Technical Details of the IX Platform: Signia's Approach to Signal Processing in Demanding Listening Environments

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- Hello and welcome to the Signia podcast. I'm Brian Taylor, and today our topic is for anyone who wants to learn more or take a deeper, more technical dive into the IX Platform. We'll be discussing the technical details of IX and Signia approach to signal processing in demanding listening situations. And here today to help us better understand these technical details of the IX platform is Eric Branda. Eric is one of those dual AUD PhD audiologists with more than 20 years of industry experience, most of it with Signia. Eric, it's great to have you with us. Welcome.

- Hey, thanks, I'm happy to be here.

- Yeah, this is okay, I think a lot of folks out there are gonna be interested in this. And before we get into the questions, I just wanna remind our listeners that we have created a PDF document that has four figures on it that you're probably gonna wanna refer to during the podcast. Because this is a technical topic. It's going to be, I think, pretty helpful to have that visual to follow along with. So that'll be in the notes, you can download it. And we'll just go ahead and get started. And Eric, the first question that I have for you is around the evolution of the IX platform. So, since you've been with Signia for such a long time, I think you're the perfect person to kind of talk about some of the milestones in the evolution of the IX platform.

- I always find our technological evolution's fascinating because I think about each generation being a combination of its own unique development and attributes, but it's also building upon each generation before it. And if we go back 10 years ago, we were just introducing Binax. And with that we had narrow directionality, which provided better speech understanding and noise for the hearing aid where as compared to normal hearing listeners. Right after that, Bimax just followed right on and led us into the Nx technology and Nx is where we introduced own voice processing, which helps provide, as Nx stands for, a more natural experience when talking. And right after that,

we brought out experience and experience that platform enhanced how we can interpret the dynamic listening environment. So really transforming traditional classification methods into that dynamic experience. Not long after that, the AX platform came out and AX platform jumped right into the innovative split processing, which was just a completely unheard of approach through processing in the field. And this approach, it gives us a separate processing of the frontal or focused target sound and then separate processing for the surrounding sounds. And this is, as opposed to the traditional way of kind of applying, say, directionality to a, you know, single processing scheme where everything has the same processing. Then we just tweak that a little bit. Now we're giving separate processing for the different inputs. And that's really what just brought us to where we are today with IX.

- Well, that's a really helpful overview and for those that are following along with the notes, what Eric just described is outlined in figure one. So thank you for that, Eric. The next thing I wanted to ask you about is motion sensors and hearing aids. I think that Signia is a real pioneer in the use of most motion sensors. So can you tell us a little bit more about how they work and how they might benefit the hearing aid wearer?

- I've really enjoyed watching the change in what we've done with motion sensors. I mean, it started out using a motion sensor in the phone as a motion input signal. The phone would just wisely transmit to the hearing aids to indicate that the wearer was in motion, and then the hearing aid could process that. But we quickly evolved into the motion sensors being integrated into the hearing aids. So specifically, the accelerometer, based on size and accuracy and low power consumption, is really well-suited to provide motion information for processing. So the accelerometer can detect motion at different speeds, different directions, and it becomes very clear if someone is walking or running, riding a bike, or even just sitting down and remaining stationary and can even look at what position they may be in. Lying, stationary. And all this information becomes a valuable input to the classification system. And it helps to

anticipate the hearing aid wearer's listening needs. So the classification system is gonna drive what features are in use. So things like directional microphones, noise reduction, and having movement information can help in determining what degree some of these features should be implemented. So if I just jump into an example, if we consider someone in a noisy restaurant, they're having dinner with a friend. The classification system based on the acoustics would say, this is a speech and noise situation. We need to apply noise reduction, we need to apply directional microphones so we can better focus on that conversation partner in front of us. However, when the two companions stand up to leave, start walking out of the restaurant, the listening needs change drastically. There's more need to be aware of environmental sounds, people moving, and the awareness of the conversation partner, they're probably to the side or maybe even behind. They're certainly not basing the hearing aid wearer right in front of them. So, the acoustic situation is not changed, but it's still the same noisy restaurant. But because the wearer is now in motion, the hearing aid can detect this and add it into the decision process of the classification system. Therefore the directionality can be relaxed and it can help extend to that environmental awareness. Same situation, just now, the wearer is in motion.

- Yeah, that's a really helpful summary and again, I wanna refer our listeners to a figure. Figure two shows polar patterns in the hearing aids that and how they change based on the information from the accelerometer inside the hearing aids. So it's a nice visual that I think can help kind of bring to life what Eric, what you just said. So, thank you for that explanation. Next thing I wanted to ask you about, another pioneering feature inside of a Signia hearing aid is our e2e wireless technology. So we kind of throw that term e2e around a little bit loosely. So could you tell us a little bit more about what we mean by that term, e2e wireless technology, and what advantages does it have with respect to bilateral beamforming?

- Yeah, so e2e wireless is something we introduced. I think that was our accurate product where we brought that in. And, I mean, it was even an award-winning technology because it was so innovative at that time. And what it is, it's NeuroField magnetic induction, which is ideal for hearing aid to hearing aid communication. So the signal can transmit between the devices really efficiently and share the information. And with that sharing of information, we can now look at common classification decisions, common noise reduction, common directional applications, so we can really make sure that, you know, for a binaural fitting that the hearing aids are, are much more aligned. Now, monaural beamformers could, of course, provide directionality for their respective sides. But by sharing the information between the devices and using the wireless communication of e2e wireless, we can now look at the bilateral beamformer. So each hearing aid can receive inputs from each microphone on each hearing aid, which allows for either a very broad directional response, or more importantly, a very narrow or tight directional beam. And, of course, you wouldn't want that in every noisy, every situation, but I think we've always been practical in using it for those very noisy situations. So when we really need to suppress the background noise, we can very aggressively cut it out and focus on the communication partner. And then if the noise gets less taxing, then we can reduce the directionality. And that's all part of the bilateral beamformer. And we actually do have research showing that with this technology, the hearing aid wearer can outperform normal hearing listeners in some of these really difficult speech and noise situations.

- That's good to know. And so far, I think we've covered sort of the legacy features inside of Signia. And with that, I want to kind of move now into more discussion around the IX platform and some of the newer innovations inside that. But before we do that, I think it's important to remind everyone that one of the motivations around what you see and hear in the IX platform is a real attention to detail when it comes to trying to improve challenging group conversation situations. And I think it's safe to say that IX was designed to optimize performance in those group conversations. And I think, of

course, a hallmark of demanding listening situations are multiple talkers. When the dialogue can quickly volley between people in a, you know, sort of in an improvisational way. You know, conversations are not scripted. They're usually free flowing and improvisational moving around in the room, generally, in the frontal plane of a person as they listen. And I think another key challenge associated with those demanding group conversations is the signal to noise ratio can oftentimes be quite unfavorable plus five, zero, even in the negative signal to noise ratios. Therefore, it's important to have a hearing aid that is geared towards optimizing performance in these challenging group conversation situations. And I think IX was designed with that in mind. So with that all said, Eric, can you tell us what are some things that are unique about IX that make it especially effective in those group situations, demanding situations, however you wanna phrase it.

- Yeah, I think IX is really the next logical step in directional processing. And I definitely don't wanna take for granted all the other features that it has in there. But to the point of those difficult multi talker conversations, that's where we really wanna look at some of the advances we see in IX 'cause even some of what I talked about before, the considerations typically been on one-to-one conversation and primarily capitalizing on the hearing aid where facing the conversation partner. In a multi talker situation, this has challenges. The hearing aid wearer will typically face the main conversation partner, but with a group, other people are going to talk, they're going to interject. They're not gonna raise a hand and wait for the hearing aid wearer to turn to them and say, "Hey, it's okay to talk." So, we need to be more dynamic in how we can address those other talkers who aren't directly in front of our nose. So, in IX, we have realtime conversation enhancer. The realtime conversation enhancer applies multiple directional beams that can highlight where other conversation partners are while still suppressing the other competing noises. So they may not be as narrow like the traditional bilateral beamformer, but we don't wanna make that unnatural. They're there to really help pick

up these other communication partners that are in front of us. So the hearing aid wearer doesn't miss parts of the conversation that might otherwise be lost.

- So let's dig a little deeper into some of the details of IX processing. And I think maybe one way to do that is to compare IX beamforming to what other manufacturers might implement in their bilateral beamforming system. Now I know not all manufacturers use bilateral beamforming. There's a couple that do. So can you tell us about what might be different in IX beamforming, bilateral beamforming compared to a couple other manufacturers that have other implementations of bilateral beamforming?

- Yeah, and I always like to take an opportunity like this to remind people that with most hearing aids now, when we say beamforming, this is today's approach to directionality. In the past, we depended on that physical separation of the microphone ports allowing the mechanical process to enhance signals from the front into that single microphone. Now with digital processing, we apply beamforming, which still requires that physical distance between the mic ports, but now each port has its own dedicated microphone and the algorithms can start to make calculations for beamforming our directional effect based on the inputs of each of those microphones. Now with these algorithms, we can do much more than we've done with traditional beamforming. And, you know, traditional beamforming is still what I think most other technologies in the marketplace are using. So one of the big enhancements with beamforming was using wireless communication between the devices for that binaural, bilateral beamforming. So this, as I said before, all four microphone inputs can be used and then we can look at these narrower directional beams and really help, you know, focus very tightly on what's in front of us. And different approaches will be used by different manufacturers to achieve this. It's the types of algorithms they approach with the technology they have. But IX is really gonna differentiate from that because now, instead of just looking at that narrow beam towards a single target, we can look at multiple beams. So we're still gonna calculate everything coming into every one of those microphones, but now we're going

to apply different beams at different widths in different directions to accommodate the multiple talkers.

- Yeah, and what you're saying, I think, can only be accomplished with a four microphone array and not all manufacturers have a four microphone array and, you know, that's a really unique aspect of our signal processing that others don't have. So, thank you for that. Now another feature that we talk about with IX is something called, real time conversational enhancement. So I'd like you to go into more detail on that feature. I guess my question, Eric, is how does IX know when speech is present and how do the regional beams inside of IX and RealTime Conversation Enhancement work?

- So, the first thing for acts to do is to identify that a signal is speech. Unfortunately, our detection algorithms have been doing this for years. When we look at the modulation of the signal, were very accurate identifying speech versus a non-speech signal. So, we can identify the speech and but with that split processing, we're very good at saying, "I have speech in the front or in my focus field where I can then suppress things that are non-speech in my surrounding field." So we're good at first identifying it and then looking at the direction that it's coming from, making sure it's the direction we wanted to start managing. And then because of the four microphones we're using in that wireless communication, we can start looking at the timing differences of those speech signals as they arrive at each microphone. Because if we think about it, you know, front to back we're gonna hit the front mic first, the back mic second, but then also with different angles, we're gonna hit the front right microphone first and the back left microphone, there will be some timing differences there. So the same types of, you know, traits we'd use for timing and differences for localization. We can apply for the hearing aid microphones to help identify where a signal is and then that RealTime Conversation Enhancement can start to see how many speakers do I have, where are they coming from, how far away are they? And it can really get that

relative distance direction and start to decide how to apply these beams very dynamically, trying to keep it as natural as possible but accepting that we have to augment it or change it so that we can optimize the group communication. So by putting all of that together, we can create a different output that accommodates multiple speakers and that's really what the RealTime Conversation Enhancement is doing is to boost all of those speech signals to improve the communication scenario.

- Now for the listeners following along with the schematic, what Eric has talked about, so far, I think is laid out really nicely in a schematic that's labeled figure three. So that kind of shows you how RealTime Conversation Enhancement fits in with some of the other features that are part of the platform. And more on, we'll get into more details on that in a moment. But my next question, Eric, is just like every other hearing aid manufacturer that I know. A platform is available at different tiers, so IX is available at different technology tiers. Tell us what does the Premium 7 IX have that the 5 and the 3 IX does not?

- Yeah, so the 7 IX really takes advantage of the capacity of that IX chip. So a key differentiator over the other tiers is the 7 IX is the only one that uses the full three region beams overlaying that front focus beam. And again, this is managed by the conversation source analyzer. So, this gives to kind of the full battery of how we can address the multi-speaker situation. So the conversation source analyzer is going to find ways to identify how many speech sources there are. Obviously, important for multi talker situations, but it's also gonna find out which directions they're coming from, where they're located, how loud they may be speaking, how far away they may be. And then inside of each of these beams, it's gonna have a conversational dynamics analyzer. And this is looking at some of the patterns of speech to help look at say things like turn taking as well as other environmental sounds to anticipate how to adapt the beam to optimize the speech in those focus beams. And again, the other technology levels, performance levels don't have this diversity in focus beams. So, it's

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a 7 IX and it has these algorithms that are there to really constantly analyze and update that output to maximize the dynamic listening situation. And then, of course, again, I don't want to take any of the other features for granted 'cause there's also things like spatial speech focus. So we know whether we want directionality forward or maybe we need to hear something behind the listener or to the sides of the listener. Auto echo shield, we need to look at hassle-free management of a reverberate space. Extended bandwidth. So getting out to 12K in the processing and then 48 channel processing. So there's so much that the 7 IX offers that puts it at the premium.

- That's very well stated. I like your emphasis on the three front facing regional beams and just for a little bit of depth on that term, some people refer to those regional beams as acoustic snapshots. I know if you go to the Ricketts, Mueller, and Bentler book in chapter 10, they talk about four mic arrays and the ability to create multiple acoustic snapshots, but that term is synonymous with regional beams, right, Eric?

- Yeah.

- Okay. Let's go ahead and now take an even deeper dive into the conversation source analyzer. As you mentioned, there's a tremendous amount of analysis happening of the wearer's listening environment. Could you walk us through some of the analysis that summarized in figure four of the handout? I think that, I want our listeners to get better appreciation for all of the stuff that's happening under the hood and how IX determines both where speech might be relative to the hearing aid wearer and how IX accounts for the movement of the talker. So, I'll let you go into some detail there. If you could, please.

- Yeah, because I think the chart you're showing there and the previous chart are really a good way to look at this because we really try to break down what might be happening step by step while also accepting there's a lot of things happening

simultaneously. There's a lot of parallel processing so that we can look at things like noise reduction, directionality, compression on, all on an unmodified signal and then make the best decision on how we put it together. And that's just part of all how all this processing is working. And one piece of this is this conversation source analyzer. And this is a kind of a really nice little brain piece of the processing because if you see the four main components there, the first is that the hemisphere detection and that's done right out with the split processing. So we know where the speech is coming from and, you know, that's very important because most hearing aids are going to look for what's the strongest signal coming in or what's the mix of signals coming in. So, it may know that there's speech coming in, it may know there's noise coming in, or there's a speech and noise situation and maybe the speech is the loudest signal. Here, we can really separate that what's coming from the front is speech and treat that one way and what's coming from the back is non-speech and treat that differently. That is very unique in hearing aid processing and that's something we capitalize on here because that's the first piece of the conversation source analyzer. So, what is coming from that front hemisphere and how do we process specifically for that with it being speech here. And then, as it identifies that, we wanna look at the further algorithms to be applied and that's where these next three really come in hand in hand. So the acoustic proximity detection, that's gonna help us determine how far away that listener may be. So as I mentioned earlier, it's gonna hit a front mic first and a rear mic second. We know that timing characteristics so we know the direction where they're coming from and we can figure out, you know, how far away they may be. 'Cause we can calculate the change in intensity as it hits each microphone. So, I don't wanna simplify it, but it is some simpler math for the hearing aid to calculate. And then we can look at the angular detection because again, wirelessly, we're sharing the information between devices, so timing and intensity cues that we would use for localization. Now we can use those between the four microphones to help pinpoint and identify the location of the target speaker or a target speaker out of multiple target speakers so we can figure out where they're all based. And then, of course, you have the speech intensity

estimation, to let us really determine the loudness of each speaker and have a better balance between the speakers so we can really see, yeah, what the dynamics of the conversation situation is. And that's all funneled into the IX processing so we can get a more complete auditory picture of the conversation. So that's how I kind of like to break that chart down.

- Yeah, thanks for that. Again, that's figure four. That's for somebody who really wants to get into the details. That's something that we don't talk about very much in seminars so thank you for sharing that. I think it's some important details that some of our listeners might want, might appreciate. I wanted to kind of circle back and see if there's anything you had to add around unique processing that's still available in 5 and 3 IX. I know we talked about this a few minutes ago, but based on what you just said, am I losing anything? If somebody moves down in technology from 7 to 5 or 5 to 3, tell us a little bit more about that if you could.

- Yeah, I always like to to consider those 'cause there is gonna be some changes in the technology as you change performance levels. Going from 7 down to the 5, down to the 3. But I also always like to start thinking about the advantages of different performance levels in the sense that, you know, we are so advanced even at a lower tier model over what used to be provided as a premium technology. I mean, in the 3 IX, we have a 32 channel device and it still provides split processing, it has some conversation source analyzation so it can really look at that front hemisphere and find ways to boost that speech signal in consideration of the surrounding background speech. So we still have a strong speech focus there and it's considerably a technical advantage over what's been done previously, especially at this performance level. And then we can jump right up into the 5 IX which is really a logical go between 3 and 7. It fits there as it should in a nice way because it gets us that step closer to the full advances we see with the 7 IX. And we've got things like 36 channels here. We have that split processing which is, of course, critical for that front hemisphere, but now we

can add a focus stream on top of that. So it's not the three focus streams we see in the 7 but now we are adding a focus stream that gives us some more discreet processing to prioritize speech and, you know, that lets it all within the variability of processing in that beam and gives us things like the proximity enhancer so we can look at how to better identify where that speech is. We can look at the distance of the conversation partner and then the conversation booster of the 5 IX can still maintain a stable balance of speech and noise to simply ease communication. And then we can't forget things like a reverberate room program, a directional hearing app to help control the directionality and speech focused to help guide directional microphones towards a primary talker in front or perhaps in the back of the wearer. Consider somebody that's driving and needs to hear who's behind them. All this is part of the 5 IX and, you know, it's all of these features that really help make it as beneficial listening situation as possible.

- Excellent. Let's shift gears a little bit and talk about research. I know that this is something that you've been involved with for a long time and it's important for many reasons. When you make product claims, you have to have data to support the claim. I think clinicians out there have added confidence when they know that there's research out there to back up different, the performance, different aspects of the device's performance. So, I first wanna ask you about a benchtop study that describes the four dB of relative improvement. Now before you get into the details of the study, I wanna remind everybody that in the notes, there will be a link to the paper or the study that you can access in the Signia library. So, Eric, what do you know about this benchtop study and why is it important?

- So, this is, I think, really kind of an interesting one and as you mentioned, you know, the study is available for people to read so I really wanna focus on the highlights of it. So forgive me if there's some things I don't talk about but I wanna kind of get the most interesting parts in here. Otherwise, this is.

- Good, good.

- Some good nighttime listening then. So, the idea behind this study was to measure signal to noise ratio improvement. So SNR improvements compared to four premium competitor products. And to do this, we implemented a technique called, the Hagerman method. And this method is something we do when we're trying to record in and look at SNRs. So with this, we're gonna make two separate recordings of the hearing aid processing and with that, it's processing a mixed speech and noise signal. So, we'll make two different recordings and the noise signal will be inverted in the second recording. So when the two output recordings are averaged together, the noise signals essentially cancel out. 'Cause one was recorded as original, the other one was recorded with the noise going in an inverted method. So when those are outputs are combined, that cancels out and that leaves us with an essentially clean speech signal. Then we can simply take that speech signal and compare it to a recording of the noise signals that were done in isolation and processed by the hearing aid to determine an actual signal to noise ratio. So, it's kind of like doing three recordings. I've got two that are done to cancel out the noise and leave me with clean speech and then I take a recording of the noise as the hearing aid would process it. So then I can now really see what type of SNR the hearing aid where is going to get. So, using that approach, then what we wanted to do is measure the Signia IX and the four competitor devices. So they were all programmed to flat 50 dB hearing loss and they were placed on KEMAR for the recording. And then as you can imagine, we want to keep this as stable as possible. So things like feedback cancellation, which has its own phase and versions there and frequency lowering were deactivated 'cause that would actually affect the quality of the recordings and the analysis. So those features were deactivated on everything so it was consistent there. And then, again, the hearing aids were placed on KEMAR for recording and in this condition, we had speech presented from 0 and 315 degrees azimuth and we alternated the speech between one and the other to simulate

a conversation. So 0 degree would go and then the 315 would go and we'd go back and forth with those. And noise was presented from the rear at 225 and 135 degree azimuth. So, a couple points behind the KEMAR. Recordings for each device were made after an initial 50 seconds, so 50 seconds, and this allowed all the devices to stabilize. So we looked at what's the longest that any one of 'em needed and kept them again all to the same protocol there. So, they all had 50 seconds to hit some stable processing, allow for any adjustments they might make. And then for each speech presentation direction, from the front end to the left was recorded twice for 10 seconds. So we had recordings each 10 seconds long and we combined all of that and the SNR was determined using both presentation directions and the background noise to figure out the overall conversation SNR. And in addition to the hearing aid recordings, we did an open ear KEMAR recording just to have what would be natural SNR be. And the results of that, we found the Signia IX provided 4.1 db SNR improvement over the nearest competitors. And three of those other ones were pretty similar. One was much poorer but over the top ones, Signia had the 4.1 dB improvement over those. And that result was actually an 8 dB signal to noise ratio improvement over the unaided condition. So, I look at this as some very positive results for a simulated multi conversational partner just using these standard measurements to look at SNR with the IX technology.

- Right, and they call it, a benchtop study 'cause it kind of shows proof of concept like in this very tightly controlled environment with KEMAR, using this objective approach. You can see that the regional beam concept really stands out compared to what other manufacturers might be doing in a similar simulated situation. So, it's a really good study, I think, to show that it works. I know another study that you were involved in, you know, looking at another layer of evidence, was a lab study and I think the... Finding was that 95% of the participants showed better performance with RealTime Conversation Enhancement with that feature activated. So if you can maybe go into a little bit of details of this study again, the link will be to the library so you can get the

details. But Eric, if you could describe the study, kind of at a high level like you did the previous one, that would be great.

- Yep, and I think this kind of study's important because I love having the objective measurements, but it is important to put the hearing aids on people and get those results.

- Exactly.

- So in this study, what we were doing, we were comparing the wearer performance with and without the RealTime Conversation Enhancement for a multi talker conversation scenario. And this was also done outside of a standard test booth, so more of a room that had more natural reverberation and acoustics than our nice dampen, you know, sound and test booth. So, we went to make this a little bit more like real world but still maintaining control over, you know, test quality. And with this, we had 20 participants, all with sensorineural hearing loss, all experienced hearing aid wearers. I think it was 10 men and 10 women. And then half of them were fitted with closed to sleeve fittings and the other half with open fittings. Although a little spoiler alert, we didn't find that that had the biggest impact on the results. So we could have just grouped them all together as well, I think. But for that, then the IX devices were programmed to the proprietary fitting targets with two programs. One program had all the features including the RealTime Conversation Enhancement activated and the other was identical except RTCE. the RealTime Conversation Enhancement was deactivated. The setup we used was a modified OLSA, the Oldenburg Sentence Test. Oldenberg being the university where we did it at, in is German for sentence. So a German sentence test. And this is an adaptive speech and noise test. So target speech was presented from two loud speakers, both were set 20 degrees to either side of the 0 degree azimuth and speech babel noise consisting of the same sentence materials used as target sentences. So the same type of sentences they would hear from the

front were also being treated as a babel from 90 degrees, 136 degrees, 180 degrees, 225, and 270 degree azimuth. So, you can kind of picture this whole array of speakers in the back and then these two speakers that had the target speech in the front. And then the background speech babel, babel was presented at 67.5 dBA and that level remains stable. So the idea as an adaptive test is we keep that background noise stable, we don't change that. And then we adjust the target sentences based on the response from the hearing aid wearer. So when the target sentence is repeated correctly, we turn the speech level down and when it's incorrect, we turn the speech level up. So this lets us start to find that signal to noise ratio that they would get, in this case, 80% correct. So we would call that, since that's like a speech reception threshold, that is our SRT 80. So we're looking at what's that SNR where they can get 80% correct, that's our SRT 80. And as you probably already guessed, the test was performed for each condition. So with the RealTime Conversation Enhancement on and then with the RealTime Conversation Enhancement off. So what we found with that is with the RealTime Conversation Enhancement on, we saw a mean improvement of 1.1 dB in SNR over the off condition. And that's significant at the P less than 0.01 level. So we have a significant result there. And when it comes to SNR measures, I think what becomes meaningful is that in the real world, this can be around a 10% improvement in speech understanding. And so we're always looking at, you know, what's that patient coming and asking for, how to do better in difficult situations, and we can start to help give them marked improvement in those situations. And then tying back to what, Brian, what you mentioned earlier is that consistent with this benefit, we saw that for 95% of the participants, they all did better with that feature on. So we saw a nice improvement with that feature activated.

- It's good to know. And again, the link to that study will be in the notes. I do wanna add, Eric, there's one other, you know, we talked about a benchtop study, you talked about a lab study. Recently, we just completed a real world, I'll call it a survey. We surveyed wearers in four or five different countries comparing IX to whatever they had

been wearing. All the hearing aids I think were, that they had been wearing were four years old or newer, and the results were pretty dramatically in favor of IX when it comes to how does the hearing aid perform in the real world using some type of a scale for the person to make judgements on. So, I'll put a link in the notes to that white paper that gives you some real world data. Again, which is reassurance that these hearing aids work in demanding listening situations the way they're supposed to. Okay, so, Eric, I can't thank you enough for the technical details that you shared with us. Before I let you go, I wanted to ask you, what I think is maybe a little bit more of a practical question. There are a lot of really good hearing aids on the market today and I want to kind of pick your brain about who you think the best candidate is for IX processing. In other words, why should a hearing care provider recommend IX over comparable products? And what would you recommend to providers out there to get the most from IX technology?

- So, to answer the question and when it, you know, first looking at candidacy and when it comes to candidacy for IX processing, I always view it from a couple different perspectives. So, if someone is regularly in highly demanding conversation situations, IX with RealTime Conversation Enhancement is designed to help with those situations. I mean, it fits right into it. And the research shows us that it can help improve the listening experience in those situations. And from a different perspective, even for someone who isn't regularly in those demanding communication situations, then it really comes down to wanting to optimize performance when they're in those situations. Because you may not be in it all the time, but when you're there, you're there, and you want the hearing aid to perform its best. And then I still like to remind people, you know, it isn't only multi talker conversations, but reverberate conditions, music enjoyment scenarios, you know, these are the types of areas where IX processing can excel. I mean, hearing speech and quiet is a basic starting point and the processing magic comes in for every additional listening challenge layered on top of that. And then that leads into why a hearing care provider should recommend IX over

comparable products. And there are a lot of good technology out there, but IX has been developed for such a broad range of listening situations and performs so well in those situations. I think the evidence is compelling and IX has just opened a new approach to dealing with classically difficult listening situations, that multi talker scenario. It's not like it's a new situation, but we've got a new innovation to approach it and patients don't generally come in looking for, you know, help for the easy situations. They want to hear best when the situation is worst. So my feeling is that with a product like IX when it comes to providing an optimal solution for the patient, the professional doesn't need to compromise. And then as a professional working with the patient, you know, when it comes to the fitting software, I think any company software is easy once you know how to use it. And what I like with the connect fitting software is that we're constantly looking at how to improve workflow, make the software, you know, as efficient as it can be while maintaining familiarity for someone who's experienced with it. So how do we keep it new and fresh and still familiar? And I think a lot of people like to come up with their little tips and tricks that they like to use in programming. And our software, I always feel is well researched that the workflow and flexibility can allow someone who's not familiar with it to still provide that expected level of professional care to the patient. Just using the default settings that are available and, you know, the access they have to programming parameters. But then also, I mean, the first fit and default settings are investigated and validated to provide an optimal start at the fitting and then someone who is experienced with the software can always work with the best fit, their fitting approach. So, I think as a culmination, there's a lot that IX and the flexibility, there's a lot it has to offer the professional who wants to provide the optimal solution for their patient.

- Yeah, that's good. I think that, you know, having fit a lot of hearing aids over the years, no one can beat the way that we implement the four mic array using the e2e processing, having the three front facing acoustic snapshots. When you're in an environment with multiple talkers, there's not a better device that's out there. And I

can't think of anybody, even if you're in quiet situations all the time, I can't think of anybody that's not in a situation where there's two or three people that you want to talk to at the same time. And there's not a better performing device on the market than 7 IX when it comes to that. So, anyway, Eric, I want to thank you for your expertise. No one knows more about the legacy of Signia products than you. So thank you for taking time to be with us today.

- Thanks again for having me.