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Interview with Dr. Mead C. Killion, PhD

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Marshall Chasin: The first time that I heard your name was when I was reading the 1981 *JSHD* article on "Earmold Options for Wideband Hearing Aids" (*Journal of Speech and Hearing Disorders* 1981;46[2]:10–20). Was that part of your PhD?

Mead Killion: No that was before my PhD. I was interested in earmold acoustics for years to achieve a sufficient amount of gain in the high frequencies, and most of the hearing aids at the time clipped badly in the higher frequencies so the distortion with louder inputs such as a cocktail party, would significantly reduce speech intelligibility. I became interested in this back in the 1960s about the same time that Dr. Keller (a German researcher) wrote that it was too bad that the Knowles transducers of the time had so many peaks in them because that degraded the sound quality of the hearing aids. I had already shown that the peaks were not in the hearing aid but in the tubing (tubing related resonances) and Hugh Knowles and I were sufficiently concerned about this at the time that we co-authored a paper ("Frequency Characteristics of Recent Broad Band Receivers." Knowles HS and Killion MC. *Journal of Audiological Technique* 1978;17:136–40) showing that with sufficient damping and an acoustic flaring (plumbing) or horn, we could obtain a flatter frequency response. I wanted to call it acoustic plumbing but Hugh thought otherwise: He didn’t like the image it evoked.

MC: Starting with the 1981 *JSHD* article and with the subsequent advent of earmolds with great names such as the 8CR, and 6R12, you seemed to feel that an acoustic high frequency amplification was probably better than an electrical one. Do you still feel that this is the case?

MK: It was only true when most hearing aid amplifiers clipped badly at high frequencies. That problem was solved in the late 1980s with the advent of the class D amplifier, and continues to be solved with the switching output stage in digital hearing aids. Before that, the common class A amplifier required either an excessive amount of battery drain or they use receivers with so many turns on the coil that they “voltage clipped” when even a moderately loud high-frequency sound came along. (If you recall, with a Class A amplifier, half of the peak current is on all of the time, continuously draining the battery even in a quiet environment. That’s why one could predict exactly the battery life of these old-style hearing aids regardless of the level of the input and the volume control used). Back in those days, it was always better to have the acoustic plumbing pick up 8–10 dB, which required only one- third the output voltage (roughly one-tenth the power) at high frequencies. I always enjoyed Hans Bergenstoff’s answer to an audience question in Chicago in 1980: “You could get the same response with electrical equalization, but that would be like driving a car with one foot on the brake and one on the gas.” Switching amplifiers are so efficient that you could (and still can) afford to do that, although some modern hearing aids could still use the response smoothing of a good horn earmold.

MC: Most people know you as an audiologist, but prior to that they would have known you as an engineer. Yet, I understand that you were a mathematician and never took any engineering classes.

MK: That’s true in terms of formal classes. I have an undergraduate and a master’s degree in mathematics. For my master’s thesis my professor gave me what I thought was an interesting problem and I solved it in two weeks. I brought it back and was told that it wasn’t very interesting. He then gave me what I thought was an impossible problem involving two-dimensional surfaces in four-dimensional Euclidean space. I worked on for five years, at the end of which I finally found a way to solve it. My professor liked it so much he had me defend it twice before the faculty. There were two parts to the problem – an easy part and a difficult one. He suggested I use the easier part for my master’s and the more difficult part for my PhD dissertation. I might have become a mathematician, but in the course of solving that problem, I realized that I didn’t like pure mathematics nearly as much as engineering and its applied mathematics.

MC: What led you to discover our field?

MK: I found a technical job working for an engineer’s engineer named Elmer Carlson, director of engineering at Industrial Research Products (a Knowles Company). He was a wonderful inventor and mentor, and even more of a mathematician than I was. After 21 years under his teaching, in 1983 I decided to try my own wings and started Etymotic Research. I started it knowing that 80% of new businesses did not succeed, but only later found out later that those were mostly restaurants; and in fact 80% of new businesses started by engineers who had previously designed saleable products, succeeded.

MC: Tell me about Elmer Carlson and what was to become known as the ER- 15 musicians’ earplug.

MK: Elmer became interested in the fact that a lot of people required less attenuation and a flatter frequency response. I believe that it was Larry and Julia Royster that were quoted as saying that only about 1/3 of the workforce needed any hearing protection at all, and that 3/4 of those needed less than 10 dB of attenuation. So, Elmer thought a moderate-attenuation, say 15 dB reduction, earplug would be useful for almost everyone, even in industry, but especially for musicians. Being the superb acoustician that he was, he designed such an earplug. When Elliott Berger tested it in his EAR-CAL laboratory years later, he found that it was indeed flat, within 1–2 dB dB or so from 80 Hz to 16,000 Hz: What we now call the ER15 Musicians Earplug reduced the pressure at the eardrum by almost exactly 15 dB, compared to the open ear, at all frequencies. It stayed on the shelf because it appeared that the market for such an expensive earplug would be too small to justify the cost of introducing it. Fortunately, one of the viola players in the Chicago Symphony Orchestra ended up with a frightening temporary threshold shift after a concert where 200 musicians and singers were so crowded on stage that his head was almost in the bell of the trombone player behind him. He and a couple colleagues formed a “sound level committee” which resonated (sorry) across the county. I was invited to be a consultant, and once attended a meeting with the orchestra directors and union representative from the six major U.S. symphonies. After that meeting I approached Knowles about the Carlson earplug, and they very generously licensed it to us to produce under the Carlson patent. He himself was quite modest, and strongly declined to have it called the “Carlson Earplug,” so we decided to call it the Musicians Earplug. Oddly enough, some viola players who thought that that was too *much* attenuation, and we later came out with the ER-9 following Elmer’s basic approach. Still later, several drummers said that for jazz and orchestral work the ER-15 was fine, but it wasn’t enough for rock. (Even unamplified, a drum can be beaten within an inch of its life to produce 135 dB peaks, as demonstrated to us recently by one of our engineers!)

MC: Before continuing on with the “ER” or “K”-prefaced other innovations, I recall in the mid-1980s you came out with a series of odd looking ear hooks, one of which was called the K-Bass (or Low-Pass) ear hook, which would allow significant (40 dB) low frequency gain with a non-occluding fitting. I have used it often for those with chronic middle ear dysfunction who require both low frequency amplification and a non- occluding vented hearing aid fitting.

MK: It’s ironic because we are now seeing people lecturing that we can only get high frequency gain and output with a non-occluding tube fitting. The “K- Bass” hearing aid was our first product in 1983, designed for someone with normal *high frequency* hearing which you didn’t want to interfere with. We started with an old Zenith power behind the ear hearing aid that could deliver 135 dB at 125 Hz. Even with the 20 dB loss for an open mold fitting, that still left 115 dB undistorted output at that frequency (which is more than some aids have now). We then coiled a long, small, tube inside the hearing aid to resonate the low-frequency response. An open-ear fitting naturally rolls off the low- frequency response at 6 dB per octave. In the final K-Bass design, the 2 cc coupler response rises at 6 dB per octave to compensate, with the result that a nearly constant 20 dB of gain was obtained from 150 to 1,500 Hz with a non- occluding tube fit. Chuck Berlin and I joked that it should have been called the Killion/Berlin ear hook since he asked if it could be done for one of his patients. The idea of a low frequency fitting with an open canal seems to have been lost but it’s entirely practical, even now.

MC: Would this be able to be redesigned to give you a broadband signal with an open mold tube fit, using today’s feedback management systems?

MK: Yes. It wouldn’t be able to fit into something as small as a pea, but if you had any of the broadband high-gain, high-power, behind the ear hearing aids commonly used for children, this can be done. With digital equalization it would be trivial to shape the frequency response in order to compensate for the roll off in the lower frequency region.

MC: Moving to 1988 you had developed the K-AMP® hearing aid (with the invaluable assistance of fellow Canadian Bill Cole of Etymonic Design). What led you to this when there were already many hearing aids available for almost any use? Specifically what led you to design a hearing aid that could reliably transduce inputs of 115 dB SPL- the limit of modern hearing aids microphones- when the most intense components of speech was around 90 dB?

MK: We invited about eight people to help consult and design various parts of the K-AMP©, including Bill Cole, Norm Matzen and some semi-conductor people. The ability of a hearing aid to handle an input of 115 dB such as many forms of music has always been a design criterion at Etymotic Research, right from the breadboard stage. If it couldn't handle my piano playing or violin playing, it wasn't even considered. Even speech is misunderstood, I believe. The typical (even now) 90 dB maximum input in many hearing aid designs is enough for conversational speech, but not for many social gatherings. I recall Margo Skinner lecturing in Texas that the maximum speech levels were about 80 dB. That night at a Country and Western dance, she was talking across a picnic bench to a colleague and I held a sound level meter to the colleague's ear. It measured peaks of 95 dB. (Mild mannered Margo, indeed!) A 95 dB peak on a SLM corresponds to 105-110 dB instantaneous peak on an oscilloscope. A hearing aid that clips at 90 dB is wildly distorted at 110 dB inputs. A wonderful study by Naidoo and Hawkins in the *Journal of the American Academy of Audiology* (Monaural/Binaural Prefer- ences: Effect of Hearing Aid Circuit on Speech Intelligibility and Sound Quality" 1997;8[3]:188–202) uncovered the reason many users reported they took off one hearing when they were in high-level noise: they heard better with only one distorting hearing aid! When there was not distortion, they preferred two.

MC: If you'll forgive me, we started talking about music levels and ended talking only about speech levels.

MK: You are quite right. Marshall, you yourself have repeatedly reported that many modern hearing aids are totally unacceptable to musicians, and your reports go back 10–15 years and still continue. Just last month we at Etymotic tested three digital aids at the request of a friend at a hearing aid company. The most recent design distorted the most on simple piano playing. Similarly, we sent some electronic BlastPLG earplugs to members of the National Symphony Orchestra, after I confirmed that they didn't distort on my own loud-as- possible playing. The orchestra musicians liked them but complained that they distorted on loud passages. (Which shouldn't have surprised me: It seems only fair that musicians in a world- class orchestra can play a violin or trumpet much more loudly than an amateur can!)

MC: That is interesting and consistent with my own experience fitting musicians at our Musicians Clinics of Canada here in Toronto, but I was hoping you would talk about what you did in the K-AMP design that changed that.

MK: I believe we were the first in the industry to use a balanced-input operational amplifier for the input stage (similar to mixing boards) that could handle 200–300 mV peaks at the input. That corresponds to 116–120 dB instantaneous peak into a typical microphone. (By the way, that input cancelled cellphone interference, so when digital cellphones came into use the K-AMP amplifier was already immune. We didn't plan on that, but it was a nice bonus.) But in order to make use of that input capability, it is important not to throw it away by amplifying loud sounds. Since most people need little or no amplification for loud sounds, the basic K-AMP design carried the undistorted reproduction from input to the ear.

MC: I would have thought you would also have mentioned that you and your design team did all that with only 300 A of battery drain, so hearing aid batteries could last for weeks. Are you as happy with the Digi-K as you were with the K- AMP?

MK: I'm happier with the modern Digi- K in the sense that it allows you to come as close to perfection as possible. In 45 seconds after it is placed in a 16 kHz soundbox, the Digi-K software measures the response, flattens all the microphone, receiver, and tubing peaks within a dB or so, and then introduces the appropriate BTE, ITC, or CIC CORFIG. Whatever goal you set for the frequency response, this approach does it better.

MC: I want to return to something that you just touched on regarding microphones. Electret microphones were invented by G.M. Sessler and J.E. West. (*Journal of the Acoustical Society of America* 1966;40[6]1433). But we don't think of Mead Killion when it comes to electret microphones – I understand that you were involved in the miniaturization of the electret microphone that makes it useful for modern hearing aids.

MK: Yes, and I am a friend of Jim West and followed his work. The first wideband microphone I helped design at Knowles was a ceramic microphone that had an 8,000 Hz bandwidth. This doubled the bandwidth of many magnetic microphones. We chose ceramic at first because that was a known technology and knew we that we could make a reliable microphone. It had the disadvantage of having a greater sensitivity to vibration. As soon as that was in production we started on the development of a stable miniaturized electret microphone. Usually when I had a design problem I showed it to Elmer Carlson who turned the problem around and clarified it and made it simple. That's why Elmer Carlson's name was usually first on any patent. In this particular case the problem was to get a stable structure for the microphone that would not be temperature and humidity sensitive. Instead of stretching the diaphragm it almost was a free-floating one sitting on a bunch of bumps. That structure allowed the microphone function to be essentially free of temperature and humidity. This was a very stable structure and if you put it in a case that was slightly larger, one could show that the internal noise was lower than that of the human ear. The much smaller microphones that are made now come close: Masking level equivalent to about 5 dB HL, which we have extensively confirmed with our BlastPLG earplug units.

MC: If you open up one of your insert earphones that are used for audiometry and research, you see Elmer Carlson's handiwork staring out at you. He invented the twin tube approach, didn't he?

MK: It's nice to have someone remember that innovation, perhaps Elmer's most brilliant. The hearing aid problem he was thinking about at the time was that you can't damp those nasty tubing resonances completely unless you put the damper at the end of the tube, which is the worst possible place to put it in an earmold from the standpoint of earwax (more about that problem below). His hero, Oliver Heaviside, had solved the electrical problem of resonances in telephone lines by realizing that a resistance placed at the end of an electrical transmission line could smooth the frequency response completely if the resistance equalled what he calculated as the "characteristic impedance" of the line. (Incidentally, Heaviside was kicked out of the British Royal Philosopical Society after demonstrating that those who said he was wrong were dunces. He also patented coaxial cable whose advantages can be viewed with a quick Google or Yahoo search.)

MC: That is interesting but what does it have to do with Elmer Carlson and his twin-tube damping method.

MK: Sorry, but you are the first one who has shown an interest in this great stuff. Anyway, Elmer knew all of Heaviside's mathematics and also understood they applied to acoustics as well. The characteristic impedance of an acoustic tube is 41/area cgs Ohms. Thus the ubiquitous #13 tubing, which has an I.D. of about 0.2 cm (0.193 cm to be exact), can be readily seen to have an acoustic impedance of 1,400 Ohms. A common damper of 1,500 Ohms is close enough to smooth the response beautifully if placed at the end of the earmold.

MC: Mead, you are usually direct. Have you forgotten the original question about twin-tube damping.

MK: Patience, my Canadian friend. I just stated that 1,400 Ohms at the tip of the earmold would smooth the response beautifully. But I also stated earlier that a damper in that location would be exposed to earwax: The hearing aid might sound beautiful for a while and then quit sounding at all! What Elmer realized, in a wild burst of intuition, was that you could add an auxiliary tube, so you have *two tubes*, the normal sound tube that is open at the (earcanal) end, and a "peak cancellation tube" blocked at its end. If you used two 1,400 Ohm dampers (in the example of #13 tubing) one damper at the beginning of the sound tube and one at the beginning of the blocked tube, *the combination would have a perfectly flat response.*

MC: What if that damper is right for smoothing the tubing resonances but is the incorrect value for smoothing the receiver response?

MK: Wonderful question. Here we see Carlson's total brilliance: Since you can choose any tubing you want (look at the wide variety of tubes used with open- canal hearing aids today), you are free to choose the tubing diameter, and thus the damper value, that gives the best damping and shaping of the receiver response. Pretty neat, huh?

MC: How has Etymotic Research exploited Elmer's invention, and didn't you need to obtain a license on his patent.

MK: Second question first: We were using Knowles transducers exclusively at the time, and that carried an implied license to such inventions – which we confirmed, of course. Our first use of the Carlson twin-tube damping was in the ER-1 and ER-2 earphones. The ER-2 is the most fun to describe, because it delivers sound to the ear at the end of 10 inches (254 mm) of tubing, and yet produces an eardrum-pressure response on the average ear (as measured on KEMAR and confirmed with probe- microphone measurements) that is flat within 2 dB from 200 Hz to 12 kHz (and 5 dB from 20 Hz to 16 kHz). It uses a #16 sound tube (1.35 mm I.D.) and thus a 1.35 mm pre-formed stainless steel cancellation tube of exactly the same length wound inside the case. The reader can readily calculate the dampers we use. By the way, it is unlikely that the sound tube will become clogged with earwax because the foam eartip is replaced with each use.

MC: What other uses has the twin tube approach been put to, and how have you been involved in them?

MK: The second twin-tube product was our ER-7C probe microphone, which also uses a coiled cancellation tube inside the case. The ER-7C uses a 0.5 mm ID tube with 0.97mm O.D., so it will fit in the smallest vent holes or around the earmold. In this case, the dampers are so tiny that the manufacturer who makes them calls them "no see ums," but this allows us to provide a flat response (with some simple electrical equalization to compensate for the attenuation of sound in small tubes) from 200 Hz to 10 kHz. We continue to provide these units for hearing research and hearing aid research.

MC: You really like Carlson's idea!

MK: Absolutely. We used basically his approach in the ER-3 insert earphone, which was our first product that anyone wanted. Indeed, we were looking at closing the doors until Nicolet placed a large order for ABR applications. With the ER-3 earphone we needed more power, but not as much fidelity because we only wanted to mimic the TDH-39 audiometric earphones. In that case we found we could produce more output (less loss) by using a "lumped element" version of the twin tube. Elmer has taught that you could use acoustic mass elements (tubing) and acoustic compliance elements (volume) to give a close approximation to the pure twin tube transmission line. The result looks like sausages (volumes) strung on tubes that "You are the most difficult company to evaluate I have ever seen. You are involved in almost everything. You are not just consumer products; you are not just diagnostics; you are not just hearing protection." But to answer your question, we don't make ABR units and have no intention to do so.

MC: And the most important question of all, in the K-Bass, the K-AMP, and the Digi-K, what does the "K" stand for?

MK: Before I answer that question, I want to state that it was the board of directors who urged me to use my own name whenever possible. It didn't take much encouragement, of course. The answer to all of those is "Killion," but in the case of the K-Bass aid Chuck Berlin was largely responsible for making such a hearing aid in the early 1980s, and so that might be considered the Killion-Berlin hearing aid.