HANIWA REAL 3D AUDIO

Full Bandwidth Integrated Phase Control

REAL 3D Audio System

HSP100 Series

Clear Focus Speaker

HDSA100

Full Range Digial Phase Control System Amp

Full Bandwidth Integrated Phase Control System Amp
with GaN Output Stage.
Throughout the Entire Musical Bandwidth,
the input waveform is accurately
played back as TrueSound.

HANIWA Audio

REAL 3D AUDIO Mission :

Reproduce the live performance in front of listeners with vivid expression and spatial reality.

We have developed the method to mine for the essential information buried in the music sources, and to reproduce the original sound authentically and vividly. For this end, we expanded our technology targets over all the audio components, from cartridges through speakers, and we re-designed a radical equipments for the optimum system.

REAL 3D AUDIO Technology Initiative

1. Deliver the sound having the same waveform as the input music signal.
2. Deliver the accurate sound space by keeping the timing of L- and R-channels.

1. Why the **Waveform?**

Do you believe your audio system truly reproduces music as it was recorded? For casual listeners streaming through earbuds, this question might seem irrelevant. But for audiophiles who know the vibrant, unfiltered sound of live music, it’ s a critical consideration—one the audio industry rarely addresses. Many manufacturers claim their sound is "authentic," but this often reflects a subjective flavor rather than a commitment to faithfully recreating the original music waveform.

In audio engineering, authenticity hinges on technical precision. We often rely on frequency analysis, specifically the frequency curve (f-curve), to evaluate equipment. A flat f-curve across the audible spectrum suggests consistent gain—a hallmark of quality. Yet, this tells only half the story. To reconstruct a waveform with true fidelity, you need both the f-curve and the phase curve (p-curve). The p-curve measures time delays across frequencies, and for waveform authenticity, it must remain as flat as the f-curve throughout the music bandwidth.

So why is the p-curve overlooked? Historically, it’ s been incredibly challenging to control. Traditional analog audio systems, built on analog technology where phase (timing) is difficult to manage, struggle to maintain consistency. Add to this the fragmented nature of the industry—components designed by different companies with varying sound philosophies—and the problem compounds. For instance, a filter engineered for a flat f-curve inevitably distorts the p-curve. In a multi-component system, each piece introduces its own uncontrolled phase shifts, leaving the final p-curve chaotic. The result? The output waveform bears little resemblance to the original input.

Haniwa Real 3D challenges this status quo, pursuing a singular goal: authentic music reproduction through waveform integrity.

The Limit of Analog Audio Technologies

Most Audio Equipment consists of electronic circuits, speaker units and cabinets, etc. These components depend on physical phenomena to process and express musical information. In short, most all of these are based on analog technologies.Physical phenomena of analog technologies are always restricted by TIME, which is totally uncontrollable. It is also true that physical values, such as voltage, current, static capacitance, induction coefficients, etc. can be treated as mutually proportional only within limited ranges. Outside of the proportional range, the relation among those physical amounts are “distorted” .

The human audible range is from a few Hz to tens of thouhsand of Hz; ... its range is 10,000 times.   This wide range of signal cannot be handled with a single piece of equipment. That means the signal range must be divided and put it back into one signal. During the process of “Divide and Connect” , signal timing is skewed and other distortions are accumulated. So, it is logical to keep the original signal wave intact throughout this process. Even if analog electronic circuits could cover the entire audible frequency range, when the signal is passed through many physical elements the sound waves become distorted.

Potential of Digital Signal Processing

Digital technologies can express the music signal with an abstract number, and that has merit as follows.

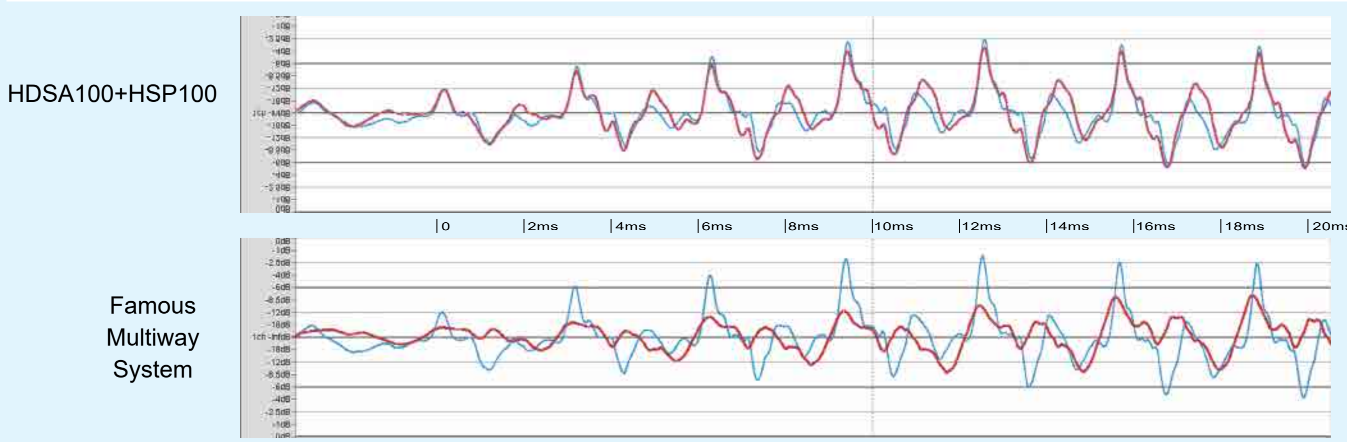
- Timing of signals can be adjusted with large and stable memory equipment.
- The wide range of data can be handled without loss, by increasing the number of digits.
- By formalizing a process in the programming Language, any signal processing is possible.

Digital limitation factors are the capacity of memory and processing speed. These limits are continuously improved by the endless progress of digital technology. For example, even today, 192kHz sampling enables 10Hz-96kHz signal processing The performance of memory and digital signal processors are available for the audio signal processing in REAL TIME.

EXAMPLE : HANIWA Waveform Processing Capability

HDSA100+HSP100 compared with a famous Multi-Way System

Blue Curve = Input Signal   Red Curve = Measured waveoform data “Art Pepper meets The Rhythm Section”

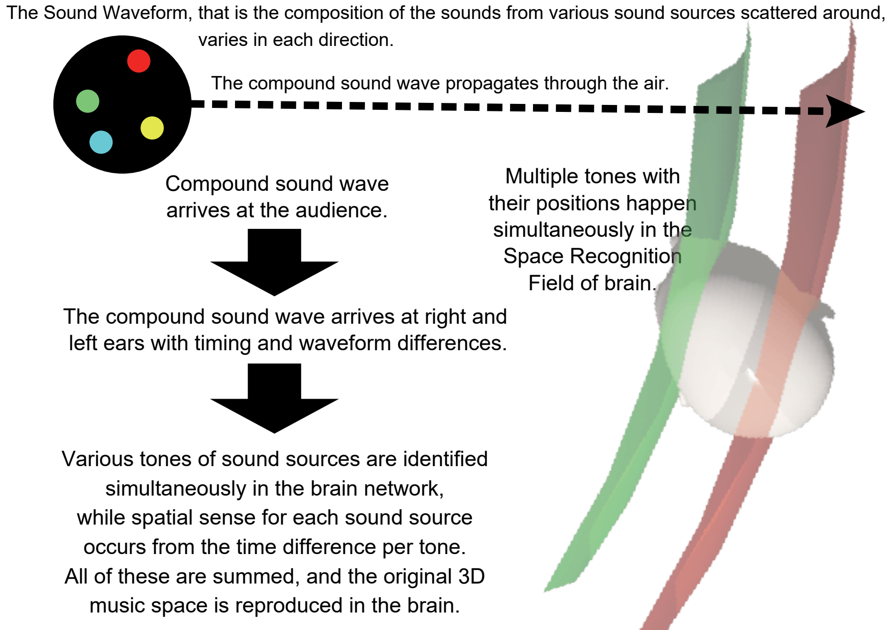


Two graphs above compare the HANIWA system with a famous multi-way speaker. The test sound is “Art Pepper Meets the Rhythm Section” . The first graph is the HANIWA system, and the second graph is the famous multi-way system, and the red curves show the output wave forms against the same input signal shown in a blue color.
The common input signal in blue color has sharp upward peaks, that are typical to the starting waveform of brass or sax instruments.
It should be noted that, with the famous multi-way system, the sharp starting peaks disappear, and its waveform is closer to the vertical symmetry, and the characteristic sharp start-up peaks are dull and delayed. This sample shows that HANIWA system output is reproducin the input waveform, and the charm of the original sound is well preserved.

2. Reproducing the Music Space

When a human recognizes 3D space using their auditory senses, the arrival time difference of sound between left and right ears plays an essential role. Since live performances are played in a hall filled with various musical instruments, this timing information is critical for the reproduction of the original performance in 3D space with clarity and reality.

The Process of 3D Space Recognition by Sound



The reality of the original sound space is recognizable only by using technology of precise and accurate sound wave reproduction.

The sense of sound has been essential for animals as their means of survival, to avoid danger and to find and catch prey. It has also been essential to warn of natural disasters and predators.

In order to vividly reproduce live music, this fundamental nature of the sense of sound should be well understood as the basis of audio equipment design and development.

A key point is to understand what is happening inside our brain while we are enjoying a live musical performance.The sound of music arrives at the left and right eardrums with slight difference in timing. If the sound of an instrument is closer to the left ear, it arrives at the left ear slightly before the right ear.

As this sound stimulation propagates over the brain network, a number of individual sounds are identified simultaneously and in real time. The recognition of timing differences for each sound source is also recognized at the same time.

Throughout this process, highly delicate features of the sound waveform are recognized, and with these features, the positions of each sound source are quickly determined in the space.

All of this information is woven into one consolidated and changing image of the performance. That is, the recreation and feeling of the music performance is felt from all directions.

Waveforms Reproduced with Haniwa Can Reproduce Live Music with Precision

By understanding the mechanism of hearing, we can clearly appreciate the importance of correct waveform reproduction by the audio system. For the listener to feel the music with the right presence and a vital vividness, the accurate reproduction of the sound waveform features by maintaining precise timing , is te key. The delicate touch of piano keys and a violin bow, the impact of a tuba or kettle drum can only be reproduced by the accurate reproduction of the original sound waveform.

“Audio systems should not make up the sound.” This is Haniwa's core belief.

3. Haniwa Solution : Full Range Integrated phase control

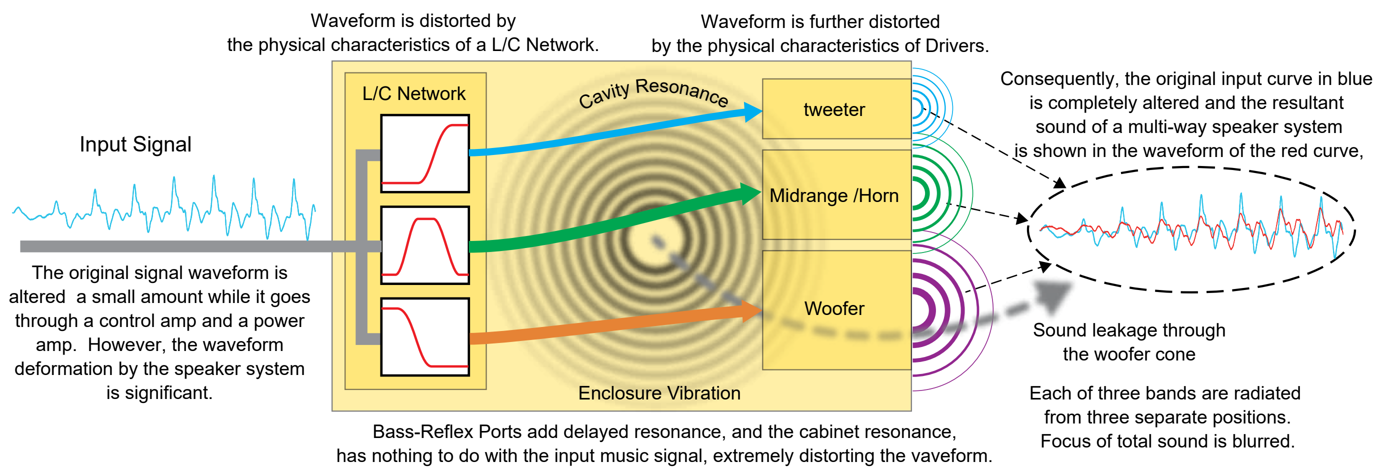
After a long search for the best speaker to reproduce waveforms accurately,
we came to the conclusion that
"we need a system that controls the full bandwidth in an integrated way",
rather than outputting each frequency band separately.

A simple solution may lead to the choice of a multi-way speaker, because their sound may be richer, and suitable for full scale grand music. It is widely believed that in order to avoid an inappropriate change of sound, the audio bandwidth should be divided into a few bandwidths, say, low, middle and high ranges, and for each range, choose drivers maintaining a flat frequency response in each range. So, as a system, it can have fairly flat frequency response curve. The biggest falicy of this method is that it considers only the amplitude of the frequency response. The phase shift of the requency response is ignored, consciously or inconsciously, resulting in a severe deformation of the output waveform.

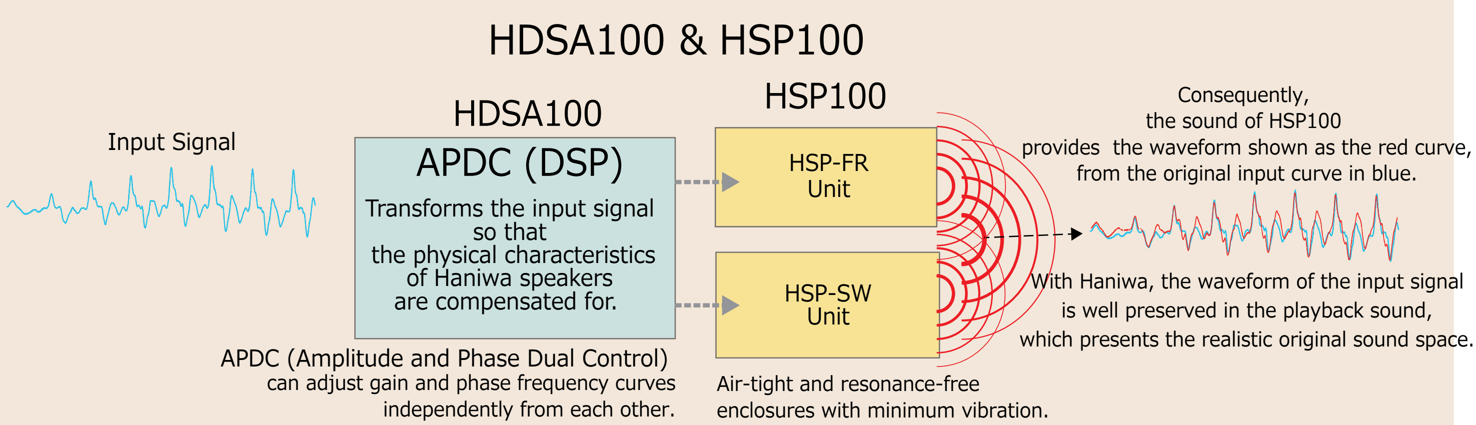
The figure below shows how the sound waveform becomes deformed as it goes through a conventional multi-way speaker system. The music signal passes through the control amp and the power amp, without significant deformation. It is then fed to L/C crossover network of the speaker system and is divided into high, mid and low bands. At this point, the waveform has been distorted by the L/C network and filter circuit. They drive one or more tweeter, midrange/Horn and woofer units respectively, additing further distortion by each unit. Thus the whole sound is a mixture of time and positional confusion.

The entire process distorts the waveform, and even worse, the speaker enclosure adds resonance and vibration by itself.

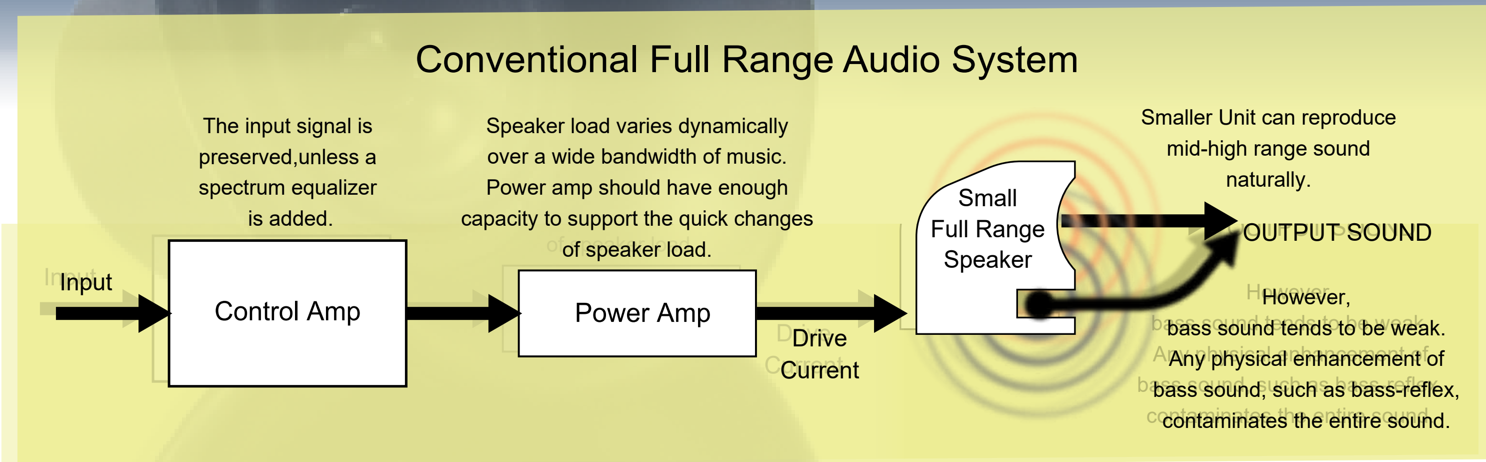
The Input Signal Waveform is deformed with Multi-way Speaker Systems.



Input Signal Waveform is precisely reproduced with HANIWA Real 3D Audio.



HSP100 Full Range



Elastic Suspension Technology

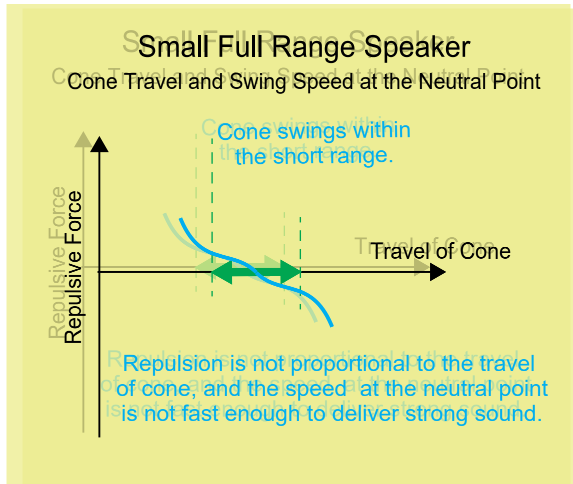
It enables small speakers express powerful performance of bass instruments.

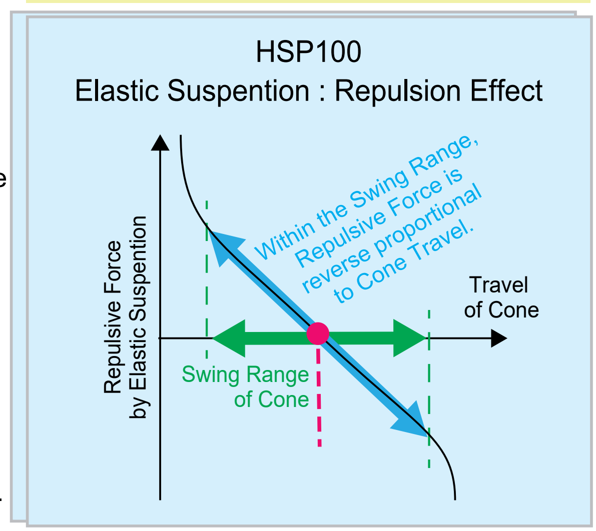
"In order to deliver ample bass sound, a large woofer is necessary." This is the common thinking of in the audio world. However, there is some misunderstanding about why.

To begin with, the sound is the wave of air pressure, and the pressure is generated by the velocity of cone's piston motion. Large speakers have longer stroke of the motion. So, even at the low frequency, the cone can have sufficient speed to generate the sound wave. The diameter of cone is related to the bass sound of woofers, but the quality of bass sound needs to be carefully considered.

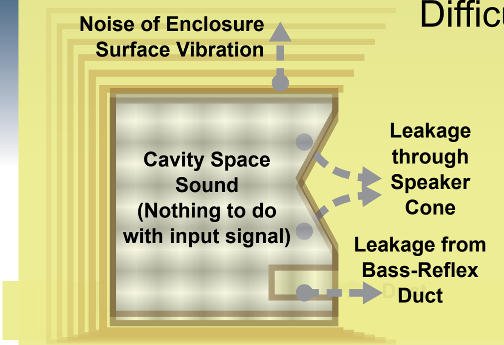
Small full range speakers are appreciated for their natural mid to high range sound, but their bass sound is always poor. Because of its small size, the cone cannot move fast enough at lower frequencies, to generate sound. With Haniwa's HSP100, this defect is overcome by doubling the unit's stroke, and by adopting this elastic suspension technology that increases the return speed from the maximum swing position.

The cone of HSP100 is powerfully controlled by a Current Driving Method. Fleming's Left Hand Rule tells that a strong magnetic field and the quick response of current is necessary to generate a powerful cone driving force. Vivid and dynamic sound is realized by driving a low impedance voice coil of less than 1Ω by high current, in a strong magnetic field created by the newly designed magnetic circuit.





e Clear Focus Speakers



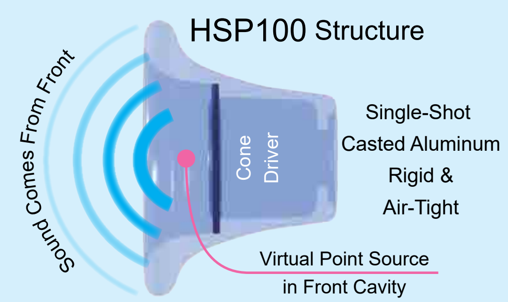
Difficulty of Enclosure Design

There is a kind of noise, known as Enclosure Sound. This is generally understood as the noise of the enclosure surface vibration, generated when the speaker is driven to play music. The ordinary countermeasure for this is simply to stiffen the enclosure structure.

However, inside the cavity space itself, there is another noise that has nothing to do with the input music signal. This non-negligible noise is radiated outside through the speaker's cone. The resonance duct of Bass-Reflex also leaks this noise.

Enclosure Design of a Clear Focus Speaker

Virtual Point Source / Monolithic, Rigid and Air-tight : Leakage and Resonance Free



Full Bandwidth comes from one Point Source.
Sound space gets clearly focussed,
and describes sharp and vivid music scene.

HSP100's enclosure shape is designed to have a Virtual Point Source in the cavity in front of a Cone Driver. Audiences feel that all the sound is radiated from that one point.

It is formed as a one-shot monolithic casting of Aluminum, to make it highly rigid and air-tight, and its cavity shape is designed to minimize cavity space noise. Any noise from other than the front face of the system is also minimized in all directions.

The speaker unit has the sturdy aluminum cone, and is set with a hermetic seal to shut off any sound leakage.

HSP100 Product Line

- One-shot cast Aluminum enclosure is air-tight, shaped to be free from harmful resonance and vibrations. The silence and the dynamism of sound is realized by this new common design concept.
- There are three types of Full range units (-M, -N, and -U) of HSP100, to match the various types of music sources.

Type-M	with 13cmφ unit :	Cone=φ10cm, Voice Coil=φ30mm (DCR=0.8Ω)
Type-N	with 16cmφ unit :	Cone=φ13cm, Voice Coil=φ36mm (DCR=1.0Ω)
Type-U	with 19cmφ unit :	Cone=φ16cm, Voice Coil=φ36mm (DCR=1.0Ω)

- HDSA100 has three presets of signal processing parameter library, optimized for each type.
- The 13cmφ small cone driver of -M is light weight and highly rigid with extremely low voice coil impedance of 0.8Ω. It plays back the full music bandwidth, and is controlled optimally to give the precise large scale spacial expression.
- A larger format model Type-U is more focussed on the expression of grand bass instruments. It has a 19cmφ cone driver, matched with the sturdy enclosure. It is fundamentally different from the conventional speakers with a large size woofer reproducing natural bass sounds with precision, impact and clarity.

HDSA100 Digital Phase Control System Amp

HDSA100 is a System Amp that is the core of whole Haniwa REAL 3D AUDIO System.
All the processes, from the beginning to the end of music reproduction, are consolidated in it.
It has precise digital phase control capability to compensate the physical characteristics of speakers.
Coupled with HSP100, it realizes the next generation audio system, realizing
the truely authentic sound to the original input waveform,
keeping original input signal waveform.

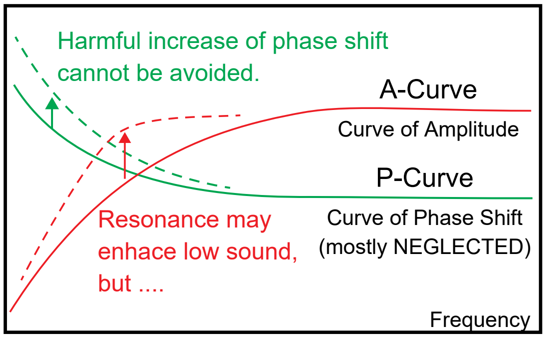
WANTED "True Bass Sound" Overcoming the Physical Limit of Speakers

Generally speaking, the sound of a dynamic speaker gets weaker toward lower frequencies, while also experiencing more phase shift. This tendency is inevitable because of physical limitations of speaker drivers.

To overcome these limitations, various countermeasures have been taken by manufacturers, such as enlarging the speaker cabinet. Today, many smaller speaker models have been introduced to the market boasting bass enhancement, but they use resonance to accomplish this.

Resonance may enhance the volume of lower sound frequencies, but resonance starts only after a time lag to the original sound. So, a lagged low resonant sound lingers after the original sound, like a vague shadow.

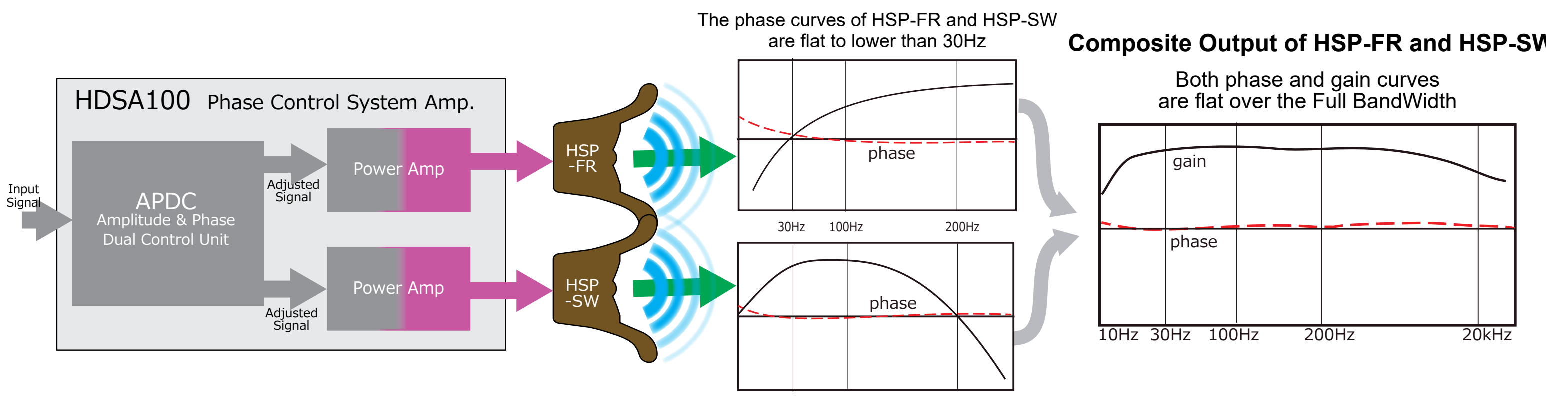
Consequently, although resonance provides more bass volume, it contaminates the waveform, and the resultant sound becomes blurred with less impact.



Bass enhancement by resonance
is accompanied by harmful phase shift.

Preserving the WAVEFORM = Signal to Sound conversion with a flat A-Curve AND P-Curve

In order to preserve the waveform, the Frequency Analysis Theory requires that both the gain and the phase lag should be constant along the frequency. That is, the Amplitude curve (A-Curve) and the Phase Shift curve (P-Curve) must be flat over the frequency range. This demand, especially concerning phase shift control, is impractical to accomplish using analog technology which depends on physical phenomena to solve the challenge. Phase shift control involves handling the time that is not altered arbitrarily by analog technology. The solution of this this phase problem, is digital signal processing technology.
Waveform preservation is not an easy problem to solve. It needs a deep understanding on both analog and digital technologies, backed up by skilful implimentation. Haniwa's APDC (Amplitude & Phase Dual Control) technology is the result of our efforts, to controls A-Curve and P-Curve independently, and it is implemented in HDSA100 "Phase Control System Amp". This unique system amp is the core of Haniwa REAL 3D AUDIO.



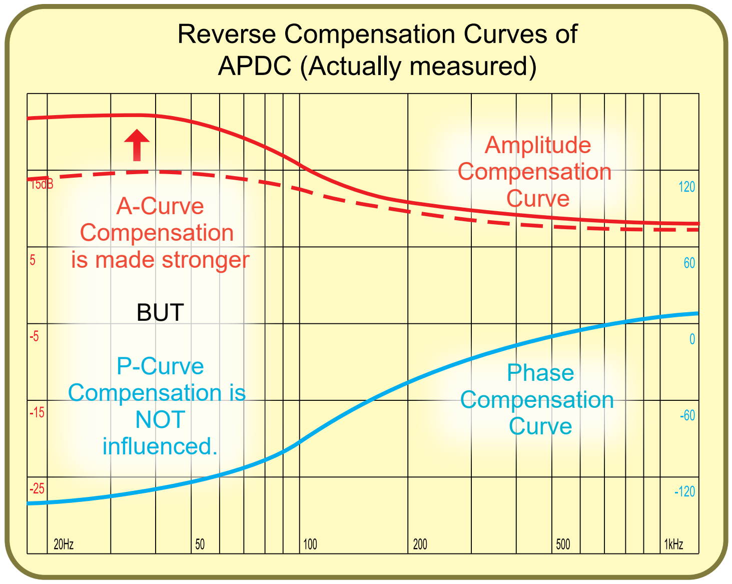
APDC Amplitude & Phase Dual Control

The Figure on the right shows the A and P- Compensation curves used in Haniwa's APDC, to perfect HSP100's sound output. It should be noted that the P-Compensation curve is not influenced while the A-Compensation curve is made stronger. It is impossible to control A and P-Curves independently using analog technologies

There are, equalizers that control the frequency response of the signal , intending to improve the sound quality. Most of them are based on analog technology, and recently Digital Graphic Equalizers are also available. However, this equipment controls only the A-Curve, and the induced change in P-Curve is ignored, even though the degradation of P-Curve seriously damages the quality of sound. Eventually, many users tend to eliminate these equalizers.

To the contrary, APDC is totally different and revolutionary, because it utilizes sophisticated digital technology to realize the independent control of amplitude profile from the phase profile.

This innovation advances audio technology well beyond what was possible before Haniwa.

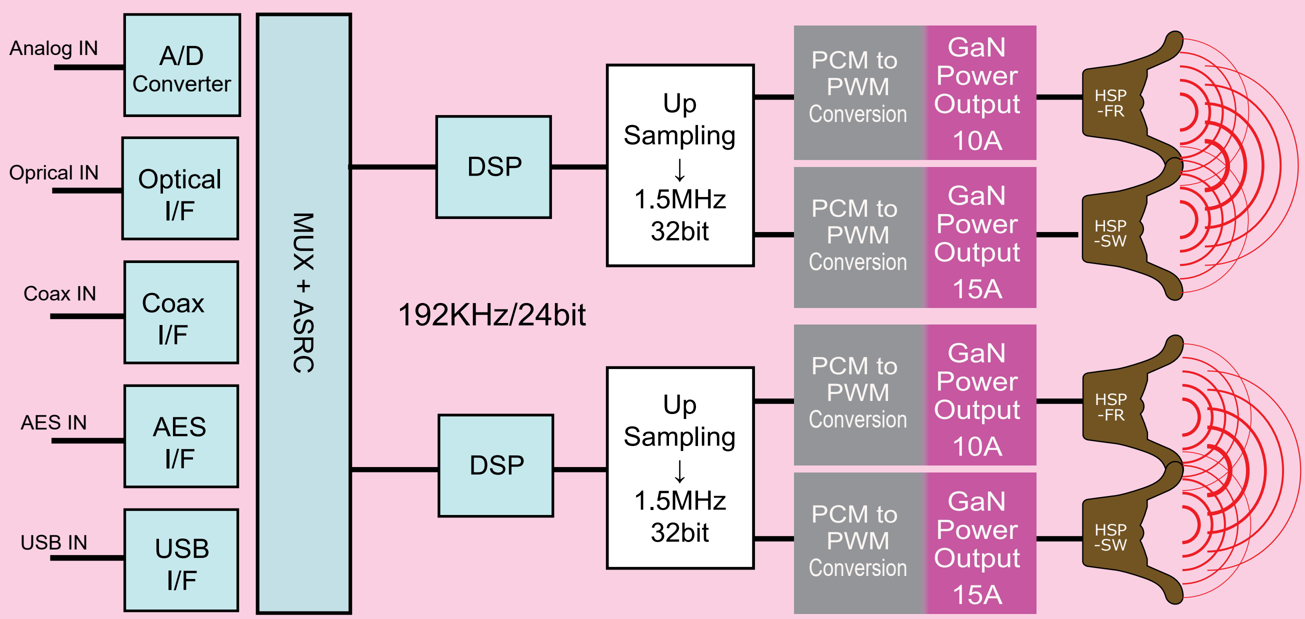


POWER AMP UNIT Leveraging the Latest Technologies

- The drivers in HSP100 speakers have Voice Coils with DCR less than 1.0Ω.
- They are driven by a Power Amp Unit that provides a highly responsive driving current.
- The driving current is provided using a GaN Power Output circuit.
- GaN technology has a very high switching speed, and can accurately react to rapid transients and wide dynamics of sound typicaly found in live performances.

192kHz/24bit PCM is over-sampled to 1.5MHz 32bit PCM, and then converted to PWM.

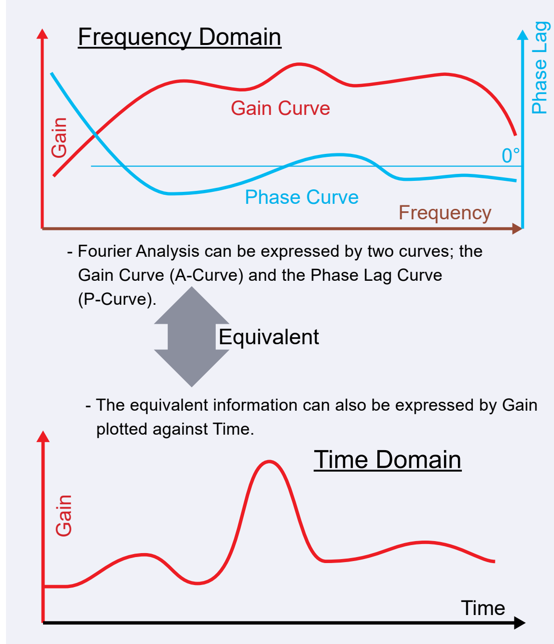
HDSA100 Flow Chart



Surpassing a conventional low resolution digital amp.
Super-high resolution of 1.5MHz/32 bit is realized.
The quality of sound, exceeds by far, any high-end analog amp.

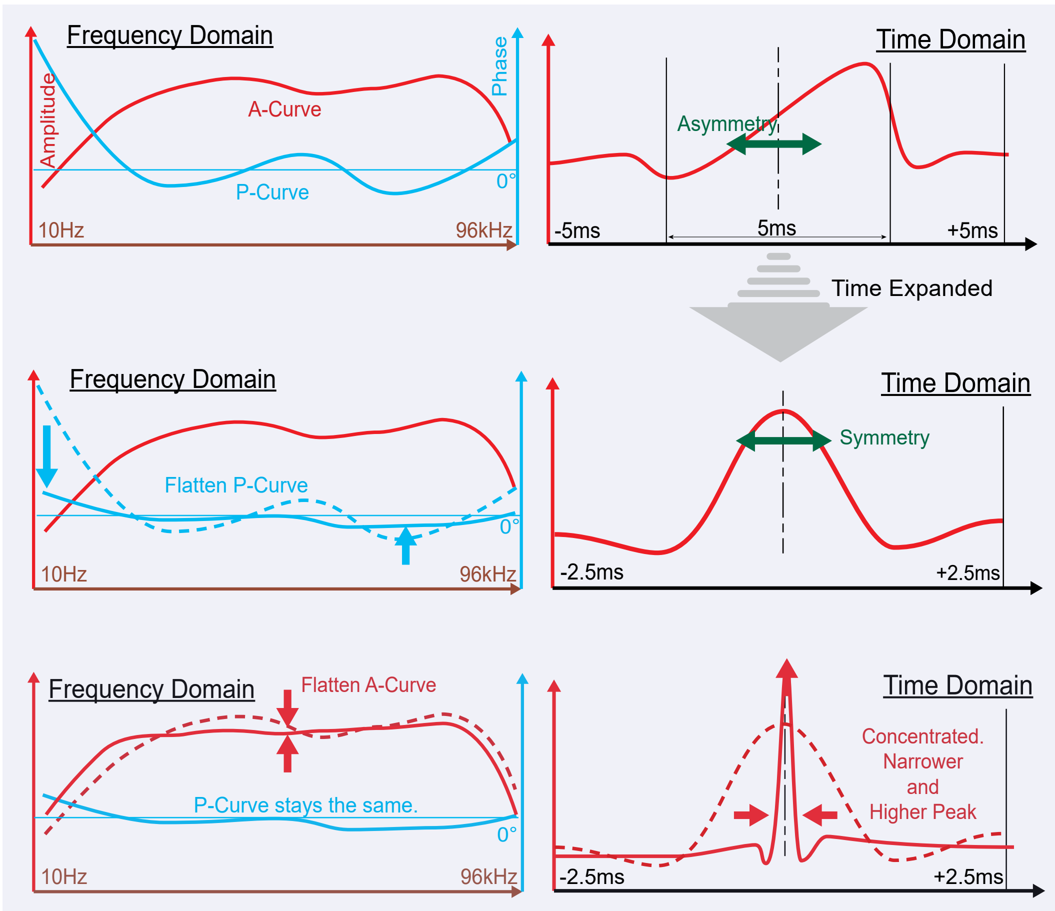
6. Frequency Domain and Time Domain Expressions of Audio Analysis

- Fourier Analysis is a general tool for analyzing system behavior against various inputs.
- The Fourier Analysis data is expressed in the Frequency Domain and is plotted against frequency, while the Time Domain data is plotted simply against time.
- These two ways of expression show the result of the same audio signal analysis.
- To show the equivalent amount of information in the frequency domain against the time domain, both the gain and the timing characteristics are necessary.
- In the Frequency Domain, the gain is expressed by a Gain Curve, and the timing is expressed by a Phase Curve plotted against frequency.
- These two curves together, carry the equivalent amount of information to the gain curve in the Time Domain.
- In the audio industry, the Gain Curve is measured and published as “Frequency Response”, while the Phase Curve, representing timing characteristics, is not often shown. Systems are evaluated mostly by Gain Curve, and the “waveform”, that carries important music information along the time, is not considered.
- ***Haniwa recognized the importance of Phase Characteristics at the early stage of development, and has developd technology to control both Gain and Phase Curves independently. The new products of HANIWA are built on this radical technology, and are now available in the market.***



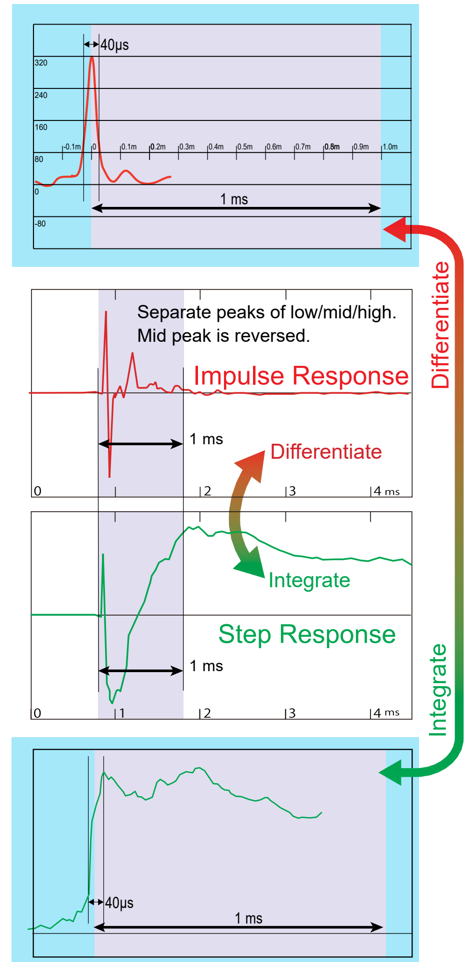
HANIWA Audio System Improvement Process

- In the audio industry, gain is called Amplitude.
- Music information is spread over 10Hz ~ 30kHz. For keeping precision, the measurement is done at a192kHz sampling, and the analysis frequency range is up to 96kHz. In the Time Domain, -5ms ~ +5ms region is closely observed.
- APDC (Amplitude & Phase Dual Control) is the method used to **flatten both the A and P-Curves**.
- The following figues show how the Time Domain expression changes by improving Curves in Frequency Domain.
- First, the P-Curve is flattened by APDC, then in the Time Domain, response of the full bandwidth appears with the same timing, and the Curve shape becomes symmetrical along the time axis.
- Then, as bumps of the A-Curve are flattened, the Time Domain response gets concentrated.
- As the A and P-Curves are flattened, response of all frequencies cancel each other to make the skirt part flat, leaving one central peak.



HANIWA vs High-End Multiway System : Comparison in Time Domain

- The impulse response figure on the right shows the HANIWA System performance, and in the Time Domain, the response is concentrated within ±20μsec range. This is only 8 clock cycles wide at a 192kHz sampling rate, which is the measurement limit. This level of sharpness of response has not been achieved to date by any audio system at any price.
- This measurement value means “The impulse response in the Time Domain is close to 1.”



High-End Multiway System

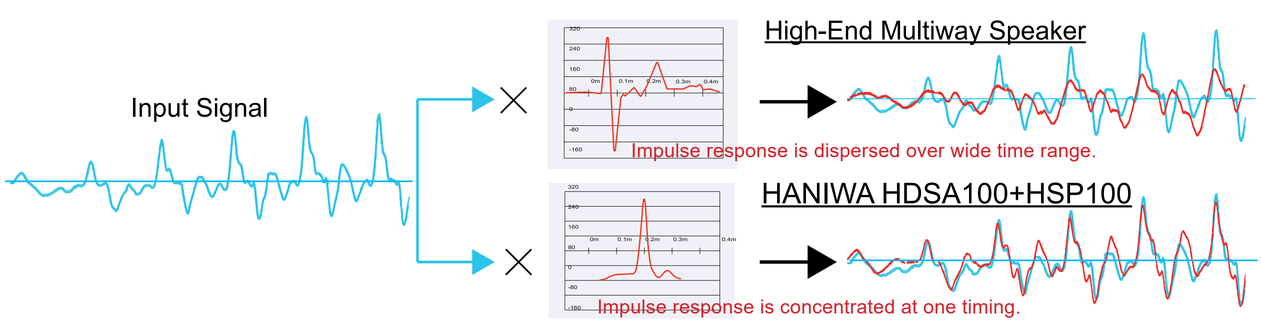
- The Step response of aformentioned high-end system is published as time domain data.
- Impulse response is suitable for observing mid to high range behavior, but not for low range observation. The low range response is distributed over a wide time range.
- The integral of impulse response is the step response, and vis-a-versa.
- The step response of this system shows that the response is spread over 1000μs, and is asymmetric, meaning the timing of high, mid and low frequency is widely shifted.
- Especially, the delay of low frequency is significant.
- High, mid and low range peaks are appearing multiple times and with the positive and negative polarity.

HANIWA HDSA100 + HSP100

- Step response of HSP100 sound is shown on the right.
- It rises up within very short time of 40μsec , and then falls down slowly.
- As shown above, its impulse response is rising and falling within 20μsec each, and has a single and sharp peak.
- This results in the clean step response, without the separation of high, mid and low ranges.

Relation between Impulse Response and Output Waveform

- Impulse response reveals how the input signal is affected by the entire sound reproduction system.
- Input to the audio system is the music information expressed as the waveform, and its response is the sound waveform.
- Input signal is transformed to sound by the system with a specific impulse response. If the impulse response is close to 1, the output sound waveform is closer to the that of input signal, and becomes almost identical to it.
- The actual waveform comparison is shown below, using actually measured waveform data.
- Conventional high-end multiway speaker has the impulse response spread over the wide timing range, also with positive and negative peaks. This resulted in the heavily deformed waveform, losing sharpness of sound, with low peaks and much lower wave height.



- HANIWA can reproduce the waveform of recorded music signal authentically, meaning that various musical instruments’ tones are precisely reproduced, and the details of a virtuoso’s performance are clearly articulated with their spatial positioning.
- This highlights the essense of Haniwa REAL 3D Audio, “Bringing music reproduction as close to the original performance as possible”.

HANIWA
FULL RANGE REAL 3D AUDIO SYSTEM

This product has been developed with the clear technology mission of "Transfer the input musical signal to the output sound, with the highest accuracy." It is not simply transfering the input signal, but rather recreating the space of performance in front of the listener. HANIWA Real 3D Audio System is a 2-Way configuration consisting of HSP-FR(Full range) and HSP-SW(Super Woofer) .

HDSA 100 Digital phase Control System Amp.

HDSA 100 precisely controls both signal gain and phase. HSP-FR provides an extremely accurate responce, and HSP-SW reproduces powerful and impactful low frequencies. Their output stages use GaN elements to provide large power efficiently.

Input	Analog	RCA:2ch(stereo) ±20dB variable gain
	Digital	Optical : 2ch / Coax : 2ch / AES : 1ch
Output	Speaker Drive:	HSP-FR (Full range) 10A at 1.0Ω HSP-SW(Super Woofer) 15A at 0.9Ω
	Analog	RCA : 1ch (stereo) for External Power Amps
USB Port		USB 2.0 (bi-directional connection with PC)
APDC Function		Gain & Phase curves mutually independent

HSP100 Clear Focus Speakers

- One-shot Cast Aluminum body is air-tight and resonance-free.
- Sound output port is shaped to form a Virtual Point Source. Accordingly, the sound focus is very sharp, and the 3D space positioning is clean and clear.
- The speaker unit was also developed from scratch, without being bound by conventional thory of speaker design.
- In the HSP-FR (Full Range) model, the cone movement is designed to follow even the most rapid changes in the input signal with ease, allowing it to deliver sufficient energy to accurately reproduce sound
- The HSP-SW (Super Woofer) is completely different from the conventional Woofer, and has a closed cabinet equivalent to the HSP-FR, but with a very small volume. Unlike conventional Woofers that try to boost the low range by using cabinet resonance, the HSP-SW delivers a low clear sound with impact.
- According to "Fleming's left-hand rule," the force generated within a magnetic field is proportional to the strength of the field and the current flowing within it. However, in the audio industry, loudspeaker performance is expressed as a relationship between power consumption and force, which has led to a great misunderstanding.
- The HSP100 series units, both HSP-FR and HSP-SW, have an extremely low impedance of less than 1Ω. And the power amplifier that drives them employs GaN elements that are suited for current output and thus provide excellent current output.
- The HSP100 series has an extremely high efficiency and low power consumption. In particular, the HSP-SW has a VoiceCoil diameter of 60mm and an extremely thick coil wire diameter of 0.6mm. The impedance is less than 1 ohm, which is the smallest impedance in the world and beyond common sense for a woofer unit. It is driven by a GaN high-current output amplifier with extremely high efficiency.
- The HDSA100 that drives the HSP100 series has an extremely high resolution of 1.5MHz 32bit PWM. Its sound quality surpasses even the highest of high-end analog amplifiers.

Characteristics of Three models of HSP100 Series
- Type M: accurate and detailed
- Type N: standard type
- Type U: larger scale music presentation

	Type M	TypeN	TypeU
HSP-FR UNIT	13cm	16cm	19cm
Impedance	0.8Ω DCR	1.0Ω DCR	1.0Ω DCR
HSP-SW UNIT	19cm	19cm	22cm
Impedance	0.9Ω DCR	0.9Ω DCR	0.9Ω DCR

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