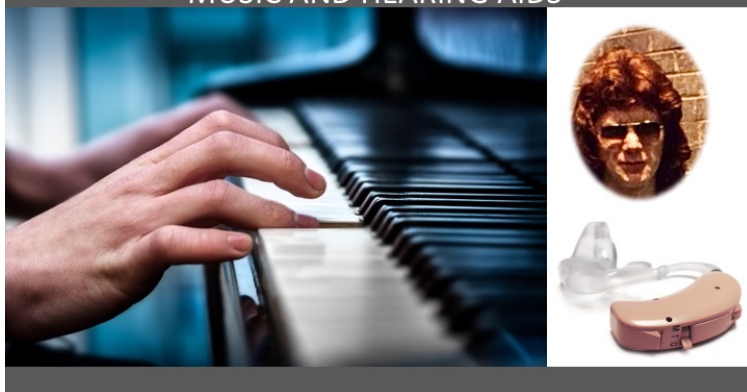


## MUSIC AND HEARING AIDS



I am an Audiologist and a Musician. I started learning piano when I was 3 years old and I started fitting hearing aids some 14 years later. Hearing aids were very basic and the majority of patients were older and were grateful for some help. As technology has moved on throughout the years I have had a particular interest in optimising hearing aids for music listening as well as for speech and I'm delighted to be passing on a few hints and tips today, learned over a number of decades, that may help professionals and technology users get the best from their devices. I work as a coach and mentor with independent hearing healthcare professionals in the UK to improve standards and quality of service and I also work with Danish Hearing Instrument Manufacturer, Widex as a Consultant.



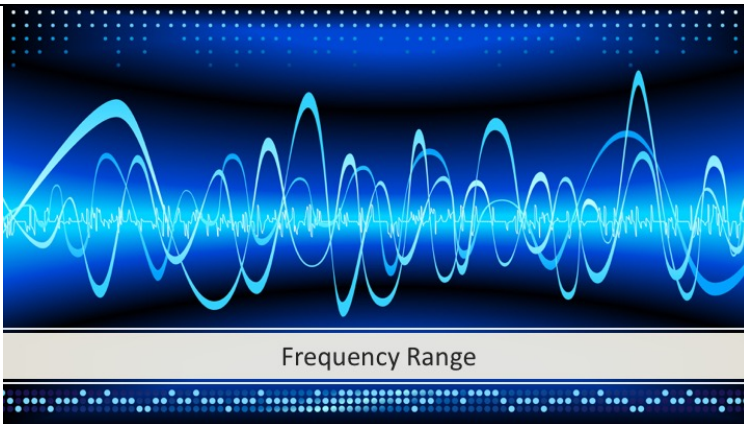
**Optimising Hearing Technology for Music**  
It's not all down to software

Like it or not, the hearing aid industry is marketing led. When digital hearing instruments were first introduced in 1996 they were marketed as the ultimate solution for hearing problems and, for many, of course they did provide a significant improvement over previous analogue aids with more control over frequency response and compression settings. However, many found that the early digital hearing instruments were not so successful in really loud environments and, in particular with live music, where they seemed to distort very easily. It is only in more recent years with work by the likes of Marshall Chasin who I am delighted to say is with us at this conference, that we have begun to learn much more about the limitations and possibilities of using digital technology.



## SPEECH vs MUSIC

As we have heard throughout the conference so far the main focus for hearing instrument manufacturers has always been on improving speech understanding, particularly in the presence of competing noise. This is of course, understandable as this is the most common complaint from those attending an Audiology or Hearing Care clinic for the first time. Anecdotally I know that a number of my own clients who were either keen musicians or music lovers did have some difficulties with technology and if speech and music were the same (or very close both in the frequency domain and the time domain) then the technology should work as well for both. As this is clearly not the case then there must be some clear differences between speech and music which create different challenges for Digital Signal Processing Hearing Instruments. So let us look at this in a bit more detail and see if we can highlight some of these differences...



The human voice has a fundamental frequency of between about 80Hz and 300 Hz. The system that we use to produce vocal sounds is composed of soft tissue and muscle and therefore acts as a kind of damper limiting high frequency components of speech. In fact there are few useful speech components above around 6kHz. Musical instruments on the other hand can cover a much wider frequency range. Take for instance a grand piano which has 88 notes...



88 NOTES

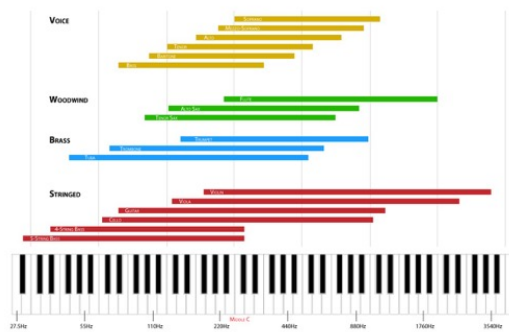
Bottom note A = 27.5Hz

A above middle C = 440Hz

Top Note C = 4186Hz

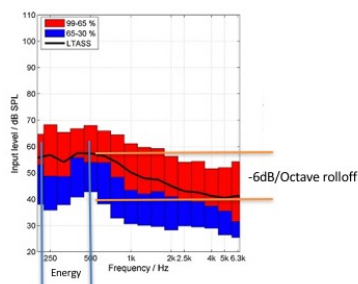
Middle C is 256 Hz, and the A above middle C on a piano is 440Hz, the lowest note on a piano is 27.5Hz whilst the top note is 4186Hz. It is important to bear in mind that these are the fundamental frequencies, and that the harmonic content of a some piano notes will extend well into the 10kHz+ domain. It is important to bear in mind that many musical instruments rely on harmonic structure to give them their unique timbre and just because a violin is playing an A at 440HZ there will be many more harmonics produced

## SPEECH vs MUSIC



This chart gives an overview of different frequency ranges of both voice and musical instruments. Stringed instruments cover the widest range (apart from the piano) whilst the majority of fundamental frequencies produced by brass instruments are in the same region as the human voice.

## Long Term Average Speech Spectrum

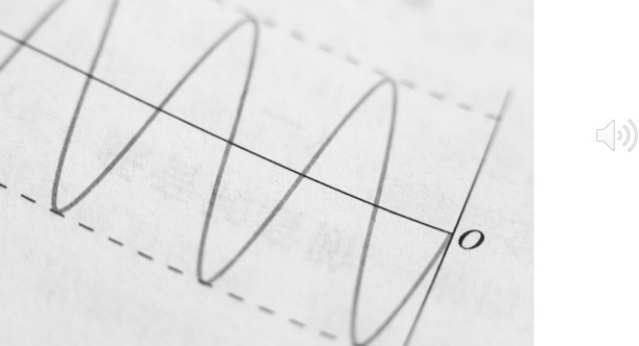

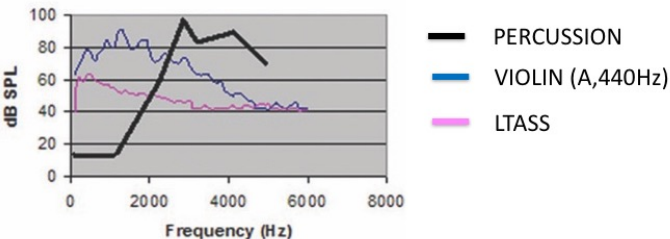


The Long Term average Speech Spectrum (LTASS) which is a universally accepted representation of the energy in speech across all languages clearly shows that most of the speech energy is at the lower frequencies and that there is a typical 6dB per octave roll off above about 800Hz-1kHz




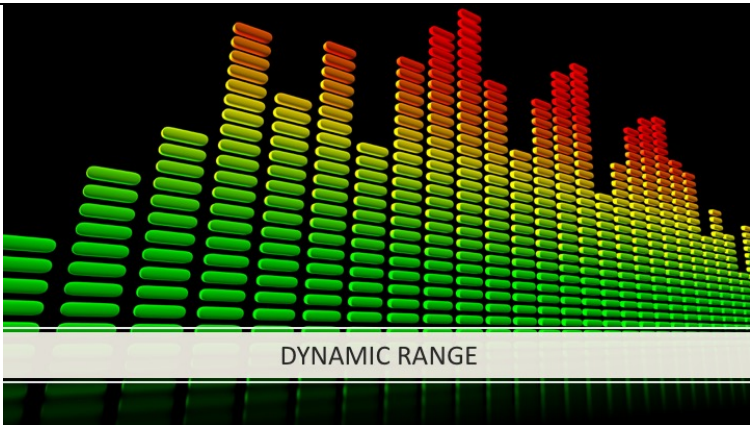
## TIMBRE



It is very important to understand why musical instruments sound different even when playing the same note. This is all down to harmonic structure. When a string is plucked or a reed is vibrated it vibrates at a fundamental frequency but also produces additional frequencies or 'overtones'. It is the level and number of these overtones that give a particular musical instrument its own unique character. So if a musical instrument is producing a sound at say 440Hz (A above middle C) then there

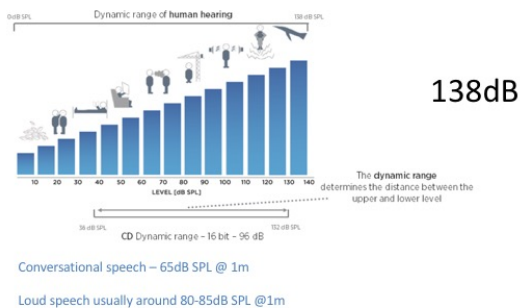
	will be additional overtones produced at much higher frequencies. If these are not reproduced accurately by amplification then the TIMBRE of the instrument will not sound right.
<div><p>Sine Wave or Pure Tone</p></div>	A pure tone is just that, a simple sine wave with no harmonics, just a fundamental frequency. It is not very interesting and does not naturally occur in music.
<div><p>TIMBRE</p></div>	Each of these instruments sounds completely different, even though they are playing the same musical note. The harmonic structure is particularly dense with the string instruments. You will also see that the amplitude envelope is very different for each instrument, This can be thought of as the time that it takes for the sound to die away after producing the note.
<div><p>LTASS vs MUSIC</p><p>M Chasin</p></div>	Here we see the Long Term Average Speech spectrum in pink/purple. If we compare the frequency range of the LTASS and a violin playing A (440Hz) in blue, They are surprisingly similar in pattern although the violin has much higher intensity. However if we superimpose the pattern of a percussion instrument then we see a completely different pattern. Because of this huge variation between individual musical instruments it is impossible to provide any prescription formula or target, however as long as the input into the AD convertor has been optimised broad band amplification and compression characteristics based on that used for speech should be effective. This is an important point as manufacturers of hearing technology often suggest that their own proprietary 'Music'



	programme offers advantages over others.
<div data-bbox="158 255 336 613"></div> <div data-bbox="368 271 812 309">Spectrum Analyzer app -iPhone</div> <div data-bbox="424 367 785 544"></div>	<p>For the professionals in the room who would like a really inexpensive way of visualising the make up of audio a fabulous app is available for the iPhone (Spectrum Analyzer by ONYX) that provides as much information in your pocket as a few tens of thousands of pounds worth of test equipment would have been needed for only a decade ago!! I also find this particularly useful when working with a musician who may bring their own instrument along to the clinic (or you go along to the rehearsal studio) I strongly believe that the more information we have to hand, the better we can work together to find the best solutions accepting that there will almost always be some compromises to be made</p>
<div data-bbox="89 999 839 1415"></div>	<p>This is a screen shot from the analyzer app showing the frequency content of a sound source with the dominant frequency highlighted in yellow. The Blue bars are in 1/3 octave bands and this can be adjusted and the yellow horizontal lines indicate the peak reading at the specific frequency – really quite amazing!</p>
<div data-bbox="89 1458 842 1881"></div>	<p>So we know that there are some significant differences between speech and music when it comes to frequency content, but what about the dynamic range (meaning the range from the quietest component to the loudest component of a signal) Well speech has a limited dynamic range conversational speech measured at around 1 meter from a talker is about 65dB SPL. If that talker now shouts really loudly the level may only increase to 80 or 85 dB at most. (If you get the analyzer app you can check this yourself) The peaks in a speech signal are at around 12dB higher than the average (RMS) value, so if the average is 85dB then the peaks will be</p>

around 97dB. This figure is an important one to remember as we will see shortly.

### DYNAMIC RANGE OF HUMAN HEARING






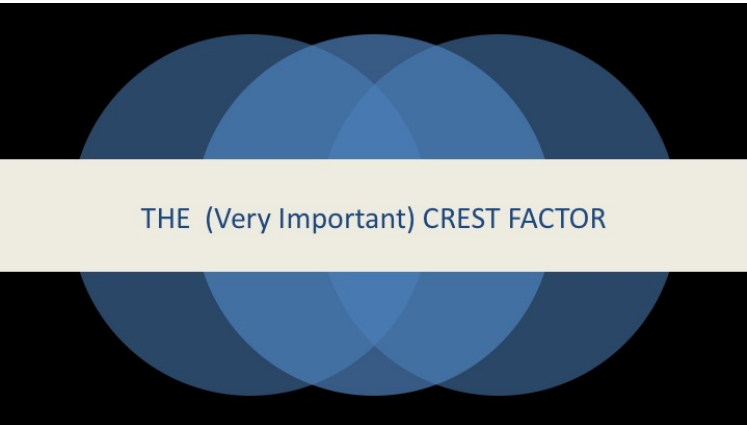
In a digital signal processing system the two main things that affect overall sound quality are:

A - The Sample Rate - this needs to be 2x higher than the highest frequency that we want to represent. For example if we want to process up to 16kHz then the sampling frequency needs to be at least 32kHz. For 20kHz it would be 40kHz etc.

B - The Bit Depth - this is the number of bits in each sample and affects the dynamic range of the circuitry. The dynamic range is described as the bit depth X 6 so, with a 16 bit system we have a dynamic range of 16 x 6 or 96dB (pretty close to the 97dB speech peaks that we had to remember from before). For this reason, most 16 bit dsp hearing instruments are able to work well with speech signals. The 96dB range does not have to go from zero to 96, in fact, in the case of a CD it goes from 36dB SPL to 132dB SPL.

NOISE SOURCE	dB	Peak
Single musicians		
Violin/viola (near left ear)	85 - 105	116
Violin/viola	80 - 90 *	104
Cello	80 - 104 *	112
Acoustic bass	70 - 94 *	99
Clarinet	69 - 92 *	112
Oboe	74 - 102 *	116
Saxophone	75 - 110 *	113
Flute	92 - 105 *	109
Flute (near right ear)	98 - 114	118
Piccolo	96 - 112 *	120
Piccolo (near right ear)	102 - 118 *	128
French horn	92 - 104 *	107
Trombone	90 - 108 *	109
Trumpet	88 - 108 *	113
Harp	90	111
Timpani and bass drum	74 - 94 *	106
Percussion (high-hat near left ear)	68 - 94	125
Percussion	90 - 105	123-134
Singer	70 - 85 *	94
Soprano	105 - 110	118
Choir	86	No data
Normal piano practice	60 - 90 *	105
Loud piano	70 - 105 *	110
Keyboards (electric)	60 - 110 *	118

Here we can see some typical levels measured at the musicians ear for various instruments. We can see that the highest peaks are from percussion instruments and this is really no surprise.

	<p>Microphones used in hearing instruments are tiny purpose made components usually with a maximum capacity of 115dB SPL. Signals arriving at the microphone diaphragm above this level will create distortion. Studio microphones can operate at much higher levels as they usually have a large surface diaphragm for the sound to impinge on and are operated at fairly high voltage 48V DC. A hearing instrument battery produces less than 1.5 volts and this is all we have to work with at the moment, although Danish manufacturer Widex had introduced a switched mode transformer function in their instruments which provides more voltage.</p>
<p style="text-align: center;"><b>HEARING INSTRUMENT MICROPHONE</b></p> <div style="display: flex; justify-content: space-around; align-items: center;"> <div style="text-align: center;">  <p>Typical Surface mount HI microphone</p> <p>Broadband frequency response Linear input range up to 115dB SPL</p> </div> <div style="text-align: center;">  <p>AKG recording mic 1" diaphragm</p> </div> </div>	<p>Modern hearing instrument microphones are typically broadband and are able to transduce signals in the main speech range effectively. The physical size of the transducer limits the performance at very low frequencies and the sensitivity at high frequencies rolls off above 6kHz. But, of course it is important to remember that this range is fine for speech where the typical fundamental frequency is around 200Hz and then rolls off at about 6dB per octave with little useful speech information above 4-5kHz.</p>
	<p>I must admit that it is some years since I first became familiar with the Crest Factor but this was not really related to Audiology and Hearing Instruments but Industrial Noise Measurement where averages and peaks had to be measured over a given time period. It is only in the last few years that the term crest factor has appeared in Audiology, but it is one of the most important, if not the most important concepts to understand when optimising hearing aids for music (in particular live music)</p>





This is defined as the ratio between the peaks and the rms value of a given waveform.  
In audio this is measured in dB and for speech the accepted (theoretical) value is 12dB.



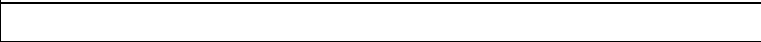
FOR SPEECH – CREST FACTOR is accepted as 12dB

For speech, the crest factor is accepted as around 12dB



THE CREST FACTOR IS SIGNIFICANTLY HIGHER WITH MUSIC – 20dB+

However, the crest factor for live music is significantly higher at 18-20dB



Speech average level @ 1 metre = 65dB

Loud Speech average level @ 1 metre 80-85dB

Crest factor for speech = 12dB

Peaks for speech = 96-97dB

Live music average level = 90– 95 dB SPL

Crest factor for music = 20+dB

Music peaks = 110+dB

So what does this mean?, Well if we have an average value for conversational speech of 65dB, adding the crest factor means that the peaks will be around 77dB. Even with our loud or shouted speech adding the 12dB to 80 gives us a peak level of around 92dB. Modern hearing aid microphones have a peak input level of around 115dB and most AD converters are happy coping with peaks of 92dB before distortion so this all looks lovely.... But..... we know that the crest factor for music is 20dB+ and we also know that the average levels of even ‘quiet’ music can exceed 85dB



	<p>This means that the peaks are in the order of 105 – 115dB or even higher causing distortion at the ‘front end’ of the hearing instruments</p>
<div data-bbox="309 611 622 647" data-label="Section-Header"> <h3>Hearing Aids Pre - DSP</h3> </div> <div data-bbox="89 703 531 996" data-label="Image"> </div> <div data-bbox="549 680 813 976" data-label="List-Group"> <ul style="list-style-type: none"> <li>• Analogue circuitry</li> <li>• Very little (if any) frequency shaping</li> <li>• Frequency shaping via earmould plumbing</li> <li>• Rear facing microphone on some!</li> <li>• Long battery life</li> <li>• Peak Clipping</li> </ul> </div>	<p>Before we had digital signal processing in hearing instruments the signal path was relatively simple. The sound was picked up by an analogue microphone and converted to an electrical signal, this electrical signal was amplified and then fed to the analogue loudspeaker which converted the electrical energy back into acoustic energy. There were no processing delays, no sampling errors, no unpredictable distortion products. However linear hearing instruments often relied on a very basic output limiting feature – Peak Clipping which sounded just like the name suggests. So they were certainly not free of distortion.</p>
<div data-bbox="169 1330 373 1563" data-label="Image"> </div> <div data-bbox="459 1330 469 1563" data-label="Image"> </div> <div data-bbox="584 1323 719 1563" data-label="Image"> </div> <div data-bbox="253 1608 679 1653" data-label="Section-Header"> <h3>Loudspeaker Technology</h3> </div>	<p>Of course in the 1970s and 80s the technology for reproducing music live was also quite basic with huge analogue power amplifiers (often valve driven) feeding even bigger loudspeakers which were often over driven so in a way, we were accepting of distortion to an extent. Modern amplifier and speaker technology means that we can produce the same power level with minimal distortion using much more compact products such as this line array system. So as music listeners we live in a much more distortion free world. Interestingly of course there has been a significant return to listening to music on vinyl where true enthusiasts appreciate the ‘warmness’ of the sound (which of course is a type of distortion)</p>

## Hearing Aids Pre - DSP

The K-AMP , 1990



**Mead Killion**  
Etymotic Research

- Analogue circuitry
- TILL (Treble Increase at Low Levels)
- Single channel compression
- Minimal distortion

This dapper chap is also a musician and founder of a company called Etymotic Research in the USA. They are famous for their custom made linear frequency response musicians earplugs (ER15, ER25 etc) But he also designed and manufactured a new hearing aid circuit – The K-Amp which was introduced in 1990.

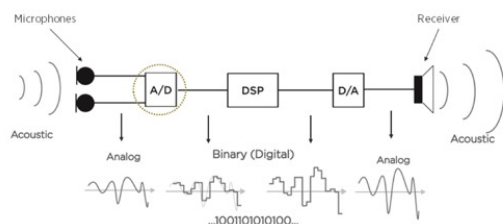
Personally, I fitted a lot of these products at this time and they were a revelation compared to linear amplifiers as they basically only amplified soft sounds. The down side is that they were a little noisy so if a person had normal low and mid frequency hearing then they could often hear the ‘hiss’ However, it worked really well, especially when listening to music. It is still available in the USA in a PSAP product – The Bean but has otherwise been superseded with DSP technology.

FOCUS  
FOCUS  
FOCUS  
FOCUS  
FOCUS



### DSP Hearing Aids Overview

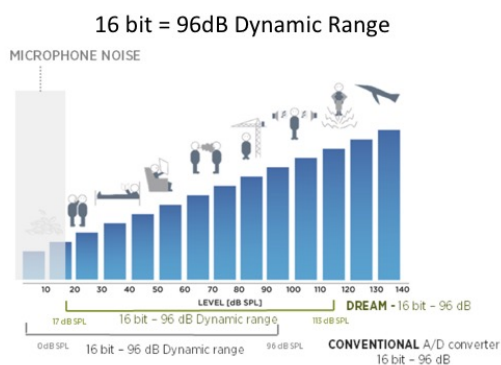
The main reason that people approach and audiologist for help is that they have noticed (or someone else has noticed) and increasing problem hearing speech, often when in a difficult listening situation. This explains why the main focus of Hearing Instrument manufacturers is on improving speech in noise. But, as we have seen there are some very large differences between speech and music (particularly Live music, so what are the difficulties that might present when a person uses DPS hearing aids to listen to music?



A typical DSP Hearing Instrument

Because of this focus on optimising speech, the technology used in modern digital hearing instruments has been optimised to provide maximum benefit for speech understanding with increasingly clever DSP algorithms and sophisticated directional microphone technology all helping significantly. We are all familiar with the basic operation of digital hearing instruments where the sound (an analogue waveform) is collected by the microphone and then passed converted by the AD converter to a digital signal before being modified by the DSP.

After processing the digital signal is then passed to the D-A converter which converts the digital signal back to analogue which the receiver then transduces into a waveform which is then passed to the ear. It is quite something to think that all of this happens in (hopefully less than) 10 Milliseconds





Until recently most DSP hearing instruments used a 16 bit A/D convertor which resulted in a dynamic range of 96dB. The 96dB range limitation with 16 bit processing means that the 'range' can be placed anywhere, so it could be from 0-96dB or 10-106dB or, in the case of the previous platform DREAM, 17 - 113dB. However, to achieve this very high maximum and maintain the response in a linear fashion requires additional power and this is where the advanced voltage doubling technique patented by Widex allows them to maximise the range whilst minimising the battery drain - its like magic!!

With the introduction of Unique the input stage is now 18 bit which gives a dynamic range of 108dB. The upper limit is now close to maximum possible with current microphone technology so the lower limit has been reduced from 17 to 5dB SPL

The figures referred to here are the

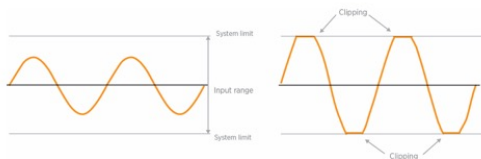


	<p>RMS levels (or averages) One very important factor that we need to consider is the peaks that might occur in an input signal.</p>								
<p><b>Higher Bit Depth Benefits</b></p> <p>16 Bit – Bit Depth is 16 x6 = 96dB  18 Bit – Bit Depth is 18 x6 = 108dB  24 Bit – Bit Depth is 24 x6 = 144dB</p> <p><b>BUT.....</b></p> <div>  <p>Microphone technology limited to around 115dB maximum and...</p> <p>Increasing Bit Depth requires additional power</p>  </div>	<p>Bit depth effects dynamic range. The dynamic range is the bit depth x6 so 16 bit = 96, 18 bit – 108 etc. The limitations for Hearing Aids is the microphone technology and power requirements</p>								
<p><b>The important bits for listening to music</b></p> <table> <tr> <td>Microphones</td><td>Broadband response and low noise</td></tr> <tr> <td>A/D Converter input range</td><td>Linear and peak as high as possible</td></tr> <tr> <td>Sampling Rate</td><td>Greater than 28kHz</td></tr> <tr> <td>Processing Speed</td><td>Quicker the better</td></tr> </table>	Microphones	Broadband response and low noise	A/D Converter input range	Linear and peak as high as possible	Sampling Rate	Greater than 28kHz	Processing Speed	Quicker the better	<p>So the important areas for music are: the microphones which should have a broadband and flat frequency response and as much linear dynamic range as possible, the AD converter which should also have the maximum linear range possible. A high sampling rate, bearing in mind that the bandwidth is ½ of the sampling rate – Nyquist theorem and the overall processing speed, which should also be as fast as possible – oh, and all of this with a 1.3V battery..... Long before the complex panorama of sound gets anywhere near the wonderful DSP section of modern hearing instruments it must first be collected by the microphones</p>
Microphones	Broadband response and low noise								
A/D Converter input range	Linear and peak as high as possible								
Sampling Rate	Greater than 28kHz								
Processing Speed	Quicker the better								

## Digital Distortion



Returning to the 1970's and every rock guitarist wanted to get more and more distortion from their instruments. This was usually achieved by 'overdriving' the input stage to their amplifiers but was also done electronically using foot pedals such as this....Digital distortion is not at all pleasant to listen too and needs to be avoided at all costs in hearing instrument fitting, however, with limited linear input range it is only too easy to overdrive the input to a DSP hearing instrument when listening to live music and the results can be very unpleasant



As soon as the input range is exceeded distortion in the form of clipping of the signal will occur

→ This all occurs before the DSP section of the instrument and cannot be removed in software ←

Remember any signal exceeding the maximum input range of our AD converter will produce distortion which cannot be later removed.

## SPOT THE DIFFERENCE



WITH LINEAR INPUT RANGE

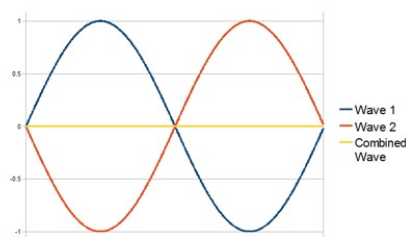


WITH A TYPICAL DSP HEARING INSTRUMENT

Listening through dsp aid with an extended linear input range or, in this case a more common 'limited' range This is what we as hearing care professionals have been providing our customers with for many years now - oh dear!

## FEEDBACK CANCELLATION SYSTEMS

### SIMPLIFIED!

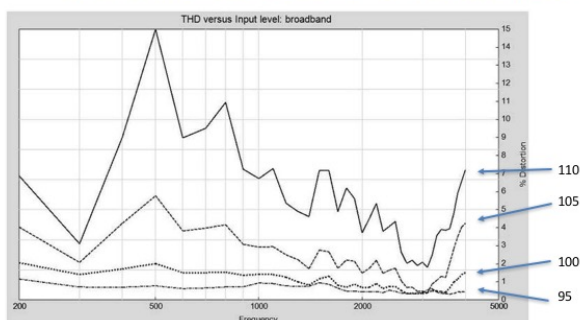


Most DSP hearing instruments include a feedback management system and this is often a 'phase'reversal' system. This means that when a feedback signal is detected an 180 degree opposite phase signal is produced to cancel it out. Of course this works quite well with real feedback. However some feedback management systems detect musical instruments as feedback and produce the same anti-phase signal which is obviously not what is required. In

	<p>general FB systems should be disabled when listening to music.</p>
<div data-bbox="108 255 828 645"> <h1>IMPORTANCE OF A LINEAR INPUT RANGE</h1> <p>The more linear the better...</p> </div>	<p>So why is a linear input range so important?</p>
<div data-bbox="153 707 780 779"> <h2>COMPARISONS OF MAX LINEAR INPUT LEVEL PERFORMANCE</h2> </div> <div data-bbox="217 813 729 1115"> </div>	<p>Widex have recently carried out their own measurements on a range of competing products with the focus being entirely on the linearity of the input range. The results make interesting reading and are seen here. If you are interested in the competing products that they measured I can reveal that the best instrument was (not surprisingly) the Unique 440 with a linear input range up to 113dB SPL. The worst performing aid was the Oticon Alto Pro which was only linear up to 92dB SPL. All of the aids on test were set to low gain, as linear a response as possible and no reduction in MPO to measure on the input limitations</p> <p>A recent conference at the Metropolitan University of Cardiff organised by Fei Zhao looked at some of the latest research into hearing instruments and music. Speakers included Brian Moore, Marshall Chasin, a development engineer from Phonak and a cochlear implant user. The presentations were thought provoking and stimulating and it a little surprising to see just a couple of hearing aid dispensers attend. However I would recommend it if it is repeated in the future.</p>

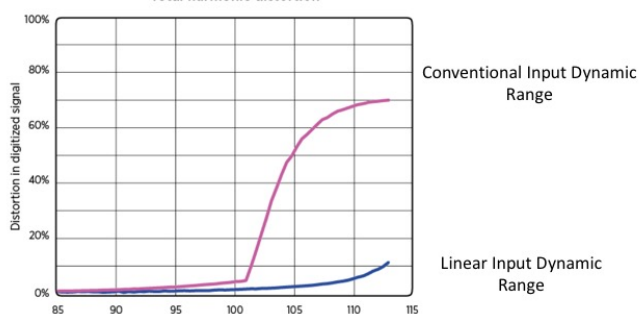


### Total Harmonic Distortion with various inputs



This diagram shows the level of Total Harmonic Distortion measured with different input levels and its clear that as soon as the levels get anywhere near 110dB the distortion becomes very unpleasant.

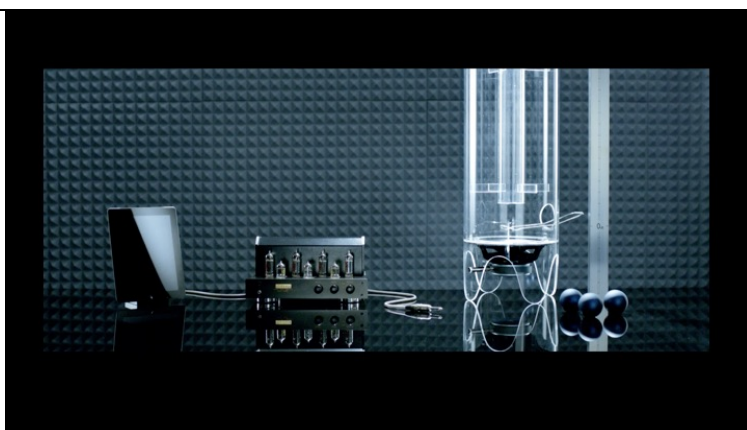
### Total harmonic distortion





This is a simplified diagram showing the same thing with conventional AD converters vs UNIQUE

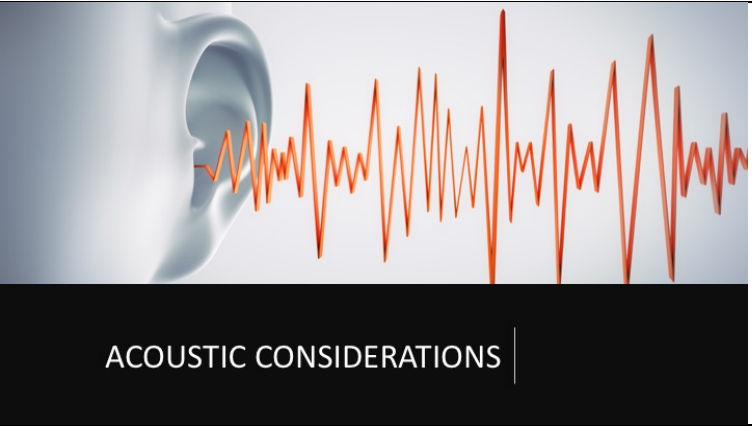
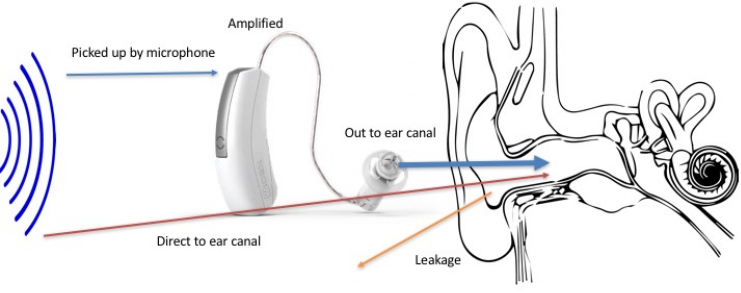


Now you may be thinking that this does not apply to your clients. However, you would be wrong. It is not just when listening to music that these artefacts appear, but anytime when the input levels exceed the maximum linear input range of the AD converter. For example, on the street with passing traffic, in a bar or at a social gathering and so on.

The following short video clip provides a very easy to understand graphical representation of what happens if the input levels exceed the capabilities of the hearing instrument front end.



I believe that this video could easily be used in your practice when you are explaining to your clients the need for as large an input range as possible.

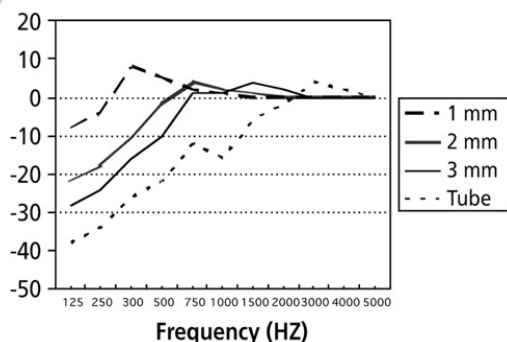
<div data-bbox="172 123 376 152" data-label="Section-Header"> <h2>THE ACHILLES HEEL</h2> </div> <div data-bbox="194 208 367 237" data-label="Section-Header"> <h3>A-D Converter</h3> </div> <div data-bbox="132 275 414 416" data-label="List-Group"> <ul style="list-style-type: none"> <li>• Pre DSP</li> <li>• Any distortion at this stage cannot be 'programmed out'</li> <li>• 16 bit dynamic range (96 dB) doesn't need to be 0-96 dB</li> </ul> </div> <div data-bbox="451 114 841 539" data-label="Image"> </div>	<p>To date the weak point in a DSP hearing instrument has been the A/D convertor and it is very important to remember that any distortion that occurs at this stage cannot be 'undone' in the fitting software</p>
<div data-bbox="143 611 786 647" data-label="Section-Header"> <h2>WIDEX CLEAR – UNIQUE INPUT COMPARISON</h2> </div> <div data-bbox="335 696 558 732" data-label="Text"> <p>WIDEX CLEAR 440 </p> </div> <div data-bbox="343 797 558 833" data-label="Text"> <p>WIDEX UNIQUE 440 </p> </div> <div data-bbox="207 934 778 965" data-label="Text"> <p>Very loud music and speech (speech level is 110dB SPL)</p> </div>	<p>Just one more sound sample before we move on. This is loud speech (around 110dB SPL) recorded firstly through a Widex CLEAR instrument and then through a Unique instrument. All programming etc was identical in each aid so we are only hearing the difference in the input range.</p>
<div data-bbox="106 1052 826 1442" data-label="Image"> </div>	<p>A more recent development has been the introduction of direct streaming of a signal from a smart phone to a hearing instrument. Products that use the 'made for iPhone' technology are of course limited technically by Apple as they do not release the information about the quality of the signal being streamed via bluetooth. However, many people really benefit from streaming technology, particularly when it comes to phone calls, skype etc. However, the sound quality when listening to music streaming can vary considerably and is also affected by basic acoustics as we will look at now.</p>
<div data-bbox="363 1691 564 1722" data-label="Section-Header"> <h3>Direct Streaming</h3> </div> <div data-bbox="146 1731 319 2083" data-label="Image"> </div> <div data-bbox="320 1856 400 1886" data-label="Text"> <p>2.4GHz</p> </div> <div data-bbox="414 1749 544 1984" data-label="Image"> </div> <div data-bbox="608 1794 745 1928" data-label="Image"> </div>	<p>Direct streaming offers real convenience for many users and considerable control over some hearing aid functions. However the CODEC used is somewhat limited for listening to music.</p>

	<p>A major consideration when thinking about fitting hearing instruments for music are the acoustic considerations</p>
<p><b>SOUND PATHS (non occluding)</b></p> 	<p>Unless a person had a completely occluding or sealed earmould there are a number of routes for sound to both enter and leave the ear canal.</p>
 <p><b>Open Fit Considerations</b></p> <ul style="list-style-type: none"> <li>• When streaming no direct sound</li> <li>• Vent effect = minimal Low Frequencies</li> <li>• May not be best solution for music lovers</li> </ul>	<p>When fitting open fit then careful thought needs to be given to these different sound paths</p>
	<p>However it is important to bear in mind that if client has an open fitting and music is being streamed you will likely have to make some adjustment to low frequency amplification to compensate for the vent effect and the lack of natural sound entering the ear. In some cases this may require a larger receiver or, in fact reducing the 'openness' of the fitting.</p>





### VENT COMPENSATION



As the fitting becomes more 'open' or less occluded the low frequency gain reduces significantly. With a truly open fitting there is no low frequency gain from the hearing instrument.



### ASSESSMENT ESSENTIALS

Is Pure Tone Audiometry Enough?

For many decades the Pure Tone Audiogram has been considered the 'gold standard' for assessing hearing – but is it enough?

NO



Unless your client only listens to sinewaves the PTA is not really enough. Include SIN tests and Loudness Scaling as a minimum

### TECHNOLOGY SELECTION ESSENTIALS



It is essential that professionals have a clear and concise understanding of the technologies provided by HI manufacturers and their limitations. Selecting the best technology for a client is a collaborative decision and a thorough understanding of the technology will negate disappointment if expectations are not met

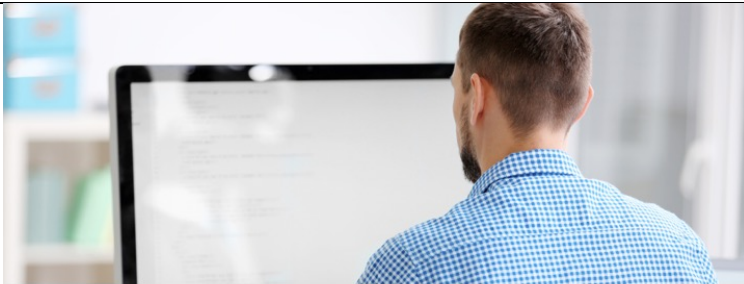
The largest possible linear input range is essential

Slow acting compression is preferable to fast acting

Minimal processing delay is essential

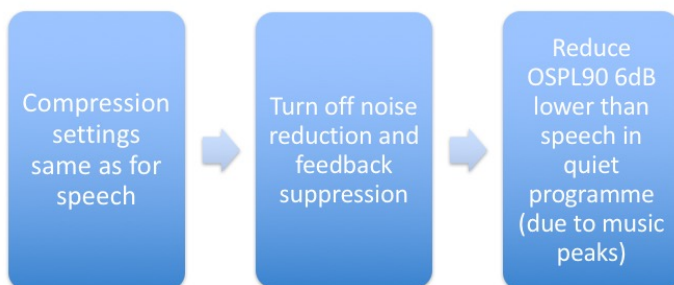
One Channel is best

Put very simply here are some of the minimum requirements when fitting hearing aids for music. NOTE: The final point 'single channel is best' is debatable. Prof Brian Moore argues that with a single channel a high level low frequency sound will trigger compression which will bring down the gain for HF also.



## Programming Tips

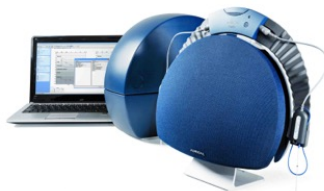
So if we have a thorough understanding of the client's listening needs, a comprehensive audiological assessment, and have selected the most appropriate hearing instruments we now need to think about programming.






The following recommendations as described by Marshall Chasin should be seen as an absolute minimum

## VERIFICATION

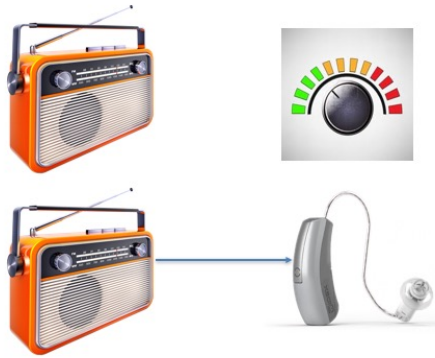
Essential for success



Any real ear programming information depicted in the manufacturers fitting software is simulated unless the system has some sort of probe microphone inserted in the users ear canal. Simulations of course are based on averages and are simply a starting point when fitting hearing technology. Using probe microphone measures to ensure that the signals being delivered to the eardrum are as expected should be standard practice in virtually all fitting situations. It is particularly important when listening to music as the professional needs to ensure that

	Maxmium Power Outputs are not being exceeded at high levels.
<p>If you don't measure – you don't know</p> 	BLANK
 <div data-bbox="598 689 842 1115" style="background-color: #4a7ebb; color: white; text-align: center; padding: 20px;">LISTEN</div>	<p>Virtually all Probe Mic Measurement systems include a 'monitor headset' option and I believe that just listening to what a hearing instrument is doing whilst in your client's ear is one of the most valuable tools that you can use to better understand any problems that might exist. I honestly believe that not a lot of people do this..... Having access to the sound environments where the client experiences problems is also very important but not always practical of course but best efforts should be used to create as realistic an environment as possible.....</p>
 <div data-bbox="92 1305 842 1731" style="background-color: #4a7ebb; color: white; text-align: center; padding: 20px;">PRACTICAL SOLUTIONS</div>	<p>We've talked a lot about the limitations of technology and some possible solutions that can help with careful selection and setting up of hearing technology. There are of course, also some really simple tips that can make a difference too..</p>





If listening to recorded music either  
Reduce the input level to the hearing aid microphone and increase the gain if required  
Stream directly to the hearing instruments where again input level may be controlled.

## THE SELLOTAPE TRICK



3-4 Layers of tape

Approximately 10dB reduction

Minimal affect on frequency response

A low tech solution which can prove very effective is to cover the microphone openings with 3-4 layers of sellotape. This has the effect of reducing the level of sound reaching the microphone by about 10dB without altering the frequency response significantly and can therefore provide a more suitable input to the A-D converters

OR..... Take the hearing instruments out.....

dB HL at 1000 Hz	65 dB input	80 dB input	95 dB input
25	2	1	0
35	8	4	0
45	14	7	0
55	20	10	1
65	28	15	2
75	36	20	3
85	44	24	4

↑  
PTA

↑  
Gain requirements at different input levels

Chasin, FIG 6

Depending on the patient's hearing loss and bearing in mind that music is often at a much higher level with higher peaks then it may be appropriate and more natural for the patient to remove hearing aids completely as these figures indicate the gain requirements at high levels can be quite minimal and sometimes even zero