Compression Overview:
The Foundation of a Dynamic Fitting
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- Hello, everyone, and welcome. Thank you for joining me for today's course, The Foundation of a Dynamic Fitting, which is a compression overview. My name is Melissa Jordan and I'm a member of the education and training team here at Starkey, and I'm happy to be with you today to talk about this important topic of compression. As I said, it's mostly an overview, a few little historical things sprinkled in here and there and how we're using it today in our hearing aids, but just a good refresher course for most of you, I'm assuming. Before we get started, just a few housekeeping notes to take care of. If you would like to ask a question, please do so by using the Q&A feature, and I will do my best to answer it at the end of today's presentation, time allowing.

If I don't get to your question, I'll still certainly go in and look at it, and I can perhaps email the answer to you. Handouts will be available under your Pending Courses on Audiology Online, or by clicking on the link in the chat within the classroom today. CEUs are available with a paid Audiology Online Unlimited CEU Access Membership. So after completing the course today, 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 the 800 number that you see on your screen there, or you can email customerexperience@continued.com, or feel free to use the Q&A window for assistance and somebody will reach out to give you the help that you need.

We have three learner outcomes for today's session. The first one is that we would like you to be able to define compression and give an example of what type of hearing losses are most likely to benefit from uses of compression. The second learner outcome is that we would like you to be able to describe the difference between compression and expansion, important difference there. And thirdly, we'd like you to be able to identify three scenarios where a more linear fit would be beneficial to a patient. The beginning of the course today will be more of a basic review, like I said, of the purpose of compression. We'll then be going into the various signal processing approaches. Linear and non-linear, of course.

We will spend some time discussing wide dynamic range compression in particular as well as expansion, and then how channels and bands fit into the whole picture. And then towards the end, we'll dive into prescriptive fitting formulas, including taking a look at Starkey's proprietary formula and compression architecture. So again, my name is Melissa Jordan, and this is The Foundation of a Dynamic Fitting. Thank you for joining me. So what is compression? Compression is a topic that is not always the easiest to understand. Many times, students will struggle with compression basics in graduate school and even for professionals who fit amplification every day. This topic can sometimes be confusing in terms of how to apply compression to a fitting and how and when to make adjustments to compression settings.

This course, along with the learner objectives that we just went over, is meant to be a basic overview for new professionals or even students who may be listening or even experienced professionals who are joining me today who really want a refresher of the basics. We'll talk about the terminology and how compression is involved in the software and hearing aid fittings. You'll find all of that here today within the next hour. Compression is an important topic to review and refresh, because it is fundamental to the way hearing aids function. How compression characteristics are set within a hearing aid really is the foundation to patient success, to the ability to hear and understand incoming speech, and also how pleasant the sound quality is for that end user.

The efficacy of all of the other advanced processing features such as noise reduction algorithms are all dependent on a strong compression architecture. If we think about what our goals are when fitting a person with hearing technology, we wanna restore the world of sound as best we can for that individual with the hearing loss. The world of sound is quite vast, as we all know, and it's amazing how dynamic the range of the human ear is, from the ability to hear the slightest rustling of the leaves outside in the wind, to family and friends conversing over a meal, even moving into loud concert venues where we have music blasting in our ears for a few hours at a time.

So just really wanting to make sure that we are giving each of our individual patients what they're looking for to get out of their hearing aids and what their goals are for better hearing. How are we able to hear all of those sounds? The reason we're able to process the 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. Let's examine the non-linear functions of the cochlea. When sound enters our cochlea, the intensity of the sound that reaches the outer hair cells is not the exact same intensity that is translated to the inner hair cells and then 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 amplifying moderate sounds differently so that they're comfortable but not too loud and amplify loud sounds so they are not painful. So not every sound that enters our cochlea is treated the same. It is amplified differently depending on the input level. This is what is meant by non-linear functions of the cochlea. And without these non-linear functions, we would not have the dynamic range for hearing that we do. In addition to these non-linear functions, tonotopic organization of the cochlea in the central auditory system further contribute to the precision in which sound is processed by our auditory system.

This is also responsible for the frequency specificity of the system as well. Now, I've been using the term dynamic range to describe how dynamic our range of hearing is, but technically speaking, dynamic range can be defined as the difference between our loudness discomfort level, which is the level at which sound begins to become uncomfortable or even painfully loud for some, minus our hearing threshold. This is an illustration 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 the LDL. The white spaces in the bar represent the range of environmental sounds, including those below our threshold that we can't hear, and those above our LDL that could be potentially painful to us.

But you can see that the dynamic range in this example really allows this person to hear a lot of sounds comfortably including soft, moderate, and intense sounds. Looking at this in a way we're a little more accustomed to on audiogram, we see a person with normal hearing can have a dynamic range as wide as 110 dB. This is indicative of a healthy inner ear utilizing those non-linearities of the cochlea and the tonotopic organization of the auditory system. So how does this change when a person has hearing loss? In the next few examples, we'll look specifically at how dynamic range differs from a person who experiences sensory neural hearing loss, which is loss, as you all know, that's due to damage of the cochlear hair cells.

Here's the same picture we saw before, but now we're showing what dynamic range looks like in someone with a sensory neural hearing loss. You can see that represented here in the blue bar. One of the most common complaints associated with hearing loss is the inability to hear soft sounds. If we look at the bottom of the blue bar, we can see that the threshold has shifted, so that now what should be a relatively easy to hear sound is now a strain for this particular individual to hear. And weak sounds are not audible at all. It's important to note that the LDL at the 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 the range here has been quite reduced for this individual. Relating this to an audiogram with a flat mild hearing loss, you can see here, the LDL again has not shifted, although the threshold has, in comparison to the normal audiogram shown before. And again, this is illustrating a reduced dynamic range. What is happening physiologically is the outer hair cells due to damage now function in a more linear fashion. And as we can see, the dynamic range is sacrificed or reduced. Another concern we see for patients who experience sensory neural hearing loss is often the increased sensitivity to loud sounds. This is also related to the deconditioning of the outer hair cells, and when this occurs, our patients are often unable to tolerate a loud environment, such as a concert or a big crowd, in addition to being unable to hear the whisper of someone sitting next to them in church or somebody just wanting to talk quietly to them.

In order to remedy this, we need technology to mimic the dynamic nature of the intact auditory system, which is an extremely difficult task to do, because it's like the old adage, we've all heard it before, nothing new here. It's like trying to squeeze an elephant into a suitcase. The goal of a hearing aid fitting then is to reproduce the full dynamic range of everyday sounds within this narrowed dynamic range of the listener. So that soft sounds are audible, average level sounds are comfortable and easy to hear, and loud sounds are loud but not uncomfortable. And that really is the bottom line in what compression does and the goal of compression, being able to make soft sounds audible to the patient or to the user.

Loud sounds should not become uncomfortably or painfully loud to the user. And we really want speech to remain at a nice comfortable level, since that is what most hearing aid users are looking to perceive the best and are the most important to them are those conversations. Next, let's think about what sounds are most important for adult listeners. We can think about this in terms of restoring the world of sound for the hearing aid user. And in order to do this, we have to think about what is typically the goal of the patient when pursuing amplification?

In other words, what are the most important sounds to your patients? These might include somebody saying, "Well, I wanna hear my grandchildren better." Or, "I really wanna be able to hear my husband when we go out to dinner to our favorite restaurant. It's been a while." Or some people might even just wanna hear their dog bark without being startled or having it be painful to their ear when they're not expecting it.

The take home message is that for adult listeners, hearing speech is usually their most important goal. For this reason, speech has been the signal of interest that research has traditionally focused on when developing hearing technology. That makes sense, right? Because of this, hearing aid compression architecture has historically been configured around how to improve speech audibility, Intensity, spectral, and temporal characteristics of speech must constantly be kept in mind in order to achieve that goal.

Here, we can see a brief history of compression development, the research on how to approach applying gain and a hearing aid has developed over the years and continues to evolve today. How we approach signal processing currently is a product of what we have done as an industry over the past several decades. So let's take a moment to discuss some of the other theories that have been utilized since it really was not until about the late '80s, early '90s that we start to see the specific fitting formulas that are available to us today. As early as the 1940s, researchers were exploring frequency specific gain application. And the very popular at the time, half-gain rule.

Now there are several versions of this rule, but in general, this is when you take half the value of the hearing loss at a particular threshold, and that is the amount of gain you apply at that frequency. There were also schools of thought developed around the same time period stating frequency-specific gain was not important, and regardless of Audiogram configuration, a flat frequency response would do the trick. Excuse me. This research though was based on patients with conductive hearing loss rather than sensory neural hearing loss, which as we know is a much more complicated situation than just giving the person additional volume. Going forward into the '70s, there was support for comparative hearing evaluations in which you would just try different hearing aids with different frequency gain responses, in whichever one yielded a high speech recognition score, was the hearing aid chosen for that patient, for that fitting?

Now, I'm sure you can assume this leaves a lot of room for error. There is a reason we do not do this anymore, because obviously, depending on room conditions, the specifics of the word list being used and other compounding variables, this technique might not be very accurate. But it was an attempt to focus on what was most important - restoring speech. It was in the late 1970s where the development of research-based prescriptive formulas began and led us to where we are today. So let's move forward now into signal processing approaches, which are also referred to as fitting formulas. Now, linear amplification is what came before the development of compression and non-linear 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 sensory neural hearing loss as well. A purely linear approach has its limitations when it comes to treating sensory neural hearing loss, but there are still cases in which more linear strategies do apply today. The hallmark of linear amplification is a fixed amount of gain regardless of input level. 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 orange line represents the gain being applied to the input sound for the resulting output of the hearing aid until you reach the dash blue line at 110 dB, which is the maximum output of this particular hearing aid.

So if you have a 30 dB input and go straight up until you hit the orange line, you see the resulting output of this hearing aid will be 30 dB, or I'm sorry, will be 60 dB. So a gain of 30 dB. If I do the same thing at 70 dB and move up, I see the output is 100 dB. That same fixed amount of 30 dB of gain. Expressed as a ratio of gain to output, a linear system always has a 1:1 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 the signal is then clipped, also known as peak clipping shown here.

So what is typically seen as the issue with peak clipping is when the signal is greater than the max output of the device, as shown in the bottom illustration there, it causes distortion for the user, as the sound goes towards the maximum output of the device. Some limitations exist in regard to linear approaches. While a linear approach can make soft sounds 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 the potential for discomfort, particularly for those experiencing recruitment. And peak clipping may result again in a distorted sound quality for the end user. With this said, we also know that some patients do prefer a more linear fit, particularly those who are longtime hearing aid wearers or have been accustomed to previously or to previous analog technology.

So here are some cases where linear approaches still apply. Think about again those longtime linear users. 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've learned to listen to, even though there is distortion with louder sounds, that distortion is how they've learned to process sound and how they know that their hearing aids are working. When moving them to a more compressive system, they may feel the hearing aids aren't working at all and do not help them hear better. So sometimes, there are those patients that continue to wear their older hearing aids. I know we've all heard that. Those patients that continue to wear their older hearing aids even after getting new ones, or tell you that the new ones aren't as good as their old ones are.

That's because they've acclimated to a different way of listening. When considering the case of severe to profound hearing losses, our primary concern is more about making sure they perceive any sound at all and that were reaching their hearing loss. Excuse me. 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. That's not what I'm saying at all. And we will be going over different compression characteristics that may be better tailored for this specific population. And lastly, considering conductive losses in which the cochlea is healthy and still operating in a non-linear manner, our objective here is to overcome the blockage of sound energy that results in a hearing loss.

These patients may not need compression and may not like the sound quality of a compressive formula. A linear fitting may be optimal in these types of cases, because all that is needed is additional volume to overcome that conductive loss. That brings us into our non-linear approaches, which we also refer to as compression. So again, since this is reviewing the basics in this course today, I wanna clarify some definitions or terms surrounding compression before relating it to software and hearing aid fittings. Some of the terms that are important as we take on this discussion are threshold knee point, compression ratio, attack and release times, output compression limiting, slow-acting compression, fast-acting compression, the whole gamut kind of there.

So let's take a look first at threshold knee point. Threshold kneepoint, also known as TK, is known as the compression threshold. The TK is the point at which an input signal is sufficiently loud enough for compression to begin to act. Or in other words, the level at 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 in any given hearing aid, or potentially multiple kneepoints in a given hearing aid. 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 just a couple of minutes ago is it's not a fixed amount of gain that you're seeing here anymore.

With compression, gain varies depending on the input. The compression threshold is the point which compression begins as shown by the shifting point in the curves labeled TK. Each curve represents a different hearing aid with a different compression threshold in this example. Before the TK or compression threshold is reached, gain is fixed. It's linear, it's fixed, regardless of the input at a 1:1 ratio, just as we defined earlier. Once we reach the TK, the function bends as compression has kicked in and we no longer have that 1:1 ratio. It's starting to change. Some considerations regarding threshold kneepoints are as follows. With a higher TK, the signal processing will remain linear for a longer amount of time. This may be beneficial for more severe hearing losses, and again, for those who have worn hearing aids previously.

With a lower TK, the signal processing becomes compressed earlier, and therefore, the signal processing is in compression over a longer period of time. This situation may be beneficial for those users experiencing recruitment or a significantly reduced dynamic range. Now, we've mentioned the term compression ratio several times now, and this refers to how much the signal is compressed. So in that input/output function shown before, how much the line is bent after the point where the TK is. Linear amplification, again, follows a one-to-one ratio since gain is fixed. A compression ratio then is anything that is 1.1 to 1 or greater. Let's go back to our input/output function. Below the TK, you see we have a 1:1 ratio linear because input has not exceeded the TK yet.

And above the TK, we have a compression ratio of 2:1. So every two dB change in input results in a 1 dB change of output. If it were a 3:1 ratio, then every 3 dB change in input would result in a 1 dB change in output. So just a little bit of a pattern there that kind of demonstrates how and when that changes as a function of input. Excuse me. We've talked about the variability and compression ratios between linear and non-linear. 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 as less than 5:1. Compression ratios at this level are typically implemented with the goal of improving audibility of the softer components of speech and restoring loudness perception.

This lower compression ratio may be beneficial for those experiencing milder hearing losses, as well as perhaps some new hearing aid users trying to get used to hearing sounds again. Higher compression ratios are considered to be greater than 8:1, and they offer protection from loudness discomfort levels. And they may be more beneficial for severe to profound hearing losses. Some considerations for the different compression ratios may be with a low compression ratio, it may over amplify softer, non-important environmental sounds being bothersome to the hearing aid user. With higher compression ratios, they may adversely affect the clarity in just the overall pleasantness of the quality of the amplified sound. The next component of compression that's important to understand is attack and release times.

These refer to the amount of time it takes for a compression circuit to respond to intensity changes of input or environmental sounds. In simpler terms, this refers to how quickly compression kicks in once a certain input level is detected, if we're talking about attack time. And in terms of release time, it's how quickly compression turns off once 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 compression will act on that louder signal. At the same time, you do not want it too fast or it can negatively, again, impact that sound quality.

Release times are typically a little longer to ensure variability and that input signal does not negatively impact the processed 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, a longer release time can sometimes interfere with that. A question that occasionally comes up for students, as well as new providers, is how the volume control interacts with compression characteristics. Compression characteristics may be independent or dependent on volume depending on your signal processing strategy. Let's quickly review just the basic components of a hearing aid. You have a microphone which picks up the input signal, which then goes through an amplifier to be sent to the receiver to deliver the output processed signal to the ear.

This system is powered by a battery, as we all know. Now, we have not quite discussed wide dynamic range compression just yet. We'll get there in just a second. But white dynamic range compression is the most widely used compression system today, and it usually utilizes an input compression circuit. In an input compression circuit, compression acts before the volume control, meaning the TK cannot be affected by volume control changes, which is good, because if the TK were to change that significantly, it changes how the hearing aids sounds. And if a patient is complaining about sound quality, but you know they manipulate their volume control quite frequently, that can make it very challenging to figure out what's wrong and why they're not satisfied.

In output compression, the compression acts after the volume control. So volume control changes can affect the TK. Now, the negative in this is that the patient can make changes to compression that negatively affect 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 sudden sound suddenly occurs like a pan dropping or something, in that case, because the compression acts after the volume control, it will not be as unpleasant for the listener. If it didn't, then that sound may be too loud. So compression limiting in a device uses compression to control or limit loud sounds from being too loud without causing distortion. Unlike the case with peak clipping, it's characterized by high TKs and high compression ratios to provide as little amplification as possible while still keeping sound natural.

That's the goal. So how would this be employed in a hearing aid? This may very well be the compression characteristic of the MPO, the maximum pressure output or the max output of the hearing aid to control those louder sounds without causing distortion. So you can see that there in that graph. So this brings our conversation into the wide dynamic range compression and expansion area. When we think of dynamic compression, we need to think in terms of time constants. Time constants relate 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 some who experience a smaller residual dynamic range, but conversely, may contribute to distortion of speech sounds and lead to reduced speech intelligibility.

Slower time constants have the ability to be less compressive and may be advantageous for those who experience some cognitive. In WDRC, the hearing aid provides different degrees of amplification depending on the input level. Softer sounds are given more amplification than louder sounds. Again, to account for that reduced dynamic range in which usually, although soft sounds are no longer heard, loud sounds still remain loud. So by using this, we are doing our best to reproduce a full dynamic range into the world of the reduced dynamic range listener. So what sets WDRC apart from what we have discussed regarding compressions so far? As I mentioned earlier, that the limitations of low TKs and low compression ratios is that sometimes, those unwanted environmental sounds can be over amplified and unpleasant for the listener.

Also, in the case of someone with normal low frequency hearing, they may be able to hear low frequency circuit noise coming from their hearing aid. So in short, a limitation is because of what its benefit really is. 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, other weak, unimportant sounds can be made too loud. When describing how expansion works, it will sound like the sound opposite of compression. So keep in mind they work together. Compression and expansion 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 only applies to the weakest sounds in the environment and ideally not speech. So it's applying these different amounts of gain depending on the input level. This graph represents an input gain function, so different from the input/output function we were looking at earlier. The blue line represents the gain in a linear hearing aid. So that's why you see it's a flat line at 30 dB gain because that is being applied regardless of the input, which varies across the horizontal axis until it reaches its maximum output as you see at the AD dB.

The orange line represents how expansion works, and where it meets the blue line is the expansion TK, which I will be defining further in just a second. What is important to note here though is that below that point labeled TK, as your input level is reducing, so is the gain that's being applied. The expansion threshold is the point that below it, expansion operates. Unlike with a compression ratio in which compression operates above the TK, with expansion, it operates below the threshold. Sometimes, these two thresholds are one and the same. So below this specific threshold is expansion and above it is compression. In some systems though, there are two separate thresholds. So some things to consider when it comes to expansion thresholds.

A low expansion threshold or or kneepoint is considered to be 40 dB SPL or lower. And an advantage of that is that it ensures speech audibility at lower levels. A disadvantage is that it may still be kind of a noisy sounding hearing device, a noisy circuit to the listener. A higher expansion threshold or kneepoint would be considered to be 50 dB SPL or greater. And an advantage to that higher setting is that the device is quieter, and then a disadvantage of course would be that it can have perhaps a negative impact on speech audibility. So considering this, depending on the listening needs and preferences of the person wearing the hearing aids, they might prefer a higher or a lower expansion threshold.

So it's always good to know when fitting a particular device, how expansion is set and what adjustability you have as the provider to tailor it to your patient's needs or desires. So expansion ratio is the extent to which an input signal is expanded. So similar to a compression ratio, it is the change in input over the change in output. So just to review on some specific numbers when discussing ratios, a 1:1 or a one ratio is considered to be linear. A 1.1:1 or greater than one ratio is compression. So expansion would be a less than one ratio. So between zero and one, for example, like a 0.8:1 would be a good example of an expansion ratio. To think about the expansion ratio in terms of, is it more or less expansion?

The closer the ratio is to zero, the greater the expansion or the more aggressive the expansion is. Greater expansion means a quieter hearing aid, good for sound quality and quiet environments. But if too low, can negatively affect soft speech audibility. The closer the ratio is to one, on the other hand, the less expansion or less aggressive expansion is, which certainly protects soft speech audibility again, but the noisier the hearing aid and the more likely circuit noise might be bothersome to the user. 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 their listening preferences is whether a high or low expansion ratio is beneficial.

It really can be very individualized. 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. So less steep of a slope, the closer the ratio is to one or less expansion. So I think that's a nice visual. That really helped me I know understand it better when I was trying to learn about the differences between expansion and compression and when they applied and what the differences were. It can be a tricky, tricky thing to figure out. So 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, potentially distortion occurring. So if they share the same threshold, although distortion occurring, if they share the same threshold, So although in some cases, some systems, that is the case. They will sometimes share a threshold, as we were talking about before. So let's turn our attention now to applications of compression as it applies to channels and bands and the differences in definitions there, as well as looking at some of the prescriptive formulas that we're all familiar with. The terminology for channels versus bands dates back to analog hearing aids when a channel was referring to a frequency band, meaning these 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 today, more than gain can be adjusted. For example, we can adjust compression characteristics as well as numerous other advanced noise management features. In general, bands are going to shape the frequency response of the hearing aid. And bands are also going to accommodate loudness tolerances. A band is where specific, or a band is where frequency-specific gain adjustments can be made and how the software better tunes the frequency response of the hearing aids to a patient's hearing loss. And depending on how many bands you have, for example, 16 versus 20 versus 24, the more tailored or personalized this fitting 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 toward the collective goal of providing precise noise control for any environment. So generally speaking, you can think bands for gain and channels for your more advanced features. Now, I know practically speaking, we all use these terms interchangeably. I just wanted to have an idea of the history of the definition and where that came from. But certainly, we are pretty prone to using channels and bands interchangeably in the industry today. Noise reduction algorithms are criterion based systems. In other words, incoming signals must meet certain 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.

All the parameters you see here are being reevaluated every six milliseconds and updates made every 20 milliseconds. That means each of these parameters is potentially resetting 50 times a second in each channel for the most accurate response possible. And the more channels you have, whether it be 8, which is pretty minimal these days, 12, you can kind of see how that's moving along and how that changes a little bit, or 16, the more accurate. So the more channels you have, the more accurate the noise spectrum in an environment can be captured for the hearing aids to make those subsequent decisions on which advanced features to implement at any given moment based on the user's acoustic environment. Now, we'll get into the prescriptive fitting formulas a little bit, and they really dictate how hearing technology sounds, right?

And how compression and expansion thresholds and kneepoints are set and is really the basis to what all other advanced processing or noise control features operate on are these prescriptive formulas. 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 manufacturers, such as Starkey's proprietary fitting formula, e-STAT. And then there are your more generic or commercially available standard fitting formulas such as NAL-1, NAL-2, as well as DSL. Let's start with NAL-R. NAL-R was one of the first standardized evidence-based fitting formulas. It is a linear fitting formula of which many future non-linear compression formulas are based upon and utilizes a loudness equalization rationale.

That is all frequency bands of speech will be amplified equally and equally contribute to the overall loudness of speech. Think back to that fixed amount of gain despite frequency and input level. Everything here is treated equally. NAL-NL1 is the compression formula that was developed based off of NAL-R. Now, prior to NAL-1, there were the development of some non-linear formulas once the limitations of NAL-R and how to remediate them were being explored. But there was a lack of regulation over these earlier non-linear formulas, and they were rarely evidence-based. NAL-R, being an already validated fitting formula, was a great place to start. So NAL-1 became the first of the validated non-linear formulas based on NAL-R.

The fitting rationale of NAL-1 was different, since it is a completely different strategy. It utilizes a loudness normalization rationale. Overall sound is normalized so that loud sound is loud, soft sound is soft. The relative loudness of different frequency components are preserved rather than just emphasized equally. Theoretically, it maximizes speech understanding by applying this type of gain frequency response. How gain is applied in this formula is based on the speech intelligibility index. So NAL-2 was created based on the feedback and empirical data collected over a decade of using NAL-1. It is also a WDRC formula, and the rationale was the same. As you just saw in NAL-1, it utilized very low compression ratios with the goal of restoring low level speech audibility, but also caused its main complaint that NAL-1 was too loud, especially for those softer inputs.

Professionals found that they were constantly reducing gain. So when producing NAL-2, conclusions drawn from an article in the hearing review in the Journal of Audiology, less gain and higher compression ratios were preferred, so that is what was implemented in NAL-2. So let's talk about DSL now. DSL was developed in the 1980s for pediatric fittings. It prescribes targets considering the variables of pediatric assessments and the fitting itself, such as differences in ear canal characteristics between children. Essentially, the latest version of DSL employs extended protocols and more flexible targets based on the information provided about that child such as age. Are they an infant? Are they 5? Are they 10? The etiology of the hearing loss and utilizing the data available since oftentimes, we don't have the benefit of having everything in terms of information when it comes to fitting a young child.

This rationale more closely resembles a loudness equalization rationale like NAL-R, because with children, depending on the age, they are still developing their speech and language and need to hear as much as possible. So what is so different about a pediatric hearing aid fitting? Incidental learning. Maybe you've heard that term before, right? Children need to be able to hear all of the sounds in the environment. It's not important to prioritize some sounds over others for comfort like we do with adults. Children need access to all the sounds in their environment, because they're still learning so much about sound. Also, sophisticated processing features are not employed in nearly the same quantity as they are with adult hearing aid fittings, or adult hearing prescriptive formulas and rationales.

So let's take a look at some specifics about Starkey's proprietary fitting formula. The value of a proprietary formula is their ability to optimize fittings and deliver sound quality with consideration for the manufacturer's approach to signal processing. So in this particular case, what's so special about e-STAT? e-STAT provides patients with a greater initial acceptance of a Starkey hearing aid by optimizing the signal based on Starkey's approach to signal processing. Utilizing e-STAT will allow providers to engage fast-acting noise management without any audible effects to the patients so that expansion kicking in, ensuring a patient's ability to hear and understand in busier environments. e-STAT is also designed to minimize vent hearing aid interaction to ensure the best sound processing for the patient regardless of the style of the device or the sides of the vent onboard the hearing aid.

When you enter into the fitting software, you can see here that the fitting algorithm will typically default to e-STAT as highlighted with that little yellow box. And that makes sense as that is Starkey's proprietary fitting formula, which is designed again to maximize acceptance of a Starkey hearing aid. But we do offer a little dropdown carrot there, and that is because we know that every patient is individual and we want you as the provider to be able to select from other fitting protocols or other fitting rationales as you see fit. So although we recommend e-STAT for our Starkey hearing aids, you always have the option to choose one of the more standard fitting formulas. e-STAT stands for evidence-based statistic analysis and is proprietary to Starkey.

Starkey recommends the use of e-STAT with our products because it really takes advantage of Starkey product expertise and engineering and how its compression architecture and vent modeling is integrated into our products. e-STAT was originally developed based on NAL-1, and like NAL-2 was in response to feedback regarding NAL-1. Now, Starkey being a privately-owned company, could develop and produce a fitting formula that addressed the issue of NAL-1 much faster than the National Acoustic Laboratory, which is a government-funded agency. So Starkey also can consistently empirically optimize e-STAT based on provider feedback and clinical product research. So it's an ever-evolving formula that's being tweaked for the benefit of the patient. Sound quality is always job number one here at Starkey, and our e-STAT is ever-evolving.

It is a wide dynamic range compression-based formula that is fast acting, so it utilizes low compression ratios and low compression thresholds that work best with fast acting formulas. By setting it up this way, it provides the best sound quality, speech clarity, and greater audibility for more sounds. It can be difficult or not as intuitive to think of patient complaints in terms of compression needs. Oftentimes, they report things in terms of loudness. Therefore, it's easier to make adjustments based on gain needs rather than compression. ISO allows the professional to focus on the gain needs of the patient and make adjustments accordingly while allowing them to view the compression ratios if they like. Gain adjustments do indirectly affect compression.

So we give that ability to view the compression behavior within the Inspire software. So if we take a look here at fitting formulas, do they perform the same across the board? If you are utilizing the same fitting prescription, do standardized fitting formulas perform the same across all manufacturers? Let's take a look. Fitting algorithms are used in programming, all hearing aids essentially, while universal fitting formulas like NAL-2 and DSL, for example, were developed based on trends and amplification to be used in the fitting of technology by multiple manufacturers. They do not account from manufacturer-specific approaches to signal processing. Here you see real ear measures, or I'm sorry, real ear responses using NAL-NL2 with hearing aids from four manufacturers using the same hearing loss, the same receiver matrix, and same acoustic options.

So you can see differences in the responses as a result of each manufacturer's unique approach to signal processing. Equivalent hearing aid responses are not possible. They're always going to vary a little bit. Let's consider Starkey's compression architecture as it pertains to music. So compression and music. We know that historically, fitting formulas have really been focused on speech, and because of that, there have been a lot of hearing aid users over the years who were not able to enjoy music as much as they would want to. And this can be a hard pill to swallow if you are a musician or if you've listened to music or played music or sang your whole life. So we wanted to take a look at what we could do about that.

Starkey was the first in the industry to take a unique approach to music and hearing technology when we introduced Twin Compressor Technology. We've been talking about how fitting strategies throughout the history of amplification have been specifically geared towards speech optimization. And rightfully so. It is the most important signal that people with hearing loss wanna hear. But part of why music historically has a poor sound quality with amplification is that it is being processed with a fitting formula geared towards speech. And we know that music is a completely different signal than speech. Starkey has developed a proprietary fitting formula specifically developed to optimize music listening separate from our proprietary formula created for speech optimization. We have two independent compression architectures.

One for optimal speech processing, and one specifically from music processing and enjoyment. When we talk about dual compression, we are truly looking at what is the input type coming in. Music is much more dynamic in frequency and intensity than speeches. So we're talking about truly using one compression architecture for speech and a completely different one for the musical range, because the two inputs are so very different and really should be treated as such. Music is so important to so many people, it contributes to relaxation, joy, it soothes a lot of people. We listen to it in celebrations. So many different uses. They do say that music is the universal language of all mankind, and we here at Starkey couldn't agree more with that.

Hearing aid wearers weren't satisfied with how they were hearing music, and so we wanted to really treat those music and speech signals differently. With our music prescription, we have a 10,000 hertz bandwidth. It's a 24-channel system, and compression variables are optimized for music. Loud music is processed without distortion, and there's linear gain at high inputs to restore that loudness perception. As I mentioned earlier, the Inspire software allows you to adjust soft versus moderate versus loud gain. And this formula restores audibility for both soft and loud music as well as the software allowing you to adjust both soft and loud music for your patient. So it restores audibility for soft music, restores that desired loudness for loud music.

And then keeping up with the goal of this course is discussing some new uses of compression and signal processing currently in hearing technology. So again, music is another area where utilizing a combination of both linear and compression approaches applies. Research has shown that these combined approaches, as well as a flat frequency response, really preserves the fidelity of music. In particular, loud music. So our dedicated music memory, again, uses that unique music centric fitting prescription for the enjoyment of music for all hearing aid users. There's a shaped gain according to hearing loss for quiet music and more of a flat linear insertion gain for loud music to restore that rich sound quality that so many musicians love.

That brings us towards the end of our content today. But I do always like to leave you with a couple of resources when it comes to learning more or learning whether that's a little bit more about the industry in general or any particular topics you might have an interest in, or if you're looking to get some resources in terms of Starkey specifically. We do always want to be here to support you, our customers. And so if there's ever a time where you need additional support during an Inspire session, you can simply click on the computer icon at the top right of Inspire and this will open up Audiology On Demand. And this allows you to connect to a live representative who can view your screen remote in and take control of your mouse to help you out if you're in a a difficult situation or you really need some technical help, something you haven't maybe run into a long time.

You always remain in control of the session and you can end it, of course, whenever you'd like to. But Audiology on Demand is a great resource to help with those challenging fittings or challenging concerns that might come up with your patients. Also, don't forget about Starkey Pro, where you can find brochures, a number of white papers, including those on compression that we talked about today. Numerous quick tips to really help you target in on some specific information that you might be looking for when it comes to Starkey hearing aids, as well as the ability to sign up for more training classes. So visit starkeypro.com for more information on any of those things. And then of course, we always invite you to learn more.

So feel free to visit AudiologyOnline.com, starkeypro.com to register for more training classes or AAA or ASHA as well to look for any topics that might be of interest to you. So this does bring us to the end of today's presentation. In conclusion, I really, hopefully, you feel like you now have a better understanding of compression and some compression characteristics and have obtained kind of a good idea of how and when to modify your fitting algorithms based on the needs of your individual patient. Thank you for your time and attention today, and I hope you have a wonderful rest of the day.