From Soft to Loud: Understanding Compression in Amplification

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Hello everyone and welcome. Thank you for joining me today for this foundational topic from Soft to Understanding compression and amplification, aka a compression overview. My name is Dr. Natalie Hein and I'm a member of the education and training team at Starkey Housekeeping. Just to start off, I managed to catch a cold this weekend, so I do apologize about my voice. Other Housekeeping if you'd like to ask question, please do so by using the Q and A feature and I will do my best to answer it at the end of today's presentation.

Handouts will be available under your pending courses on Audiology Online or by clicking the link in the chat within the classroom. CEUs are available with a paid Audiology Online Unlimited CEU Access membership. After completing the course, you can go to your pending courses and take the multiple choice exam and then claim that ceu. And lastly, if you need any technical support, you have the option to call email or use the Q and A window for assistance. And then on this slide you can see the risks and benefits that may be associated with this course.

I'll give you a second here to review them.

And then that brings us to our learning outcomes. We have three learning outcomes for today's session. All right, you're going to identify the three clinical scenarios in which more linear fitting approaches may be beneficial. You'll be able to describe the difference between compression and expansion and then define compression and identify the types of hearing loss most likely to benefit from the use. And this course is going to be review for many, so it is not always the easiest topic, compression, especially when you're just learning it.

So many students, including myself back in the day, struggled with compression basics. So this course, along with the learning objectives that we just went over, is meant to be a basic overview. So compression is an important topic to review and refresh because it is fundamental in the way hearing aids function and so how compression characteristics are set within the hearing aids. Really, again, is that foundation to patient success, to the ability to hear and understand incoming speech and how pleasant that sound quality is for our patients? The efficacy of all our other advanced processing features, such as noise reduction algorithms, are all dependent on a strong compression architecture.

Again, for many this will be a review, but a good review.

The world of sound is quite vast. It's really amazing how dynamic the range of the human ear is, from the leaves rustling in the wind to friends and family conversing during a meal, all the way to the other end of the spectrum music and loud concerts like that Taylor Swift eras tour that caused small Earthquakes at some of the locations.

And the reason we're able to process that vast dynamic range that we do is because of the physiological functions of the cochlea. This is also the underlying reason compression is needed in hearing aid technology. First, the nonlinear function of the cochlea. When sound enters our cochlea, the intensity of the sound that reaches the outer hair cells is not quite the same exact intensity that is translated to the inner hair cells and subsequently up the auditory pathway. You can think about the outer hair cells as amplifiers that, when healthy, ensure soft sounds are amplified so we can hear them, but amplify moderate sounds to a different degree so that they are comfortable but not made too loud.

And then, of course, that we amplify loud sounds so that they are not painful. Not every sound that enters our cochlea is treated the same. It is amplified differently dependent on that input level. And that is what is meant by the nonlinear function of the cochlea. And without these nonlinear functions, we would not have that dynamic range that we do.

In addition to these nonlinear functions, tonotopic organization of the cochlea and central auditory processing system further contribute to the precision in which sound is processed by our auditory system and is responsible for the frequency specificity of that system.

Now, I just used the term dynamic range. And it can be defined as a difference between our loudness discomfort level, which is a level at which sound begins to become uncomfortable or even painfully loud, minus our hearing threshold. This illustration is demonstrating the dynamic range in a person with normal hearing. The blue bar represents the full dynamic range. The bottom of the bar is the threshold, and the top of the bar is that loudness discomfort level.

The white pieces of the bar represent the range of the environmental sounds, including those below our thresholds and we can't hear and those above the ldhels are the painful ones to us. But you can see that the dynamic range in this example really allows this person to hear a lot comfortably, including soft, moderate and intense sounds.

Looking at this in a way we are more accustomed to an audiogram, we see a person with normal hearing can have a dynamic range as wide as 110dB. This is indicative of a healthy inner ear. Those non linearities of the cochlea, and that tonotopic organization of the auditory system, how does this change when a person has hearing loss? In the next few examples, we will look specifically at how the dynamic range differs for a person who experiences sensorineural hearing loss, which is, as you know, that loss due to damage in the cochlear hair cells Compared. So here's that same picture that we saw before, but now you see, we're showing what dynamic range looks like in someone with that sensorineural hearing loss.

You can see it represented here in that blue bar once again. And one of the most common complaints associated with hearing loss is the inability to hear those soft sounds. And if we look at that bottom of the blue bar, we can see that the threshold has shifted. So that now what should be relatively easy to hear sounds is a strain to hear and weak sounds, they're not audible at all. It is important to note that the loudness discomfort level, that top of the blue bar has not moved.

So the point at which sound becomes uncomfortable in this particular example is the same, despite a loss of hearing sensitivity to soft and moderate sounds. So our range is reduced.

Relating this to that audiogram of a flat mild hearing loss, you see, in this case, the loudness, discomfort level has not shifted, although the threshold has in comparison to the normal audiogram shown just a few slides before, really illustrating that reduced dynamic range. What is happening physiologically is the outer hair cells due to damage now function in a more linear fashion, as we see in the dynamic range being reduced.

Another concern that we see for patients who experience sensor neural hearing loss is often the increased sensitivity to loud sounds. This is also related to the deconditioning of those outer hair cells. And when this occurs, our patients are often unable to tolerate loud environments such as concerts, in addition to being unable to hear those whispers of someone sitting next to them in church. In order to remedy this, we need technology to mimic that dynamic nature of the intact auditory system, which is extremely difficult task because it is really like squeezing that elephant into a suitcase.

And the goal of the hearing aid thinning then is to reproduce that full dynamic range of everyday sounds within this narrow dynamic range of the listener, so that the soft sounds are audible, the average sounds are comfortable and easily heard, and those loud sounds are loud but not uncomfortable.

And that is the bottom line and for what compression does and the goal of compression.

So let's think about what sounds are most important for adult listeners. When your patient comes in. We can think about this in terms of restoring the world of sound for hearing aid users. And in order to do this, we have to think about what is their goal of the patient when pursuing amplification. For me, it's always been being connected to the world around them, being connected to their loved ones, technology, all of the above.

So that is the most important sound to most patients. And it can include the scenarios of I want to hear my grandchildren better, I want to hear my husband when we're out to eat, I want to hear my dog bark without being startled and the take home message again. For adult listeners, hearing speech is usually the most important goal so that they can be connected to the world around them. And for this reason, speech has been the stimulus. Research has traditionally focused on when developing hearing technology, and because of this, hearing aid compression architecture has historically been configured around how to improve speech intelligibility and audibility.

The intensity, the spectral and temporal characteristics of speech must constantly be kept in mind.

So let's begin discussing signal processing approaches. This can also be referred to as fitting formulas.

Now, linear amplification is what came before the development of compression and nonlinear amplification. So sometimes we think of this as an outdated strategy. However, linear amplification is still used today when treating conductive hearing losses and plays a role in the treatment of some sensor neural hearing losses as well. A purely linear approach has its limitations when it comes to treating sensor neural hearing losses, but there are still cases in which a more linear strategy do apply. Today, the hallmark of a linear amplification is a fixed amount of gain regardless of the input level.

So here is an input output function of a linear hearing aid. The horizontal axis represents the input and the vertical axis is the output. The loop, the blue line is representing the gain being applied to the input sound for the resulting output of the hearing aid until you reach that dashed gray line at 110 dB, which is the maximum output for this example on this aid. So if you have that 30D input and you go straight up until you hit the blue line, you see the resulting output of this hearing aid will be 60dB. So a gain of 30dB.

If I do the same thing at 70 and I move up, I see the output of 100, that same fixed amount of 30db expressed as a ratio of gain to output. A linear system always has a one to one ratio.

The maximum output is the highest possible signal a hearing aid is capable of delivering. When an input signal exceeds this point in a linear hearing aid, the hearing aid goes into saturation and that signal is then clipped, also known as peak clipping.

So what is typically seen as the issue with peak clipping is when the signal is greater than the maximum output of the device, as shown here, the bottom is illustrating what it does cause distort.

Now there are some limitations that exist in regards to linear approaches. While a linear approach can make sound more audible, soft sounds more audible, average conversational speech is often regarded as loud and intense sounds are amplified beyond the upper end of the patient's dynamic range, increasing that potential for discomfort. Particularly again for those experiencing recruitment and peak clipping that could result in a distorted sound quality. So with this said, we also know that some patients do prefer a more linear fitting, particularly those that are long time hearing aid wearers or have been accustomed to previous analog technology.

So here are some cases where that linear approach does still apply. Think about that long time linear user, even if they have that reduced dynamic range and experience recruitment if they have already acclimated to linear technology and that is the sound that they have learned to listen to, even though there is distortion with louder sounds, that distortion of is how they have learned to process sound and how they know their hearing aids are working. When moving them to a more compressive system, they may feel that the hearing aids aren't working at all and they do not help them hear better. So sometimes there is those patients that continue to wear their older hearing aids even after getting those newer ones, or tell you that the new ones aren't as good as their old ones, they don't hear as much.

And then you also can consider the case of severe to profound hearing losses. Our primary concern is more about making sure they perceive any sound at all and that we are reaching their hearing loss. A linear fitting in many cases will be perceived as a louder fitting and provide more gain than most compression formulas. Not to say that all severe to profound hearing losses should be fit with a linear fitting formula. So we will be going over different compression characteristics that may be better and tailored for that population.

And then lastly, considering conductive hearing losses in which the cochlea is healthy and still operating in a nonlinear manner, Our objective is to overcome the blockage of sound energy that results in that hearing loss. These patients, they may not need compression and may not like the sound quality of a compressive formula. A linear fitting may be optimal in these cases because all that is needed is additional volume to overcome that loss.

And then that brings us to nonlinear approaches, which we also refer to as compression. So before we get into the specifics of compression, I do want to hit on some terms that are important to note, including threshold knee point, compression ratio, attack and release time and output compression limiting. So let's review.

We're going to start with the threshold knee point. You'll see it as TK in different books, different literature, all that jazz. Threshold knee point can be also referred to as tk. This is the compression threshold. The TK is the point at which the input signal is sufficiently loud enough for compression to begin to act.

Or in other words, the level in which the input signal volume has to be before compression kicks in, dependent upon the input the hearing aid receives, it is likely there will be multiple knee points employed so simultaneously to a signal.

So here is an input output function of a hearing aid utilizing compression. So the main difference, in short from the linear input output function I showed you before is it's not a fixed amount of gain. With compression gain varies depending on the input. The compression threshold is the point which compression begins at, as shown by the shifting point in the curve labeled tk. Each curve represents a different hearing aid with a different compression threshold.

Before the TK or compression threshold is reached. Gain is linear, aka fixed, regardless of the input at a 1 to 1 ratio. And once we reach that TK threshold knee point, the function bends as compression has kicked in and we no longer have that one to one ratio.

We do have some considerations regarding threshold knee points, and that is, with a higher threshold, the knee point, the signal processing will remain linear for a longer amount of time. And this may be beneficial for the more severe hearing losses and those who have worn hearing aids previously. With a lower threshold knee point, the signal processing becomes compressed earlier and therefore the signal processing is in a compression over a longer period of time. And this might be beneficial for those users experiencing recruitment or a severely reduced dynamic range and then compression ratio. So this you'll also see as cr.

This refers to how much the signal is compressed and that input output function shown before, how much the line it bent after that threshold knee point linear amplification follows that one to one ratio. Since that gain is fixed, a compression ratio is anything greater. So the one to one ratio or greater or you'll see greater than one.

And we're going to get back into our input output function. So below the threshold knee point you see, again we have that one to one ratio linear because input has not exceeded. So I'm seeing a few notes coming through and I go back, I got distracted, I apologize. That. Okay, so let's get back to our input output.

We've got that function. And then again, below the threshold knee point, you see, we have a one to one ratio. It's linear because input has not exceeded that threshold knee point yet. And above the threshold need point we have a compression ratio of 2 to 1. So every 2dB of change in input results in a 1dB change of output.

If it were 3 to 1, every 3dB change in input would result in a 1dB change to the output.

Now we've talked about the variability and compression ratios between linear and nonlinear. We can also think of compression ratios in terms of low and high. The lower the compression ratio, the less compression is being implemented. A low compression ratio is typically defined between 1 to 3. Compression ratios at this level are typically implemented with the goal of improving audibility of the softer components of speech and restoring that loudness perception.

This may be beneficial for those experiencing that mild hearing losses or even sometimes that new hearing aid wear.

Some things to understand in how compression ratios are set. The lower the ratio, the more audibility you have for those softer components of speech without moderate and loud sounds being too loud.

Sorry, I'm seeing all the notes coming through. I'm going to pause for a second.

And Kimberly, Audiology Online team Do you guys have a preference? What would you like me to do? Because I see things about being joining late.

And I do want to note for those that are joining late, this is being recorded so it can watch those first few minutes. I think I saw a few parts of that. So the first 17 minutes. Okay, So we will. I'll.

We'll find out from Audiology Online what is the preference on the entry point. So I do not have control of the entry point, but I know that this is being recorded so I'm going to jump back in to be respectful of their time to end on time. But we are currently hitting on compression ratio considerations, right?

Not getting any audio. So hopefully others are getting audio.

I mean a lot come through the Q and A right now. I do apologize.

So conversely, a high compression ratio may to be defined as anything above a 3 compression ratio 3 and while this may help fit more sounds into the patient's dynamic range can often lead to a perception of distortion. That being said, a limitation of low ratio is that softer non important environmental sounds may be amplified too. High of a compression ratio though, can adversely affect the clarity and the pleasantness of that amplified sound.

The next component of compression that is important to understand are attack and release times. This refers to the amount of time it takes for a compression circuit to respond to the intensity changes of that input or environmental sounds. In simpler terms, this refers to how quickly compression kicks in once a certain input level is detected and how quickly compression turns off after a certain input level is detected.

Attack times tend to be set faster so that an individual does not have to listen to sounds louder than desired for too long. Because the faster the attack time, the faster the compression will act on that louder signal. At the same Time, you do not want it too fast, or it can negatively impact the sound quality.

Release times are typically longer to ensure variability in that input signal does not negatively impact the process signal. Too long of a release time can negatively affect speech intelligibility after a transient or a brief loud sound. For example, if someone begins speaking to the hearing aid user immediately following a door slam. Or in my scenario, my husband loves to slam cabinet doors. And then, of course, how does the volume control interact with the compression setting?

This is a question that occasionally comes up, especially for students that I work with and new providers. How the volume control interacts with compression characteristics. Compression characteristics may be independent or dependent on volume, again, depending on that signal processing strategy. So let's take a quick review of the basic components of a hearing aid. You have that microphone which picks up the input signal and goes through an amplifier to be sent to the receiver to deliver that output process signal to the ear canal.

And this system does need a power source, so it is powered by a battery.

Now, we have not quite discussed why dynamic range compression yet. We will be momentarily. But it is the most widely used compression system today, and it utilizes an input compression circuit. In an input compression circuit, compression acts before the volume control, meaning that the threshold knee point cannot be affected by the volume control changes. Which is good, because if the threshold knee point were to change that significantly, it changes how the hearing aid sounds.

And if a patient is complaining about sound quality, but you know that they manipulate their volume control often, it can be very challenging to figure out why.

In output compression, the compression acts after the volume control. So volume control changes can affect that threshold knee point. Now, the negative in this is that the patient can make changes to compression that negatively affects sound quality, as I said before. But an upside is if the patient, let's say, raises their volume a lot, and then a loud sound suddenly occurs like a pan dropping. In this case, because the compression acts after the volume control, it will not be as unpleasant if it did not.

That sound may be too loud. So compression limiting in a device uses compression to control or limit that loud sound from being too loud without causing distortion. Unlike the case with peak clipping, it is characterized by high threshold mutual and high compression ratios to provide as little amplification as possible while keeping it still sounding natural. So how would this be employed in a hearing aid? Is this.

This may very well be the compression characteristics of the mpo, that maximum pressure output or max output of the hearing aid to control those louder sounds without causing distortion.

Yeah. And then that brings us to wide dynamic range compression, WDRC and expansion.

When we think of dynamic compression, we need to think of it in terms of time constants. Time constants related to changes over time or changes in compression throughout the duration of signal processing. Faster time constants have the ability to compress things more. This may be beneficial to someone that apologize got another note coming through, someone experiencing a small residual dynamic range, but conversely may contribute to distortion of speech sounds and lead to that reduced speech intelligibility. Slower time constants have the ability to be less compressive and may be advantageous for those experiencing cognitive impairments.

Now, in the wide dynamic range compression, the hearing aid provides different degrees of amplification depending on that input level. Softer sounds are given more amplification than louder sounds. Again to account for that reduced dynamic range in which usually soft sound is no longer heard, but loud still remains loud. So by using this, we are doing our best to reproduce a full dynamic range into the world of reduced dynamic range listening. Remember that elephant in the suitcase?

What sets that wide dynamic range compression apart from what we've been discussing regarding compression earlier is a handful of things, right? So expansion and limitations of compression. I mentioned earlier that the limitation of low threshold knee points and low compression ratios is that sometimes those unwanted environmental sounds can be over amplified. Also, in the case of someone with normal low frequency hearing, they may be able to hear low frequency circuit noise coming from the hearing aid really just sitting so close or even inside their ear. In short, a limitation is because of what it benefits.

The weak sounds are amplified to a greater extent than louder ones, improving speech intelligibility. But then unwanted sounds are made audible as well. Expansion works to counteract one of the drawbacks of compression that in the process of making soft speech audible or weak unimportant sounds can be made too loud. And when describing how expansion works, it will sound like the opposite of compression. So keep in mind that they do work together.

They do not cancel each other out or work against each other. The principle behind expansion is soft sounds are amplified less than louder sounds to ensure you do not provide gain below a certain input level. Since it's only below a certain input level, it it only applies to the weakest sounds in the environment and not speech ideally. So it is applying these different amounts of gain dependent on that input level.

And this graph here, it does represent an input gain function so different from the input output functions I have been showing throughout this talk. The blue line represents the gain in a linear hearing aid. So that's why you see it As a flat line at 30 decibels, because that is being applied regardless of the input which values across that horizontal axis until it reaches its maximum output. You see right there at 80 dB. And then I have an orange line represents how expansion and where it meets the blue line is that expansion threshold knee point, which I will be defining further on that next slide.

What is important to note here is that below that point labeled tk, as your input level is reducing, so the gain being applied.

Now, the expansion threshold is the point below it, right? Expansion operates below, Unlike with compression ratio, in which compression operates above, with expansion it operates again below the threshold. Sometimes these thresholds are one in the same. So below this specific threshold is expansion and above it is compression. In some systems though, there are two separate thresholds.

And then you do have different considerations when it comes to expansion thresholds. There's pros and cons to having low as well as high knee points. And you can see some of the pros of low ensure speech audibility. At lower levels. The con would be you still could have noisy hearing devices.

If it's high, you have a pro of quiet hearing devices. The con, it could negatively impact speech audibility. So considering this, depending on the listening needs and preferences of of your patient wearing the hearing aids, they might prefer a higher or lower expansion threshold. So it is always good to know when fitting a particular device, how the expansion is set and what adjustability you have as the professional to tailor it to your patients and to minimize or expand any unwanted circuit noise, low level, environmental hymns, hums, bips, whatever. Similar to the compression ratio, it is the change in input over the change in output.

So just to review on specific numbers, when discussing ratios, a one to one or one ratio is linear. Anything greater than one ratio is compression. So expansion would be less than one ratio. So anything between zero and one.

To think about the expansion ratio in terms of its more or less expansion, the closer the ratio is to zero, the greater the expansion or the more aggressive the expansion. Greater expansion means a quieter hearing aid, good for sound quality and quiet environments. But if it's too low, it can negatively affect that soft speech audibility. The closer the ratio is to one, on the other hand, the less expansion or less aggressive expansion, which certainly protects soft speech audibility. But the noisier the hearing aid and the more likely circuit noise can really just bother the users.

The goal clinically of expansion is to keep the hearing aid quiet in quiet environments. So again, depending on the patient and their audiogram and listening preferences is whether a higher or lower expansion ratio would be beneficial.

Now, looking at an expansion ratio on an input output function, the steeper the slope, the greater the degree of expansion, the closer that ratio is to zero, less steep of a slope, the closer the ratio is to one less expansion.

Now this I started to mention before. Typically the expansion ratio is separate from the compression ratio in hearing aids because that is a lot of constant processing and technically distortion occurring if they start at the same threshold. Although in some systems again that is the case.

And then that brings us to our applications of compression. Those channels and bands and prescriptive formulas. It's a mouthful today for me. Now terminology dates back to analog hearing aids when a channel was referring to a frequency band, meaning the frequency bands were separated by filtering from other frequency bands. Back then, only gain and subsequent output could be adjusted.

With digital technology we know more than gain can be adjusted, for example, compression characteristics and other advanced noise management features.

Now, in general shape of the frequency response of the hearing aid to accommodate loudness tolerance. So a band is where frequency specific gain adjustments can be made and how the software better tunes the frequency response of the hearing aid to that patient's hearing loss. And depending on how many bands you have, for example, you could have 16, you could have 20, or you could have 24 for the more tailored or personalized fitting that can be for your patient.

Now let's distinguish channels from bands. A channel is a range of frequencies where advanced signal processing adjustments take place. In other words, channels are the functional units where noise reduction algorithms take place and work towards a collective goal of providing precise noise control for any environment. So think bands are gain. Channels are your other advanced features.

Noise reduction algorithms are criterion based systems. In other words, incoming signals must meet criteria for a response to occur. The ability of these systems to accurately identify speech and noise is dependent on the specific nature of the sounds in the acoustic environment. So all the parameters you see here are being reevaluated in milliseconds and updates made in milliseconds. So that means each of these parameters is potentially resetting over 50 times each second in each channel for that most accurate response possible.

And the more channels you have, whether it be 8 or 12.

Or 16, the more accurate of the noise spectrum in an environmental can be captured for those hearing aids to make subsequent decisions on which advanced features to implement at any given moment. Especially now with the use of deep neural network processing with Starkey products for sure.

Now we'll get into prescriptive. Still having trouble with that word today. Fitting formulas that really dictate how hearing technology sounds and control how compression and expansion thresholds and knee points are set and is the basis with all other advanced processing or noise control features operate on prescriptive fitting formulas are also known as the compression architecture of a hearing aid. And there are different types. There are proprietary fitting formulas that are manufacturer specific and developed by the manufacturer.

For example Starkey. Starkey's proprietary fitting formula is estat 2.0. And then there are the more generic or commercially available fitting formulas such as such as NLR, NL1, 2 and now NL3 or DSL.

Now, NALR was one of the first standardized evidence based fitting formulas. It is a linear fitting formula of which many future nonlinear compression formulas are based upon and utilize a loudness equalization rationale. This is all frequency bands of speech will be amplified equally and equally contribute to the overall loudness of speech. Now let's think back to that fixed amount of gain despite frequency and input level. Everything again is treated equally.

NAL NL1 is a compression formula that was developed based off of NALR. Now prior to NL1 there were the development of some nonlinear formulas. What's the limitations of NAL are and how to remedy them and remedy them were being explored and there was a lack of regulation over these early nonlinear formulas that were rarely evidence based as well. So NALR being an already validated fitting formula was a great place to start for NAL1 became the first of the validated nonlinear formulas again based off that NALR. Now the fitting rationale of NL1 was different since it is a completely different strategy.

It uses and utilizes a loudness normalization rationale. So overall sound normalized so that loud sound is loud, soft is soft, et cetera. Relative loudness of different frequency components preserved rather than just emphasized equally theoretically is maximizing sound speech understanding by applying this type of gain frequency response and then also how gain is applied in this formula is based on the speech Intelligibility Index.

So NL2 was created based on the feedback and empirical data collected over a decade using NL1. It is also a wide dynamic range compression formula. And the rationale was the same as you just saw in NL1. It utilized very low compression ratios with the goal of restoring low level speech audibility, but also caused its main complaint NL1 was too loud, especially for those softer inputs. And professionals hearing care professionals found they were constantly having to take the gain down.

And that brings us to the latest greatest if any of you were at AAA Last year in 2025 you saw the release of NL3. Mike have gone to that talk I did as well. This is the cumulation of nearly 50 years innovation in hearing aid prescriptions at NL. So NL1 again back in 1999 introduced that fitting strategies based on loudness equalization and speech intelligibility for Nonlinear hearing aids. NL2 was 2011.

It incorporated that user specific variables such as gender, weight, language, hearing aid experience and even cultural preferences. And in 2025, NL3 that is powered by machine learning. So large scale data, real world user input for their most personalized and precise fitting formula yet.

And then of course we have developed in 1980s for the Pediatric Fittings DSL. Its prescribed targets considering the variables in that pediatric assessment and fitting itself, such as differences in ear canal characteristics. Essentially, the latest version of DSL employs extended protocols and more flexible targets based off the information provided about that child. So whether they are an infant or an older child, the etiology of the hearing loss utilizes data available. Since oftentimes we don't have everything, especially when fitting that young child.

This rationale is more closely resemblance of a loudness equalization rationale like that nlr because with children depending on the age, they're still developing their speech and language and need to hear as much as possible. So why what is the difference about pediatric hearing aid fittings versus versus adults? And that's incidental learning. I'm sure you guys all know that, right? Children need to be able to hear all sounds in the environment.

Not important to prioritize some sounds over others for comfort like we do with adults. Children, they need to hear it all right? So also sophisticated processing features are not employed in nearly the same quantity as they are with adult hearing aids as we do in pediatric.

So what does this mean for a Starkey hearing aid with Omega AI hearing aids? That is our latest family. The past few families have been utilizing two independent compressors that run in parallel and calculate gain independently based on the measure of input. So the two gains are then added together. Hence the name additive compressor.

So even though this compression system uses two compressors, we're not calling it dual compression or twin compression. The legacy twin compressor system featured a compressor for speech and a compressor for music. However, the compressors did not run in parallel, meaning patients were either using speech architecture or music compression architecture, so they ran separately. Also, the twin compressor technology used a single set of time and attack release times, which define how quickly the hearing aid goes in and out of compression. The new additive compressor has two sets of attack and release times, one set for slow compressors and an attack release time for that fast compressor.

So the key driver behind the new compression system was to leverage the benefit of two compressors that each process sound differently, with the goal to achieve better perceptual outcomes then we've been able to do before. And this shows our empirical evidence is really supporting this.

So our slow compressor accounts for the audiogram and ensures that the environmental input is within that patient's dynamic range. This means that soft sounds are soft but audible, average sounds are comfortable, and those loud sounds are just not too loud. The fast compressor operates on the dynamic, fast moving broadband components of that input signal. So it's providing the better audibility for those soft components while reducing the contrast between the louder and quieter portions of that speech signal. This means more gain for soft sounds like those consonants.

And then the additive compressor is really that perfect balance of comfort and clarity. Using that additive compression architecture again, I'm going to repeat myself, has both fast and slow compression logic running independently and in parallel, which is later combined. The fast compressor, again operating on that dynamic, fast moving broadband compressor, accounts for the audiogram and ensures the environmental input is within the patient's dynamic range. Meaning soft sounds are soft but audible, average sounds are comfortable and loud sounds are just not too loud. That slow compressor operates on the components of an input signal, providing better audibility for those soft sounds.

And again, getting the benefits of both worlds right. We want the slow and the fast. Now we hit on Starkey's fitting formula. Fitting formulas are not the same for all manufacturers. And here you can see the fitting algorithms are used in programming for all hearing aids.

While universal fitting formulas like NAL, NL2 and DSL for example, were developed based on trends in amplification to be used in the fitting of technology by multiple manufacturers, they do not account for manufacturing specific approaches to signal processing. So here you can see a relay response using NL2 with hearing aids from four different manufacturers. Using the same hearing loss receiver matrix acoustic options, you can see the differences in the response As a result of each manufacturer's unique approach to signal processing. Equivalent hearing aid responses are not possible.

Now, with Starkey's compression architecture in mind, I do want to highlight some programming scenarios that may need special attention from a compression standpoint. So we're going to go into two of them. But just to give us a starting point, here we're starting with a 2cc coupler view and profit of soft, moderate and loud input curves all overlaid on each other. And I know this is not how we typically look at things, but I think this view can help solidify what I'm discussing. So, generally speaking, if the patient is restoring that specific environment is too soft and you want more compression for soft sounds and less for the loud, you can see here when we increase the gain for the soft inputs but not touching other input curves, that increases the compression ratio for the adjusted frequencies, which is what we're looking for.

On the flip side, if a patient is reporting an environment is too loud, then you want less compression for soft sounds and more compression for the loud. So by dropping the loud input curve, we are increasing that compression ratio.

Now let's put this into practice for another specific patient complaint. This patient is reporting trouble with comfort and clarity in a restaurant environment. If you think about the restaurant acoustics, it is generally a loud environment. So we as people naturally raise our voices in an attempt to overcome that environment. If you are trying to program for comfort by dropping the loud input curve, this will provide that comfort to patient is looking for, but could smush speech as a result in poor clarity.

So instead consider looking at the soft input curves and having the aids be linear between 150 and 750 Hertz and between 4000 and above. There is not much usable information from a speech standpoint within these frequency ranges in a restaurant. So by going more linear for these frequencies and providing compression with that speech focused frequencies of 1000 to 3900 Hz, we can provide more comfort without sacrificing clarity.

Now, typically in H VAC systems are above expansion but below that knee point. So the quiet algorithm might not touch these. These machine algorithms could touch this one. But if you still need to make adjustments from frequency standpoint, the machine is reducing the noise across that entire frequency range. So as soon as speech becomes present, it disengages and prioritizes speech.

This leads to that gain fluctuation from quiet to loud, with the speech focus back to quiet again. Pardon me. If you do target things from a compression standpoint, then you will get less overall environmental gain fluctuations. So instead consider looking at the soft input curves and having the aids be linear between 150 to 750 Hz and between 4000 and above, providing that compression again within that speech focus. This will allow for the sounds from the H VAC system to be more better managed from a compression standpoint without that large gain fluctuation.

And will also allow for speech to maintain that clarity and a stronger sound presence for that hearing aid wearer.

Now, resources, we have a ton of of different resources available to you. Whether it is that handy tech book which I'LL talk about momentarily. We have brochures, we have white papers, and then of course the Starkey Pro website. You'll find a wide range of tools. Everything from our product information, fitting guides, quick reference materials, and again, that compression handbook that I'm sure many of you guys are familiar with.

I had it in school and we're on our fourth generation. And again, this is designed to be a one stop hub. So if you are on the Starkey Pro website, you can see I'm showing you now the search box. It's my favorite thing. Whether you want to learn about tinnitus, see some white papers or compression to have that handbook populate, you can do so there.

For patients. They can access Starkey.com and they can find important quick tips and videos to support them. And just as a reminder, I do want to say that you do have your whole Starkey team available. You're not in this journey alone. Whether it's a sales rep, technical support, education and training, we're here to answer any questions you might have, troubleshoot, brainstorm best practices with you.

So do think of us as an extension of your clinic team. We're here to help you feel confident and support you as you integrate these tools into your patient care. Whether it is calling Starkey and having audiology staff help or going through the PROFIT software and having audiology on demand literally come in and join you in your computer for that fitting. So here are some reference. I do love to highlight these references.

Some of them will sound very familiar to you all, whether it's the compression handbook, compression for clinicians, hearing aids, and then that compression handbook I mentioned. We're on our fourth edition. I have a QR up here on the screen so that you can utilize it and download it. Now again, we're on our fourth edition. I've seen many presentations utilize the diagrams, the images, and not even Starkey presentations for that feature and it's very helpful, has great reminders and of course if you are new to the field, it is a great tool to utilize.

If any students are listening today, I highly recommend utilizing the compression handbook.

And that is my time for today. I'm going to go and check in for any questions that there might be. I know that there was a lot of chatter, so bear with me as I go through because I know there might have been some technical issues. And then while I do that, I'm going to go back to the reference slide so that you can see that QR code once again. If you do not have your phone out in front of you.

Bear with me. I'm going to pull up this chat. Okay.

And go back to the Q and A. I guess the audiology online team cleaned up that chat. So there was 20 plus, but now I just see one. So now going into the Q and A, we have technical issue. Yep, technical issue. Waiting room issues.

Still waiting room issues. So I'm hoping I did see something come through from Kimberly about the technical issues. This is recorded. I do apologize if there were any technical issues. You can review those first 9 minutes or 17 minutes.

I see a few different comments of what was missed. And do not hesitate to reach out to your Starkey team. We're happy to help. And again, this handbook that is on the screen right now is a lifesaver. If you have any interns externs, I highly, highly recommend it can be found in the printed form if you need it as well.

Okay, so that is my time for today. Again, thank you guys for joining me during your lunch hour. And in conclusion, you hopefully now have a better understanding of compression and compression characteristics and have obtained some good ideas on how and when to modify those fitting algorithms based on the needs of that individual patient. We always have to be listening to our patients. So again, thank you for your time and I do hope you have a wonderful rest of your day.