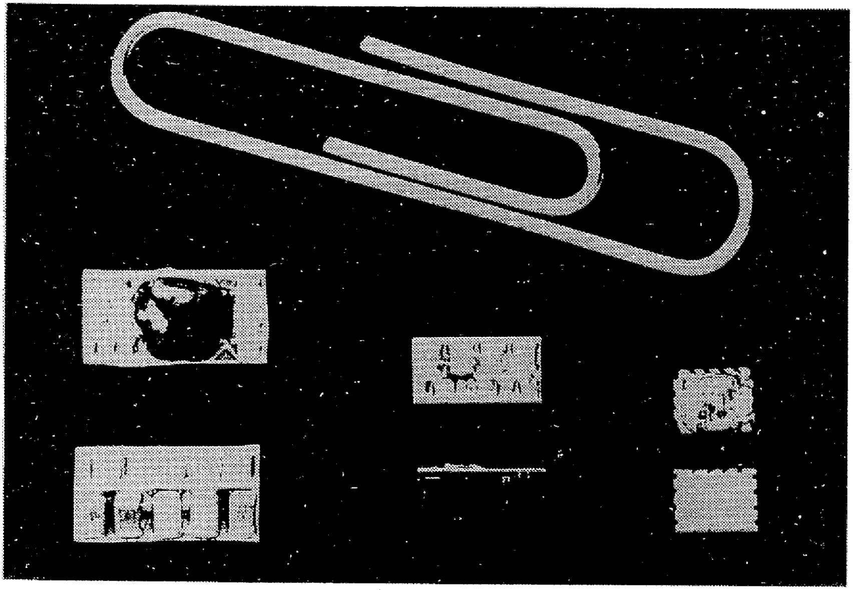


Fig 28.
Enlarged view of three K-AMP hybrids: the 19D, 24D, and buried-capacitor-substrate 28D.

The 28D makes possible the tiny K-AMP hearing aid that is shown in my ear and on my finger in Figure 29. This is what several of us suggested be called a CIC hearing aid, a Completely In the Canal hearing aid. (To qualify as a CIC hearing aid, all parts of the hearing aid - including its volume control, battery drawer, etc. - must be 1-2 mm inside the entrance to the earcanal). This particular CIC hearing aid is a work of art, (and quite expensive). Indeed, to make such a K-AMP hearing aid with present-day parts requires a sculptor who is willing to spend hours visualizing how the parts might be placed and sculpting the shell so that it slips in smoothly without pressing against the skin in the bony part of the earcanal. Mine slips in and is completely comfortable, with relatively low occlusion effect. I'm grateful to Randolph Giller for proving that it was possible. (As an added flourish, Giller included the BF-1743 metal damped coupling assembly, an assembly some manufacturers have claimed was too large even for ITE hearing aids.)

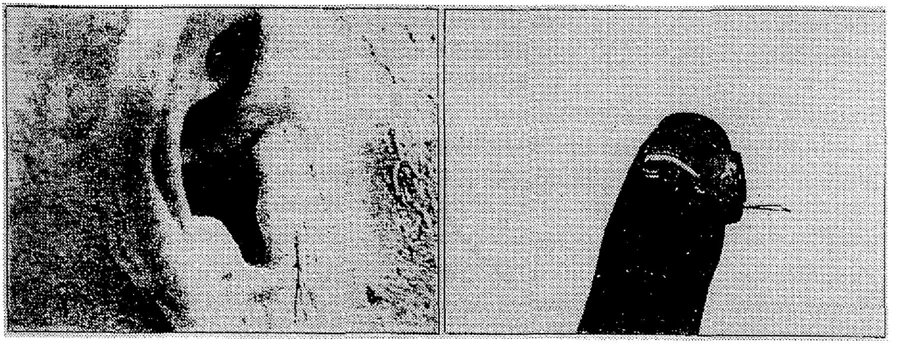


Fig 29.
Completely-In-Canal (CIC) hearing aid made possible by the small size of the 28D (not to mention the small size of the microphone, receiver, trimpot VC and battery) and an enormous expenditure of time.

The CIC hearing aid may be difficult at the moment, perhaps, but once something is proven possible improvements follow rapidly. It is only a matter of time before even smaller parts will make it possible - on a regular production basis - to put hearing aids inside the ear so deeply that you can ask someone to look at your hearing aid and have them say "what hearing aid?" And really not be able to see it unless they pull your ear back. Until wearing a visible hearing aid carries no stigma, the CIC aid solves the problem for someone who doesn't want to appear handicapped.

E. Two features contributing to high fidelity

Two circuit tricks have contributed substantially to the overall fidelity of the K-AMP circuit. First of all, I mentioned earlier that the input circuits of many hearing aids typically overloaded at 85-90 dB SPL. The circuit trick that was part of the first K-AMP patent added 30 dB of undistorted input range, illustrated in Figure 30. Instead of overloading at 90 SPL, the circuit itself can go up to 120 dB. Dave Preves, who was kind enough to have the first K-Amps built for me three years ago, also made coherence measurements on them.

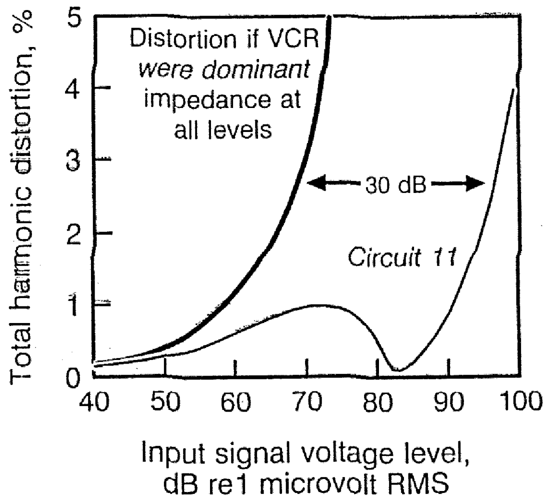


Fig 30.
Reduced distortion for high-level input signals in K-AMP circuit, compared to typical hearing-aid input amplifiers.

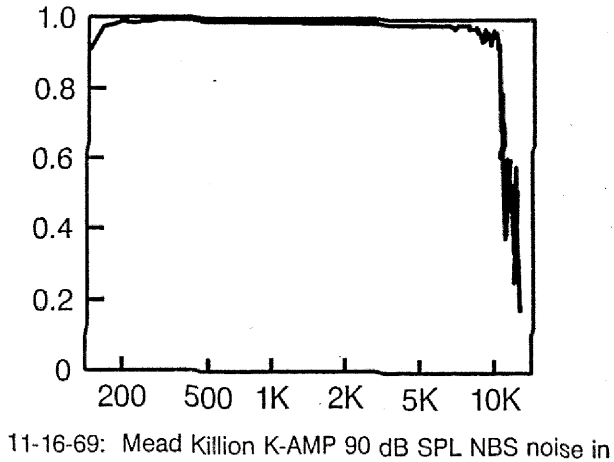


Fig 31.
Coherence measurements on first K-AMP hearing aid: 90 dB SPL input.

Figure 31 shows those results, verifying that at 90 dB SPL input the complete K-AMP hearing aids had a coherence of essentially 1.0 (i.e., zero noise and zero distortion) out to 13 kHz.

The other circuit trick was patented by Dave Hotvet, and is called Adaptive Compression\ (a registered trademark of Telex Communications in the U.S.). I enjoy explaining how it works, so I'm going to do so.

First, let's review the dilemma in trying to choose a single time constant when you don't have Adaptive Compression. If you choose a fast time constant, background noise rises up immediately when the talker pauses giving rise to what is called "pumping" of the gain. In the city of Chicago, for example, baseball broadcasts are made with an extremely short-time-constant compression amplifier (I'd guess 50 milliseconds or less). When the announcer says "Johnson is coming up to bat" and pauses, the crowd noise comes up as soon as his voice stops. So what you hear is "Johnson is coming up to bat...KHSCCSKCHSCKK...and he swings...KSSKHSKHHKHSS...and misses...KHSSKSKHHKHSS..." with the...KHSSKSKHHKHSS...of the crowd noise as loud as the announcer's voice. The broadcast engineers could solve that problem anytime they wished, but it's apparently a part of the Chicago baseball mystique, intended to convey some of the excitement of the game, I guess.

Exciting for baseball, perhaps, but such a fast time constant can be especially irritating if you're listening to a live lecture, for a reason that I'll explain in a moment. We learned about this problem the hard way while we were relearning the importance of actually wearing samples of the hearing aid designs you were working on. We had spent hours and hours in the laboratory listening to the K-AMP circuit over very-high-fidelity loudspeakers. We listened to classical orchestra, jazz ensembles, male and female voices in quiet and with cafeteria noise, street noise, and indeed everything we could think of in the background.

Based on these listening evaluations, we convinced ourselves that the K-Amp hearing aid did not need Adaptive Compression, even though Harry Teder thought I was crazy (I think those were his exact words) not to consider it seriously. I explained to Harry that because the K-AMP circuit used a wide-dynamic-range compressor with a relatively low compression ratio - just a little over 2:1 - the gain change was spread out over such a wide range of input levels that "his" problem wasn't a problem for us. Adaptive Compression made a dramatic improvement with a high-compression-ratio output limiting circuit such as Telex used, but our listening tests convinced us that we didn't need it in the K-AMP circuit. (We weren't the only ones. We recently obtained a U.S. patent on Adaptive Compression used with wide-dynamic-range compression amplifiers which increased the gain for quiet sounds, successfully arguing that all previous uses - broadcast, recording, and hearing aid - had been restricted to compression-limiting circuits which reduce the gain for loud sounds. [Killion et al, 1992])

The problem with our listening tests was that they used loudspeakers. Fine for checking fidelity and the like, but with all sounds coming from one source location the sounds combine into one Gestalt and the "pumping" effects of compression are less noticeable. In real-world listening with a fast-time-constant compressor in headworn hearing aids, however, it becomes very irritating when the voice of the lecturer up in front modulates the shhhhh of the projector fan on your right. No problem when both come together from a single source location, but once you have normal localization in a real room, it is very unnatural for the sound of the projector to modulate up and down as the lecturer speaks. The SHHHshhhshhhhhhhhshhhSHHHHshhSHHHHshhSHHshhhhhSHH sound from the projector becomes annoying. You can live with it; you can get used to it; I got used to it in the first week or so of three months wearing K-AMP hearing aids, trying to convince myself that it was not really annoying, but it was.

During this denial period, Harry Teder finally convinced me to come up to Telex for technical assistance and to seriously consider adding Adaptive Compression to the K-AMP circuit. We both became convinced that in the real world, wearing headworn hearing aids which allow you to localize different sound sources, Adaptive Compression made a significant improvement to the K-AMP circuit. (An interesting postscript to the story: When Harry and Art Johnston and Steve Hanke and I finished the day at Telex, we set out to demonstrate to management what a big improvement we had. We wired the K-AMP circuit up for a demonstration using - you guessed it - loudspeakers, and then spent about 1/2 hour trying to find demonstration conditions that would clearly illustrate the improvement. We ended up with a weak demonstration that was not convincing at all, even to us. But if you put an On-Off switch on the adaptive compression in a pair of K-AMP ITE hearing aids and wear them, it's a night and day difference.)

When you don't have Adaptive Compression, the standard solution to the pumping problem in audio equipment is to use a long, slow time constant; perhaps 2 to 3 seconds. You'll find that sort of time constant in every one of the portable cassette tape recorders we all use. They don't have a recording volume control, only a high-compression-ratio compressor with a long recovery time. So why not do that in hearing aids? The problem is that if you have a 2 to 3 second recovery time, every loud sharp click or snap will take you "off the air" for a second or so. Telex has a nice demonstration cassette that illustrates this phenomenon.

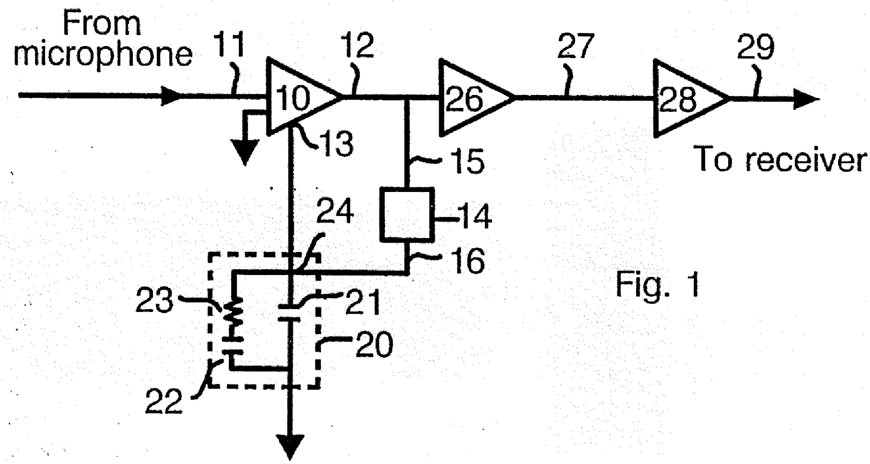


Fig 32.
Adaptive Compression circuit. (After Hotvet, 1988, U.S. Patent 4,718,099).

So with regular compressors you're caught between the devil and the deep blue sea: pumping sound or off-the-air-with-every-click sound. The Adaptive Compression solution illustrated in Figure 32 is really cute. Hang in there with me and I'll explain it simply. See the two capacitors labeled 21 and 22. Capacitor 21 is a small one, which gives the standard fast-recovery-time (50-100 mS) behavior typical of hearing aid circuits. (With capacitor 21 alone, the hearing aid is never off the air, but listeners complain of the pumping sound. A few listeners would even rather have linear circuits.) When a loud sound comes along, the voltage on capacitor 21 is yanked down - so the gain of the hearing aid goes down - and then you have a current source which is pulling it back up so that 50-100 mS later the gain will be restored. When you add the large capacitor 22 and resistor 23, what happens with a loud click is that the small capacitor goes down quickly just as before, but the big capacitor doesn't know anything has happened, it's just sitting there fat and happy because the resistor isolates the big capacitor from sudden transients. But now the big capacitor has its normal voltage and the small capacitor has a low (low-gain) voltage and so a relatively large current (10 times normal) immediately flows from the big one into the small one to help it recover. In effect, the big capacitor reaches down and yanks the little one back up! The result is that instead of having a 50-100 millisecond recovery time after a short intense transient, you have a 20 millisecond recovery time. In fact, if you're listening in quiet and you have 25 dB of gain and you snap your fingers, it sounds as if nothing happens. The gain drops 25 dB and recovers so quickly your ear doesn't know anything has happened. It is a magic circuit for handling transients. (There is solid psychoacoustic evidence to substantiate this somewhat loosely worded explanation, incidentally.)

For prolonged steady sounds, on the other hand, the big capacitor gradually discharges and then holds on to the small capacitor to prevent the gain from pumping up and down. With the big capacitor as an anchor, rapid changes in gain are held to 5-6 dB, which are not perceived as pumping. As Harry argues, these rapid changes may even operate to improve speech intelligibility. In ongoing speech, the weaker consonant sounds after strong vowels experience a rapid gain recovery of 5 or 6 dB. This 5 to 6 dB relative boost for weaker consonants could act as consonant enhancement.

What Adaptive Compression does in terms of sound quality is circumvent the time-constant dilemma by making the fast-recovery time constant even faster and the slow-recovery time even slower, as shown in Figure 33. For short (1-10 millisecond long) transients, you have about a 20 millisecond recovery time. For prolonged sounds, you have about a 600 mS second recovery time. For speech sounds such as vowels, you have roughly a 100 millisecond recovery time. By the way, this provides the basis for a quick listening check of a K-AMP hearing aid to see if it is working properly. This is what I call the "long loud aaaaaah test." You say a loud aaaaaah long enough (1 to 2 seconds) to shove the voltage on the big capacitor down completely, and then listen for the momentary delay you should hear before the background noise comes back up.

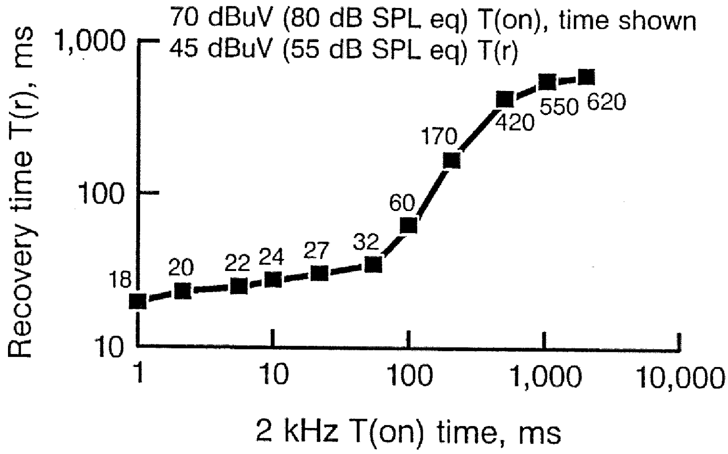


Fig 33.
Recovery time vs. transient duration: variable recovery time of K-AMP Adaptive Compression circuit.

F. 95% Succes Vs. 50% Succes

So we have a hearing aid with a 16 kHz band width, no audible distortion, a sensible level-dependent frequency response, and Adaptive Compression. That combination was so successful in the hands of the leading-edge dispensers who started using K-AMP hearing aids that they kept pushing the envelope and fitting it on more and more people. And after a while, they were reporting 90-95% success rate on almost everyone they put the K-Amp hearing aid on. Those dispensers were carefully reading the articles and data sheets and Etymotic's first Ordering and Fitting Guide (which wasn't particularly easy to read). They called the factory when they had a problem, did real-ear measurements, and became sophisticated at modifying and adjusting the aids.

Any manufacturer who has introduced an innovative new design can tell you the next part of the story. The second generation of dispensers, hearing that the K-AMP hearing aid was a panacea that solved all problems, started dispensing them on their difficult losses, but didn't read the data sheets as carefully (after all, they heard it was supposed to almost fit itself), they didn't call the factory to find out how to adjust the trimmers, and the hearing aids didn't always work. The manufacturers starting seeing 20-30% overall return rates.

So we have now taken a somewhat more cautious, sadder but wiser approach, and have attempted to separate high-risk from low-risk fittings. For the range of hearing losses shown in Figure 34 we said OK, based on our experience and what we learned from our customers; this is a low risk fitting. With a loss in this range, you probably don't have to read the data sheet or even adjust the trimmers. You can probably take the aids as they come from the manufacturer, put them on someone, and with an adjustment of the volume control they'll probably be happy.

Figure 34 showed one of the low-risk fitting categories. The other two were both shown in Figure 21: Mild-moderate flat loss and mild-moderate reverse loss.

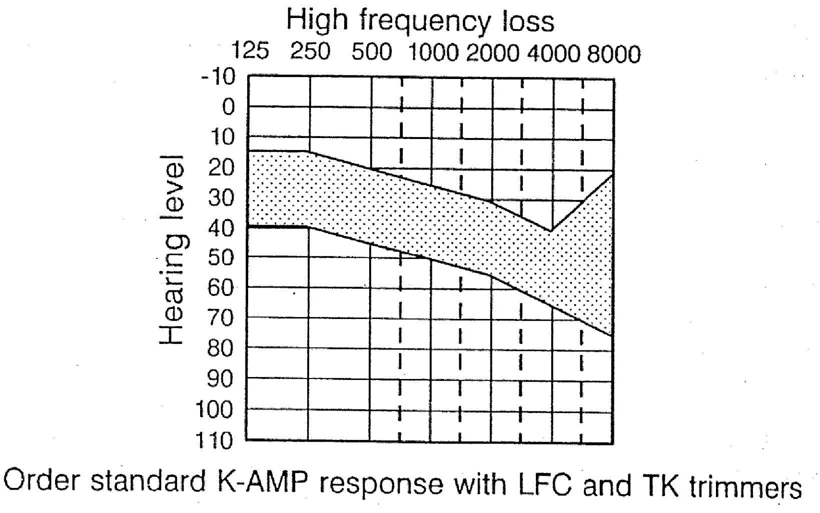


Fig 34.
Low-risk fittings for K-AMP hearing aids: moderate high-frequency losses having high probability of a successful fitting without special effort.

G. High risk: Moderate-severe flat loss

Figure 35 shows a high risk fitting. You need to use a high power receiver and you're pushing the envelope, but there are thousands of these that are highly successful K-AMP fittings, individuals who report that these are better than anything they've ever heard. But there are also thousands of aids ordered for losses like those which have come back to the factory. I would like to just take a few minutes describing how to keep them from coming back.

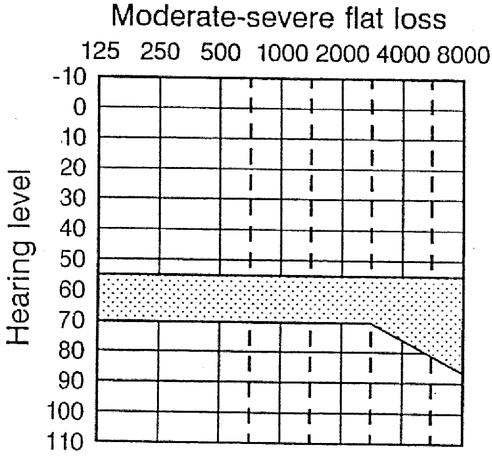


Fig 35.
High-risk fittings for K-AMP hearing aids: moderate-severe flat losses which require special transducers, circuit modifications, and/or counseling.

First of all you need to understand the volume control in a K-AMP hearing aid. We have spent so much time talking about the fact that the ideal K-AMP hearing aid doesn't amplify loud sounds that some dispensers think that no matter what you do with the volume control K-AMP hearing aids won't amplify loud sounds. If you ask what the volume control does, the answer is that it makes things louder, except loud sounds don't get any louder. Now that's a little too much. In fact the volume control is there to allow you to achieve the doesn't-amplify-loud-sounds result should you want to.

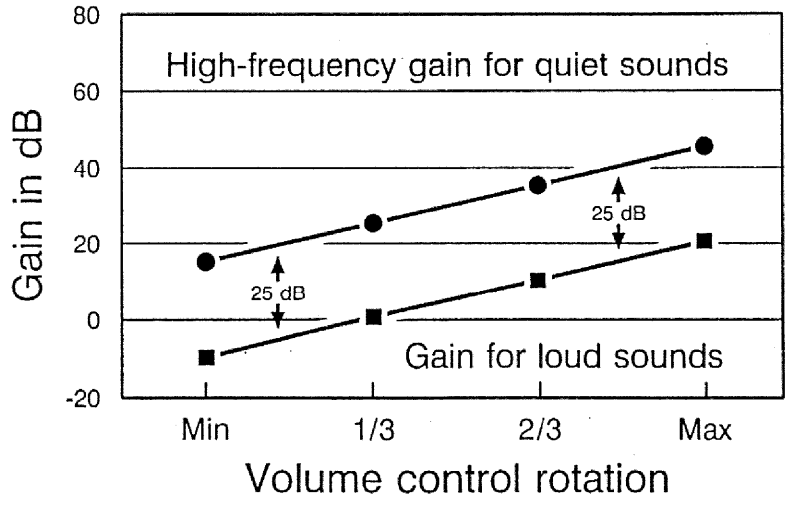


Fig 36.
Automatic 25 dB increase in high-frequency gain is unaffected by volume control setting.

Figure 36 illustrates the action of the K-AMP volume control on the standard-power version measured on an average ear. If you set the volume control for about 1/3, you will produce 0 dB gain for loud sounds (everything from 90 dB SPL on up will come out unamplified). You will get 25 dB more gain for quiet sounds.

If you turn the volume control all the way <u>down</u> you will have a hearing protector. The K-AMP hearing aid will actually attenuate loud sounds by about 10 dB and give you about 15 dB gain for quiet sounds. If you turn the volume control all the way <u>up</u>, on the other hand, you can have 20 dB of gain for loud sounds and 45 dB for quiet sounds. What is fixed here is the 25 dB difference in gain for loud sounds and quiet sounds. What is <u>not</u> fixed is the <u>overall</u> gain. Technically, the K-AMP hearing aid is an input-compression aid, so that the volume control determines the output range into which the compressed input range will be fit.

Now we can see how we might fit someone with the moderate to severe flat loss shown in Figure 35. Clearly they need more than 25 dB of gain for quiet sounds. Something like 40 or 45 dB would be more appropriate. They will obtain that when they turn up the volume control, but if they do that loud sounds will be made louder also as shown in Figure 36.

So we start by asking whether or not we are going to get into trouble when they choose 15 or 20 dB of gain for loud sounds in order to obtain 40-45 dB gain for quiet sounds. Fortunately, it looks as if the answer is generally no. The data that Barfod accumulated back in the 70's on the level at which complete recruitment sets in (which means loudness comes back to normal) is shown in Figure 37. His data showed that by the time you have a 70 dB hearing loss you need about 14 dB of gain for a 90 dB input in order to reach full normal loudness. Will 14 dB of gain for loud sounds cause discomfort for very loud sounds? Pleasantly enough, we have the impeccable loudness discomfort data of Pascoe (1989), shown in Figure 38, obtained on 500 ears at 4 frequencies: 500, 1000, 2000, 4000. Pascoes's data indicate that the average individual with a 70 dB hearing loss has a loudness discomfort level that has been elevated on the average by 17 dB.

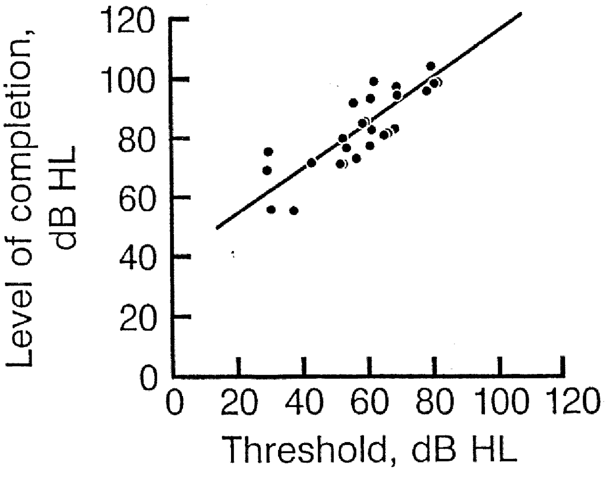


Fig 37.
Level at which complete recruitment (return to normal loudness) occurs as a function of hearing loss, from all subjects and frequencies (From Barfod, 1978).

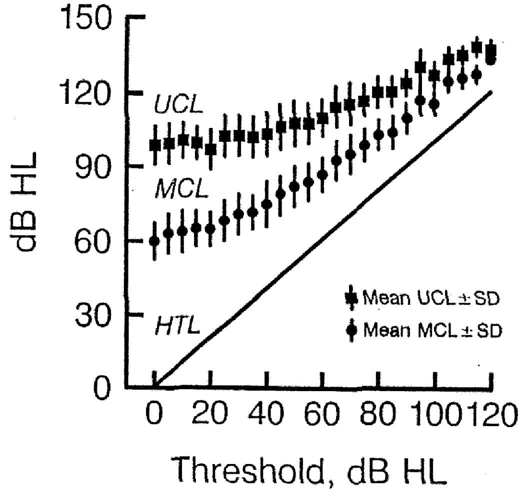


Fig 38.
Mean comfort and discomfort levels for pulsed pure tones as a function of hearing loss, from 500 subjects and four frequencies. Data above 120 dB HL extrapolated from lower-level loudness-scale judgements (From Pascoe, 1989).

We conclude that the person with a 70 dB loss, using 14 dB of gain for all loud sounds, would have no less headroom with regard to discomfort (actually 3 dB more) than the normal-hearing person without gain. The nice thing about this way of looking at the problem is that the answer doesn't depend on input level: Whatever a normal-hearing person would be exposed to would cause no more discomfort to the hearing-aid wearer than to the normal-hearing person.

H. Moderate-severe high frequency loss.

Figure 39 illustrates individuals with a moderate-severe high frequency loss, representing another high-risk category. The upper right audiogram is for an experienced dispenser who has tried many different hearing aids over the years, and reports that the K-Amp hearing aids are overall the best yet.

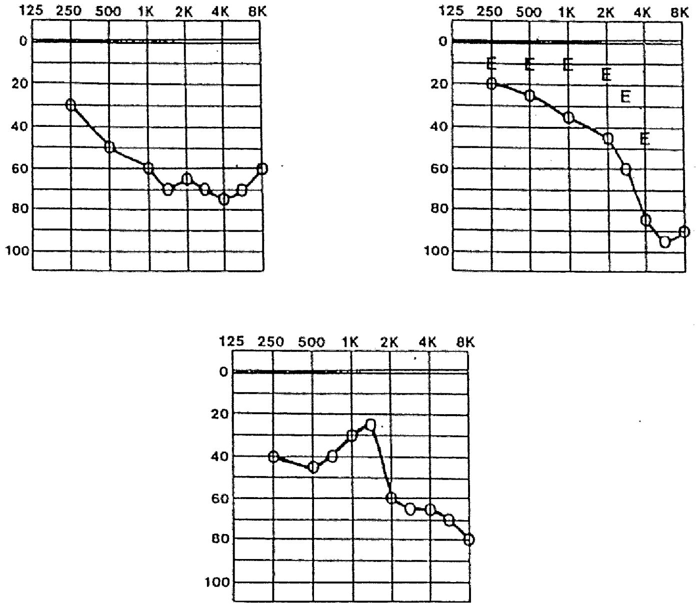


Fig 39.
Audiograms of three successful K-AMP hearing aid wearers ("E" stands for Expected aided threshold).

But fitting someone like this is even trickier than the moderate-severe flat loss individuals, and you need to know how another K-AMP control works. This one is what we call the low frequency control, the LFC trimmer. In the standard high-fidelity version of the K-AMP hearing aid, this trimmer is wired in series with the volume control. Most hearing aid engineers think this is a questionable design: it's not a good response control because it interacts with the volume control. When the volume control is turned down near minimum, for example, the LFC trimmer has very little effect on the response as shown in Figure 40. (We have received phone calls to say the tone control doesn't work, they can't see much difference at either extreme setting.) If you turn the volume control all the way up, on the other hand, the LFC trimmer has a lot of effect; some 30 dB range at 125 Hz as shown in Figure 41.

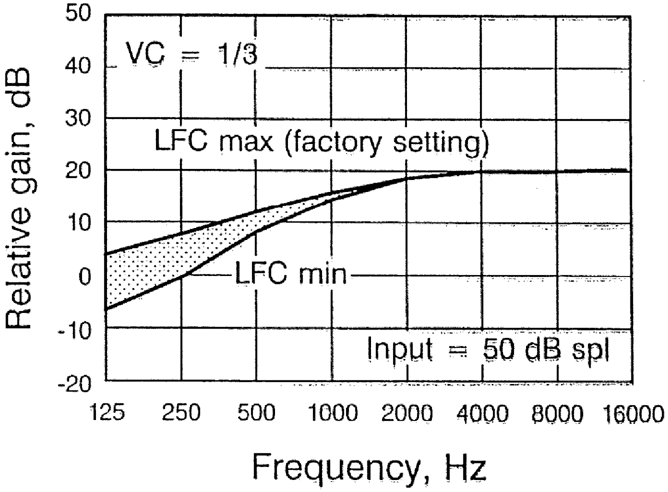


Fig 40.
LFC has little effect at low volume settings.

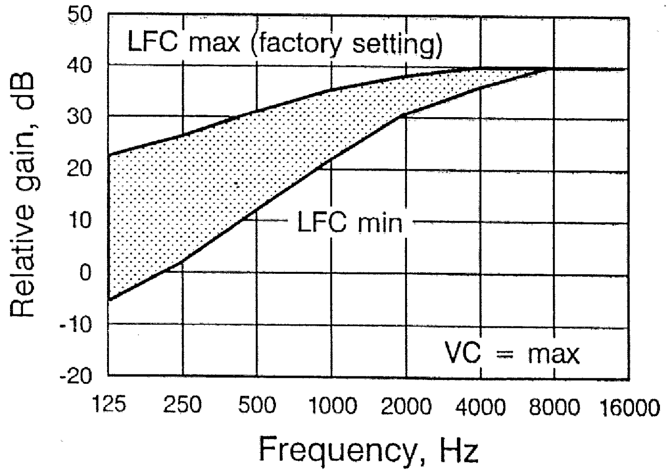


Fig 41.
LFC has large effect at high volume settings.

Having a tone control interact with the volume control may or may not be a disadvantage but, following the Tektronics adage (if you can't fix it, feature it), we choose to feature it. At the minimum-bass-response setting of the LFC trimmer, it turns the volume control into a treble boost control. This is shown in Figure 42. When you have individuals with the audiogram I just showed, who never need any low frequency gain, you can now give them a volume control which allows them to decide for themselves just how much boost they want for loud sounds at high frequencies (and they're always going to get 25 dB more high-frequency gain for quiet sounds).

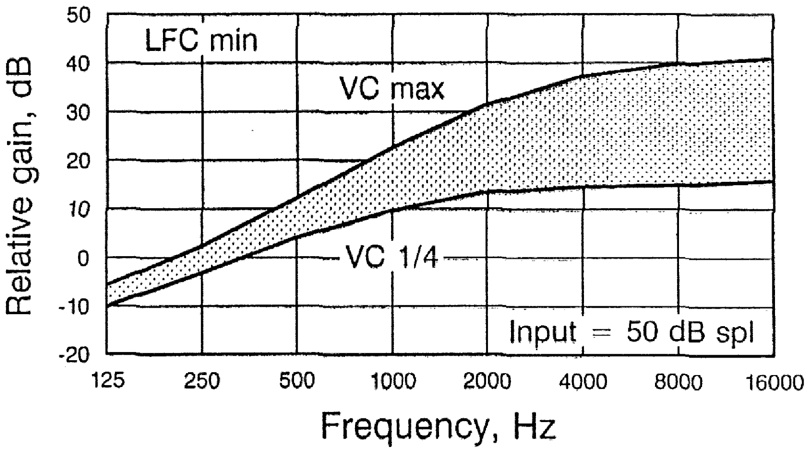


Fig 42.
Volume control becomes treble boost control at reduced LFC settings.

Figure 43 shows the insertion response available at the extreme settings: You can have a hearing aid that gives you 20 dB of gain for loud sounds and 45 dB gain for quiet sounds at high frequencies, without ever giving you any low-frequency gain. I believe the reason that the K-AMP hearing aid has been so successful on moderate-severe high-frequency hearing losses is that it can provide the flexibility just described when it is properly adjusted. Needless to say, if you don't adjust it properly it probably is not going to be the right hearing aid for those types of loss. (Unlike typical hearing aids, a K-AMP hearing aid operated near full-on volume setting may be perfectly reasonable. Nonetheless, the high-power receiver is a good choice for this type of loss because it gives 5 dB more gain margin.)

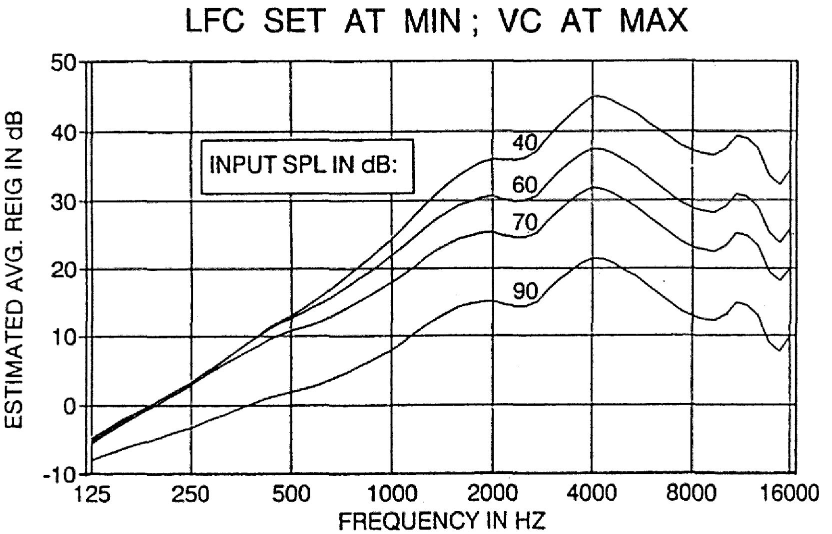


Fig 43.
Insertion response available at extreme settings of LFC trimmer and volume control.

IV. What's wrong with the K-Amp hearing aid?

I don't want to spend much time on this topic, but there are some limitations that should be mentioned.

A. Real-ear response irregularity.

The typical real-ear response of the K-AMP hearing aid has a dip at 2.5 kHz and a boost at 4 kHz. That is a receiver problem that Knowles will shortly solve, I believe.

B. Ultrasonic alarms.

Ultrasonic burglar alarms and ultrasonic light-control sensors can affect the hearing aid gain. Harry Teder pointed out this problem to me some 15 years ago; it's not just a problem for the K-Amp hearing aid. The level of the 25 kHz or 40 kHz ultrasonic signals can exceed 100 dB SPL, and every small motion in the room affects the ultrasonic level at the hearing aid input (and thus the gain). There are two solutions: Roll off the high-frequency response of the microphone (Etymotic makes a glue-on microphone filter that we have used successfully for this purpose) or, with the K-AMP circuit, you can use what we call the Loudness Boost switch. With this, the user can flip in a fixed amount of gain for those circumstances where an ultrasonic system would otherwise run the gain up and down.

C. Increased feedback problems.

Increase difficulty with feedback is absolutely a problem with the K-AMP hearing aid and with <u>any</u> hearing aid that makes it practical for the wearer to obtain the gain needed for quiet sounds. If the user doesn't have to turn down the gain to live with it he or she probably won't. This means they're going to be dealing with more gain, and thus you're going to have more trouble with feedback. You're going to have to get a better fit and you'll have more trouble juggling the amount of venting vs. the occlusion effect. (I recommend Etymotic's foam E-A-R rings for temporarily stopping slit-leak feedback as a diagnostic measure).

D. Occlusion effect not solved.

The occlusion or hollow-voice problem has not been solved by the wideband response of the K-AMP hearing aids, although at first we thought it might have been. Adjustment of the low frequency response with the LFC trimmer often relieves the problem, as it does in any hearing aid, but appropriate venting is still the first line of attack.

E. Response-smoothing dampers get clogged.

We have a couple of new damper removal tools that ease the problem somewhat, usually making it possible to replace the dampers in the office rather than sending the hearing aid back in for repair. The dampers make great wax traps, saving an expensive receiver replacement, but become clogged sooner than the more deeply-placed receiver would. Some manufacturers have switched to a modified-response receiver from Knowles. This gives a reasonably smooth response without the need for dampers.

V. Noise complaints.

Circuit noise and background noise are the basis for the two types of noise complaints.

A. Circuit noise.

An individual with a region of normal hearing, or someone with a mild hearing loss who turns the volume control way up, may indeed be hearing circuit noise. Precisely speaking, it's the amplified microphone noise that they hear, not the the circuit noise which is much below that. One possibility is that the person is used to linear hearing aids and has turned the gain up too high, trying to make loud sounds sound as loud as they do with the old linear aids. The solution here is to counsel them that while they are accustomed to a lot of loudness, they don't need loudness. What they need instead is clarity, and to turn the gain down. That may solve the complaint.

Another reason for a circuit noise complaint is that someone has a region of normal hearing and receives too much gain increase in a quiet room. For that individual, an adjustment of the Threshold Knee (TK) trimmer is required. This TK trimmer is a poor man's compression-ratio control. As shown in Figure 44, instead of changing the K-AMP's 2.2:1 compression ratio itself, you change the lower threshold knee so that the gain stops increasing as the level drops below 65 dB SPL. Instead of increasing 25 dB over the entire range from 90 dB down to 40 dB SPL (TK min setting), you have only a 12 dB increase from 90 dB down to 65 dB SPL and constant gain below 65 dB (TK max setting). If you take the average gain change over the full 90 to 40 dB range, it effectively works out to reducing the average compression ratio to about 1.5:1.

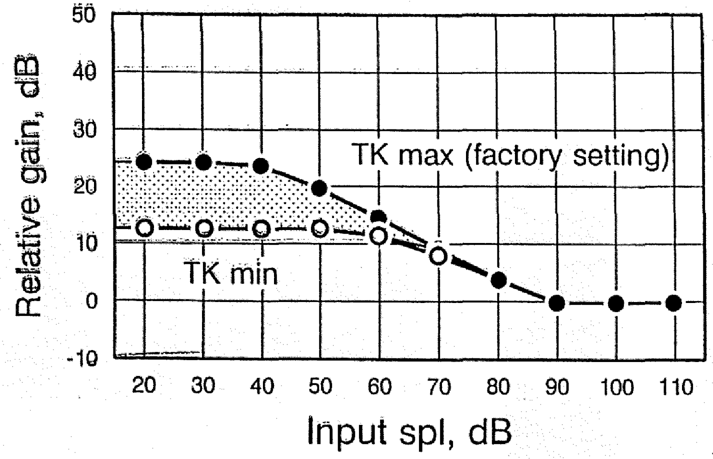


Fig 44.
TK trimmer reduces high-frequency gain in quiet by as much as 12 dB.

But your client doesn't care about compression ratios. The point is that with the TK trimmer turned up, your client won't have as much gain in quiet and won't hear the circuit/microphone noise. For someone who really wants more loudness, on the other hand, the loudness boost (LB) trimmer illustrated in Figure 45 may be even more useful. The LB trimmer also acts to reduce the average compression ratio, but instead of reducing the gain (and high-frequency boost) for quiet sounds, the LB trimmer can be used to increase the gain for loud sounds.

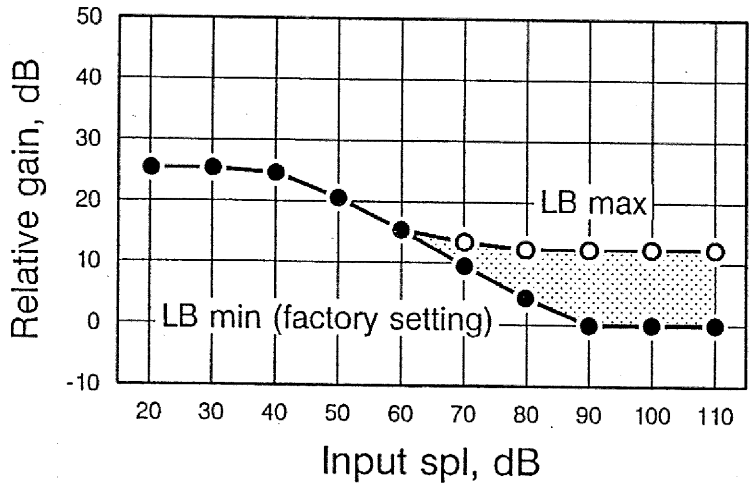


Fig 45.
LB trimmer increases gain for loud sounds.

B. Background noise.

The other type of noise complaint comes from background noise in the room. When an individual hears noises but can't localize them because they haven't been heard for a long time, you may hear complaints that the hearing aid is noisy. You can check this out by simply having such individuals put their fingers over the microphone inlets. If they put their fingers over the microphone inlets and the noises go away, the noises are not in the hearing aid! Those people may tell you that it sounds like noises in the hearing aid, but that's because they just can't localize such noises anymore. The noises they hear are actually out in the room. We have a relearning problem.

VI. Learning to hear again in noise.

Our first temptation is to attempt to filter out the noises. What we all would like is a hearing aid that would accept speech and noise at the input and give only filtered speech at the output, giving a noise-free speech signal at the output. Unfortunately, you can't have it both ways. As Villchur explained, you can suppress the noise, or you can hear speech clearly <u>in</u> noise, <u>but not both</u>.

A. Filter the noise or hear clearly.

Figure 46 provides a visual example of the problem. In the upper left picture, you see the face of U.S. Surgeon General Koop clearly in visual quiet. You can still make him out in visual noise, as shown in the lower left picture. You can also still make him out if you cut out the highs and lows, as in the lower center picture (sort of a visual analog to a narrow band hearing aid). But if you have a narrowband look in noise, you have trouble. In quiet you can get by with a narrowband hearing aid or a narrowband visual system, but in noise you run into trouble. With a narrowband system, you've lost the redundant information that allows you to hear in noise or see in noise.

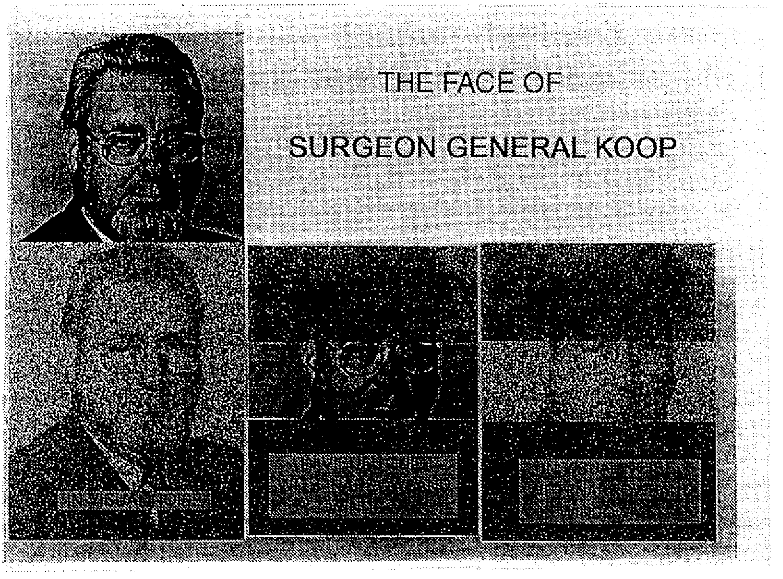


Fig 46.
A visual analogy to the problem of hearing in noise with narrowband hearing aids.

Figure 47 provides another example that Don Wilson made from Dave Pascoe's visually noisy illustration of the sentence: "Words that are not clear when they come into the hearing aid will not be clear when they come out of it." When Don masked off the highs and the lows in visual quiet, you don't have any trouble. But in visual noise, it is virtually impossible to determine what those words are when you've knocked off the visual analog of the high frequencies and the low frequencies.

FROM PASCOE (1991)

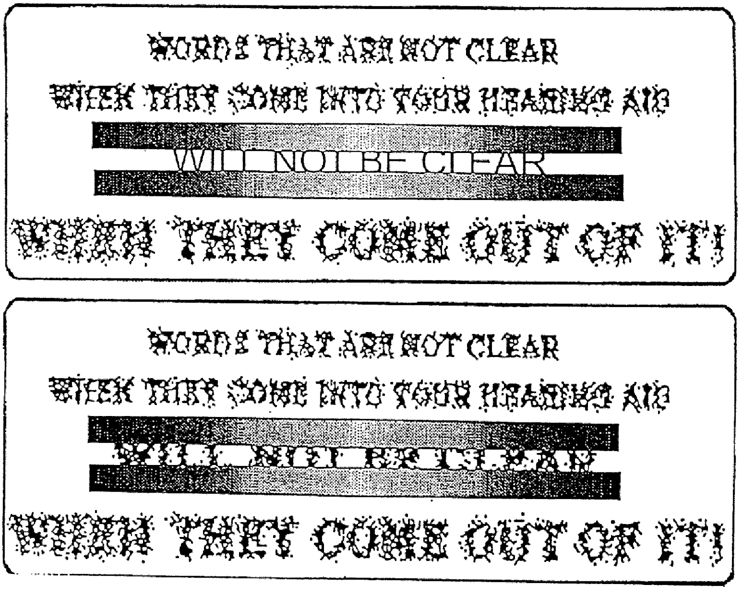


Fig 47.
Another visual analogy to the problem of narrowband hearing aids. (Adapted from a figure of Pascoe, 1991.)

This brings us all the way back to Skinner, Karstaedt, and Miller's (1982) data, summarized in Figure 48. In a series of experiments where they rolled off sometimes the high frequencies and sometimes the low frequencies and sometimes both, no matter what they rolled off the word recognition score went down. By the time they got down to the bandwidth of the old fashioned hearing aids, the score had gone way way down.

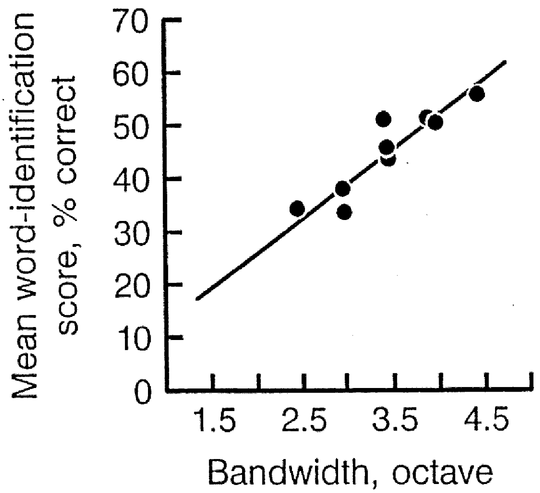


Fig. 48.
Intelligibility vs. bandwidth for two hearing impaired listeners. (Skinner et al 1982).

B. Abonso: The only solution.

Well, we would like to have a good noise filter that makes speech clearer, but we don't have one. As Villchur has argued, we've been taking the wrong approach to the problem. The problem is <u>hearing clearly in noise,</u> and the problem of the hearing impaired person is he's lost ABONSO: the Automatic Brain Operated Noise Suppressor Option.

That can happen to normals. In fact, if you take some of the narrow band, peaky, distorting hearing aids that you used to put on people, and you wear them yourself, you'll find that you go through several weeks in which it sounds as though you have noisy hearing aids. A few weeks of wearing them and you'll recognize that it's the air conditioner, and the fan in ceiling, and the hard-disc drive in the computer. But at first the noises seem to all be in the hearing aids. You've changed the spectral pattern of all those sounds and it takes quite a while for your brain to learn the new patterns. So normals can lose ABONSO.

I experienced that for myself the first time back in the 60's when I wore a pair of lumpy-response, distorting, standard-issue hearing aids for some months to see what it was like to wear hearing aids. At first, I was surrounded by a sea of noise from sources I couldn't identify or localize. After six weeks, I got used to the sound of the aids and they subjectively disappeared; I could identify and localize all the noises. (But I still couldn't understand speech in noise, of course.)

Well how likely are we to get a real noise suppressor?

The most powerful super-computer today, the Thinking Machine, has 65,000 computers wired together, operating simultaneously on the problem in an attempt to mimic the brain. It takes one-half hour to recognize a face, something a baby can do in about one-half second. Yet we have been hoping for something that will operate on a 10A battery, fit in your ear, and replace the brain!

I'd be foolish to say it can't happen, but the best high-speed-number-crunching digital-signal-processing noise-reduction system in the U.S. would require some 40,000 hearing aid batteries a week to keep it alive if it were to be operated from hearing aid batteries. And even with all that horsepower, it doesn't improve the intelligibility of speech in noise - either for normals or hearing impaired subjects. It reduces the noise, but it doesn't clarify the speech.

C. Why the brain wins.

The reason the brain is so hard to replace is that it is so incredibly powerful. We're so used to the miracle of speech perception that we take it for granted. I'd like to give some quick examples from vision that illustrate the powerful processor we all carry around. You can take inverting glasses, tape them on your forehead, and at first you can't read (unless you're a salesman), you can't write, you stumble around, and you certainly can't ride a bicycle. Yet in 4-6 weeks of constant wearing of those inverting glasses, you become able to do all those things. The world is back to right side up. When you finally take the glasses off, however, you go through a period of time in which you can't ride a bike, read, write etc.! (The experiments were motivated by an interest in whether the inverted image on the back of your retina is hard-wired to appear right side up to the brain, or if the necessary visual inversion can be learned.)

An even more powerful example. You can take glasses that distort differently depending on whether you look to the left or the right. So if you're looking at a building of windows, the windows on the left might exhibit barrel distortion, while the windows on the right exhibit the opposite distortion. Worse, if you look straight ahead and move your head quickly back and forth, the shape of the windows goes through wild gyrations. After six weeks of wearing these funny glasses, however, the windows all look square <u>and you can move your head quickly back and forth while looking straight ahead and the windows don't move!</u> Your brain, <u>in real time</u>, can deconvolve all of that and filter it out so that everything stands still! (When you first take the glasses off, the windows go through wild gyrations when you move your head back and forth quickly.)

D. An auditory example.

I will only give you one auditory example of my own. When I was first working on high-fidelity hearing aids a decade or so ago, I wore a pair of ITE hearing aids with smooth response but a response that rolled off sharply above 8 kHz. I wore those aids regularly for several weeks. One night, in a fit of enthusiasm, I wore them to bed. The next morning when I got up, I had forgotten that I had them on, turned on the shower, and was about to step into it when I suddenly realized I was about to ruin two hearing aids. I quickly took them out, and all of a sudden heard a very-high-pitched ssssss. I couldn't locate it. It wasn't quite located in my head, but I couldn't find it in the room. I spent two minutes walking around the room, cocking my head, listening to this very high frequency ssssss. Then, suddenly, it swept up to the shower and joined the broad-band shhhhh of the shower, clearly localized at the shower head. In just a few weeks, I had lost the ability to localize the octave band of sound between 8 and 16 kHz! Now it didn't take me months to get the ability back, it was only minutes, but when you have someone who hasn't heard high-frequency sounds for several years, experience teaches us that it may take several weeks before that person relearns the task, even if you could give him or her perfect hearing aids.

E. Wear it awhile and you'll get used to it?

So we're back to saying "wear it a while and you'll get used to it." But we've been saying that since the 1940's. What's the difference?

The difference is that now you can hear in noise with a hearing aid, whereas back even in the 1960's somebody without a hearing loss needed the 9 dB signal-to-noise ratio shown again in Figure 50, and with a hearing aid they needed an impossible 27 dB signal-to-noise ratio. Today with the right hearing aid you can bring them down close to 0 dB signal-to-noise ratio as shown in Figure 49. You aren't going to bring them to normal in most cases, but depending on the loss and how well you select and adjust the hearing aid, you can bring them down into in the region of signal-to-noise ratios where normal-hearing people operate.

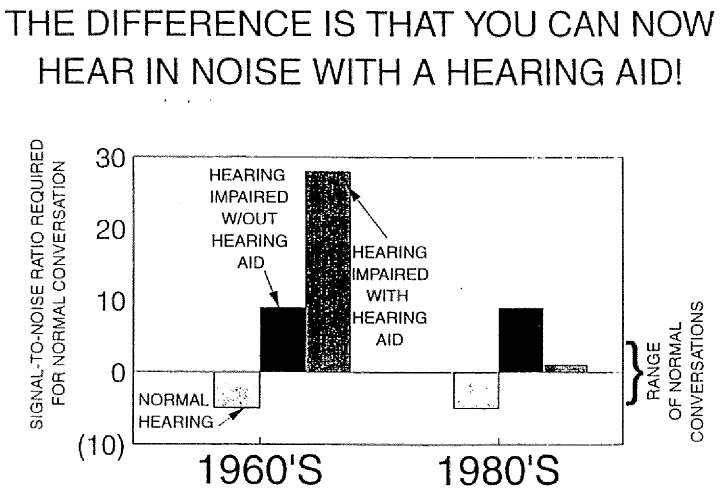


Fig 49.
The difference between the old days and now

The difference is that now you get comments such as, "Now at square dances I can stand in the back of the room with the normal hearing people and still hear the caller," or "With these new hearing aids I can go to a ball game and carry on a conversation," or "I can enjoy a conversation in a noisy restaurant; I never thought I'd hear this well with hearing aids," or "One hearing aid wearer has actually accused another of pretending to hear in situations that are impossible to hear with a hearing aid." We employed a consultant recently that we fit with high-fidelity K-AMP canal aids. Connie Miezio, our office manager, told me that he was almost in tears telling her how much better his new aids were than the (linear) canal aids an audiologist had dispensed to him just 6 months ago. To me personally, he complained that he now had a new problem: he was not getting adequate sleep. He was staying up every night listening to all the CD's in his music collection because they sounded so beautiful.

The summary I have is simple. The world's most powerful data processor (the one in your head) can do almost anything, including learning - in a few weeks - a completely new set of speech patterns. You can get used to even the worst hearing aids. What the brain <u>can't</u> do is separate speech from noise when it is starved for information.

FROM PASCOE (1991)

Success in using hearing aids is learning to ignore the background noise and forgetting that you have them on.

Fig 50.
Pascoe's conclusion.

Dave Pascoe, who has dispensed thousands of hearing aids in addition to providing some of our fundamental research findings, has written a delightful little book that I recommend (Pascoe, 1992). Figure 50 is from that book. "Success in using hearing aids is learning to ignore the background noise and forgetting that you have them on." That is a lot easier with high fidelity hearing aids. And with properly adjusted high-fidelity hearing aids, your brain will no longer be operating on a starvation diet. Thank you.

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