
HANIWA REAL 3D AUDIO
HSP01
Full Range
Clear-Focus Speakers
This Compact System Delivers
the Grand Music Performances of True Virtuosos,
with Almost Touch-able Reality
HDSA01 Digial Phase Control System Amp
Haniwa
INPUT
UP
DOWN
POWER
MODE
MUTE
-20.0
Digital System Amplifier HDSA01
Integrated REAL 3D AUDIO SYSTEM
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 R- and L-channels.

1. Why the **Waveform?**

Do you believe that audio systems are replaying the music exactly as it is recorded? This question may not make much sense if you are not interested in THE AUTHENTICITY of the sound coming out of electronic gadgets. But if you are an audiophile, you should have the serious doubt about this question, and the doubt is fair. The audio industry is not answering this question. They may even avoid to answer this question. When they say AUTHENTIC, it mostly means that they are royal to their OWN TASTE OF SOUND, and does not mean that they persue the objective technology target such as KEEPING THE SOUND WAVE FORM.

By the way, when we talk technically about audio equipments, we depend on the Frequency Analysis, and practically use the frequency curve (***f-curve*** in short) of equipments. If f-curve were flat all through the audio frequency, that equipment is stamped AUTHENTIC. But you should be careful, because this f-curve shows only about the gain. The basic knowledge of frequency analysis tells there are two curves required to reconstruct the waveform; the gain curve and the phase curve (***p-curve*** in short). ***The p-curve shows how much of time delay happens at each frequency.*** In order that the WAVEFORM AUTHENTICITY is assured, ***the p-curve should also be flat through the music bandwidth as well as f-curve***.

Well, then why the p-curve has not been the subject of discussion? Simply speaking, ***it has been prohibitively difficult to control.***

It has been difficult, because the conventional audio equipments depend on the analog technology with which the phase (timing) factor is not fully controllable, and equipments are developed separately by companies whose sound policies are not consistent. For example, if a filter is developed trying to make f-curve flat, the p-curve of that filter cannot be flat at all. Furthermore, modern audio system is made from a series of components, and each component adds un-controlled p-curve. No wonder the final system p-curve gets totally confused. So, it does not make sense to expect that the waveform of the final output sound is similar to the waveform of the input music data.

Limitation of Analog Audio Technology

Most of the conventional audio components consist of electronic circuits, electric circuits, speaker units, the enclosures, etc. These components directly use physical phenomena to express and process the music information. As such, it is called ANALOG technology as a whole.

By the way, the physical phenomena is always under the restriction of PASSAGE OF TIME that is uncontrollable. It is also true that relations among physical quantities, such as voltage, current, resistance, capacitance, inductance, etc., are proportional only within some limited range, and out of that range, the proportional relations get distorted. The range of human audible sound is from a few Hz to tens of kHz, covering ten thousand times of variance. It is rather difficult to cover this wide range of signal by only one physical unit. So, “divide-and-patch” works become necessary. As the input signal goes through multiple stages of these process, varience of timing shifts per frequency and other deformation are accumulated, and the same for the waveform distortion.

Though the electronic circuit may cover the whole audible bandwidth, through the process of electric circuits, the speaker units and the enclosure, waveform preservation gets inhibitively difficult. Actually, the distortion of waveform happens most evidently in this electricity-to-sound conversion process.

Possibility of Digital Signal Processing

On the other hand, in the digital technology, the signal is expressed in abstractive number, having the merits of the followings.

- Timing can be shifted by using the large and stable memory.
- By adding digits, large amount can be handled without the loss of precision.
- Any signal processing function can be realized, as far as it is formalized in programming language.

The limitations are memory capacity and processing speed. These limitations are continuously overcome, by the dramatic progress of digital technologies. For example, by the 192kHz sampling, the signal ranging from 10Hz to 96kHz can be processed on the large main memory. The memory technology can already cope with the processing demand in both speed and capacity, while the processor performance is drastically improved with the advanced DSP (Digital Signal Processor), so that necessary time axis control has become feasible.

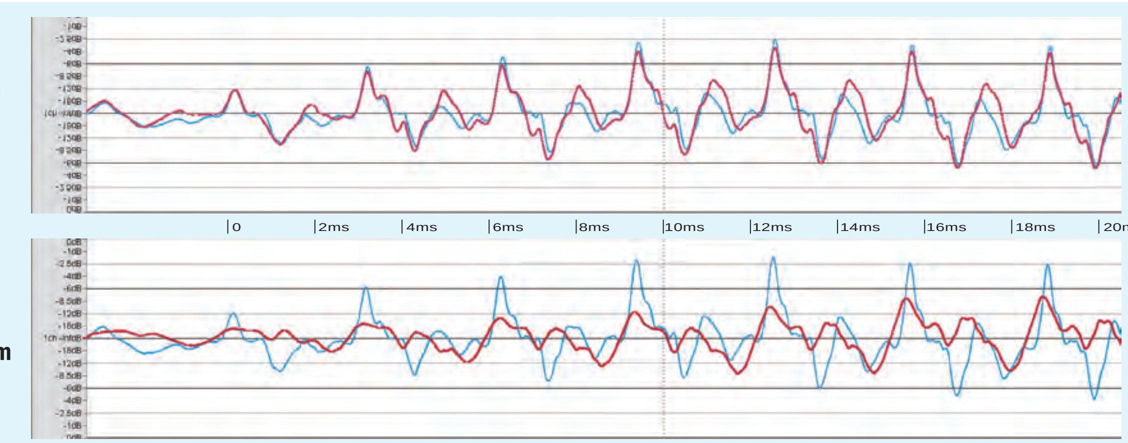
The Proof of HANIWA Waveform Preservation Technology

HDSA01+HSP01 Audio System vs A Renouned Multi-way System

Blue Curve = Input Signal Red Curve = Measured Sound Waveform Tested with “Art Pepper meets The Rhythm Section”

HDSA01+HSP01 Audio System

Renouned Multi-way System

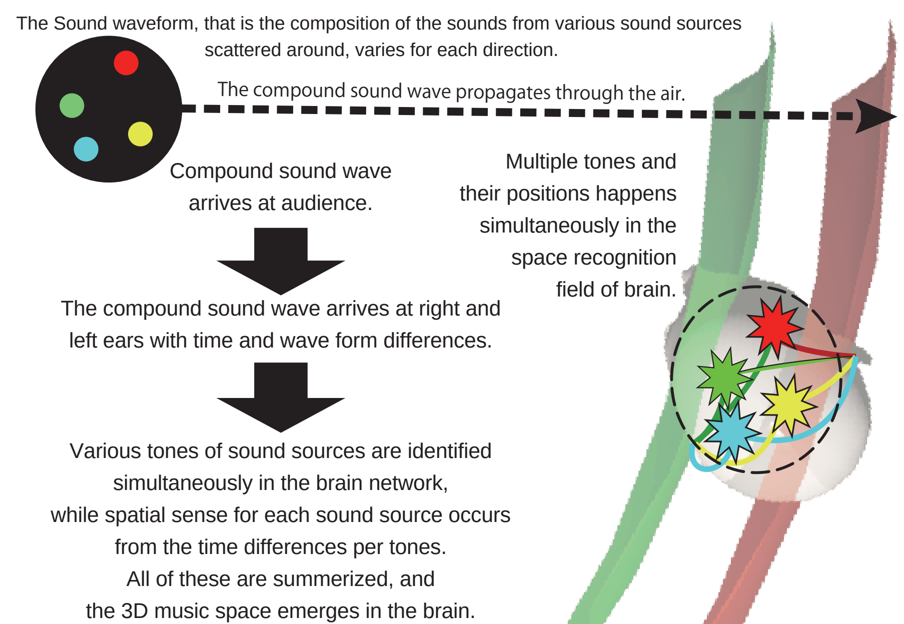


Two graphs above compare the output sound waveforms of Haniwa system and of a renouned Multi-way speaker system, when they play back the “Art Pepper meets The Rhythm Section.” On top is a graph showing Haniwa system and below shows a renouned multiway system, where the output sound waveforms are overlayed on the same input sound signal waveform.
The input sound (the same blue curves for both) shows sharp upward peaks, which is typical for the leading edge of brass and sax explosive sound. This characteristic waveform of input is well reproduced with Haniwa system, together with the timing and the peak height. The renouned multi-way system misses the sharpness of peaks, the waveform gets closer to vertically symmetric and the characteristic leading edge is dull and late, thus resulting in the less articulated sound. This experiment shows that Haniwa system can keep the input waveform accurately as the sound, and can play back the charm of original sound.

2. Reproducing the Music Space

When the human recognizes the 3D space by the audio sense, the arrival time difference between left and right ears plays the essential role. Since the live performances are played in the hall filled with various musical instruments, this timing information is critical for the reproduction of the whole performance with clarity and reality.

The Process of 3D Space Recognition by Sound



The reality of sound space is expressed only with the technology of precise and accurate sound wave transfer.

The sense of sound has been essential for animals as their means of survival, such as to avoid dangers and to catch foods. It has also been essential for avoiding natural disasters and predators.

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

By the way, what is happening inside our brain while we are enjoying live perfomances?

The sound of music arrives at left and right eardrums with slite difference of timing. If it is played at closer to the left ear, its sound arrives at the left ear slightly earlier than the right ear.

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

Throuhgh this process, highly delicate features of sound waveform are picked up, and with these features, positions of each sound sources are quickly decided stably in the space.

All these information are woven into one consolidated image of the performance. That is, the feeling of total experience of emotion by the superb music performance spreads out in all directions.

The Precision of Sound Waveform Brings back the Whole Nature of Live Music.

By understanding the mechanism of hearing, we can clearly understand the importance of correct waveform reproduction by the audio system. For us to feel the music with right presence and vital vividness, the accurate reproduction of the sound waveform features, by keeping the timing relations, is the key. The delicate touch of pianos and violins, the impact of tubes and drums can be revived only by the accurate reproduction of sound waveform.

"Audio systems should not make up the sound." That is what Haniwa insists.

3. Haniwa Solution : The choice of Full Range Single Unit Speaker

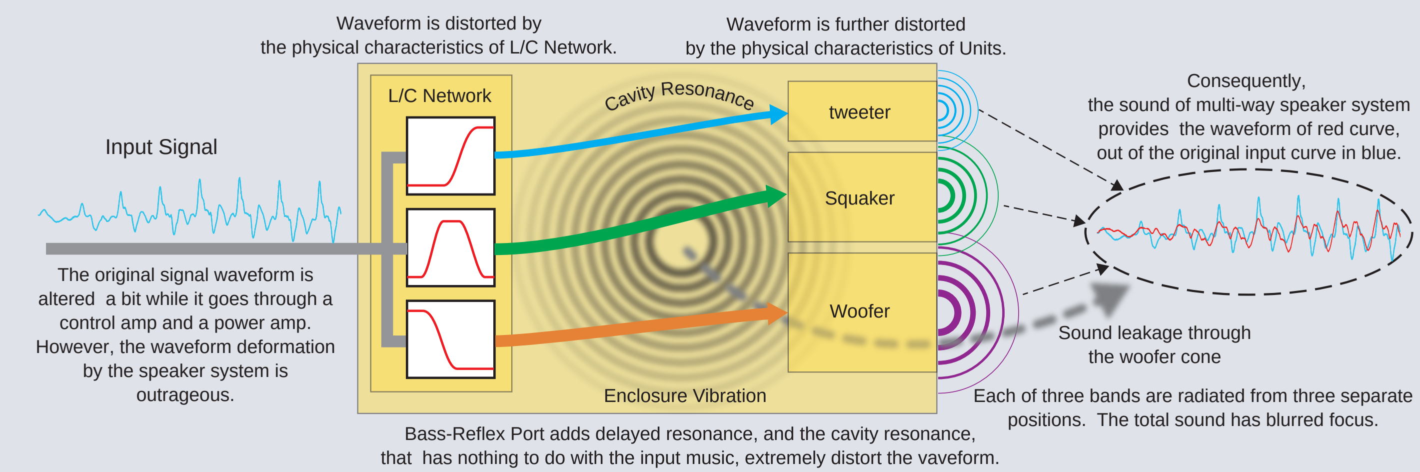
After the long research for the best speaker to reproduce the waveform accurately, we chose the full range single unit speaker system, while we found the multi-way systems is inappropriate especially for the correct spatial reproduction.

Simple thoughts may lead to the choice of multi-way speakers, because their sound may be richer, and suitable for the full scale grand music. It is widely believed that in order to avoid 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 units keeping the frequency response curve flat in each range. So, as a system, it can have fairly flat frequency response curve. The biggest defect of this thought is that it considers only the frequency response of amplitude. They leave the frequency response of phase shift out of argument, consciously or inconsciously, accepting the deformation of wavform.

The figure below shows how the sound signal waveform is deformed as it goes through the 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 network of the speaker system and divided into high, mid and low bands. At this point, the waveform has been distorted by the L/C networks of filter circuits. They drive a tweeter, a squaker and a woofer units respectively, additing further distortions by each unit. Thus the whole sound is a mixture of time and positional confusion.

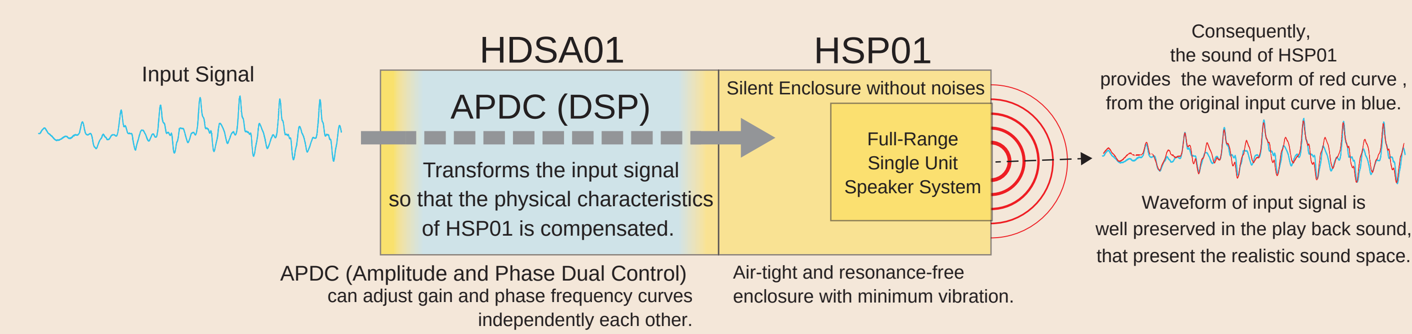
The whole this process distorts the waveform, and even worse, the speaker enclosure also adds the resonance and vibration by itself. Listeners are exposed to the confused sound that hides the true sound space infomation.

How the Input Signal Waveform is deformed with Multi-way Speaker Systems?



Haniwa System drives the simplest full-range speaker HSP01, by the technology enriching the bass sound and correcting sound waveform distortion. HDSA01 (Digital Phase Control System Amp) is quipped with the APDC (Amplitude & Phase Dual Control) that utilizes the DSP (Digital Signal Processor) to its limit. APDC compensates the original signal with the pre-measured adjustment for the HSP01 so that its output sound should have the closest waveform to the input signal. By this restoration of sound waveform, Haniwa System provides the realistic sound space with clearest focus.

HDSA01+HSP01 generates sound keeping the waveform closest to the original.

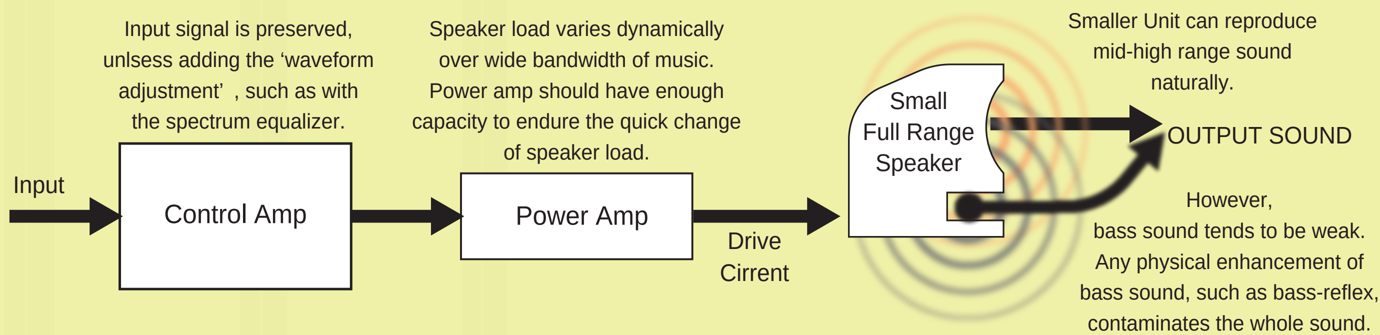


HSP01-S, -M & -L

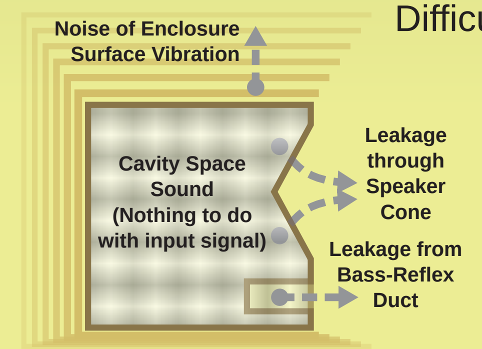
Full Range Clear Focus Speakers

Controllabillty and Silence Define the New Generation Speaker System

Conventional Full Range Audio System



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.

Elastic Suspension Technology

It enables small speakers express powerful performance of bass instruments.

"In order to present ample bass sound, the large woofer is necessary." This is the common knowledge of contemporary 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 has nothing to do with the bass sound of woofers.

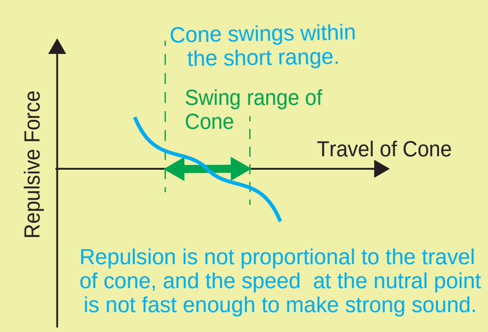
By the way, small full range speakers are appreciated for their natural mid to high range sound, but with their poor base sound. Because of its small size, the cone cannot move fast enough at lower frequency, to generate sound.

In HSP01, this defect is overcome by doubling the unit's stroke, and by adopting the elastic suspension technology that increases the return speed from the maximum swing position.

The cone of HSP01 is powerfully controlled by the **Current Driving Method**. Fleming's Left Hand Rule tells that strong magnetic field and the quick response of current is necessary to generate the powerful cone driving force. The vivid and dynamic sound is realized by driving a low impedance voice coil of only 2 ohms by the high current, in the strong magnetic field created by the newly designed magnetic circuit.

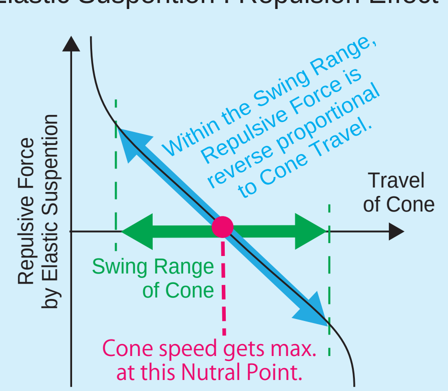
Small Full Range Speaker

Cone Travel and Swing Speed at the Nutral Point



HSP01

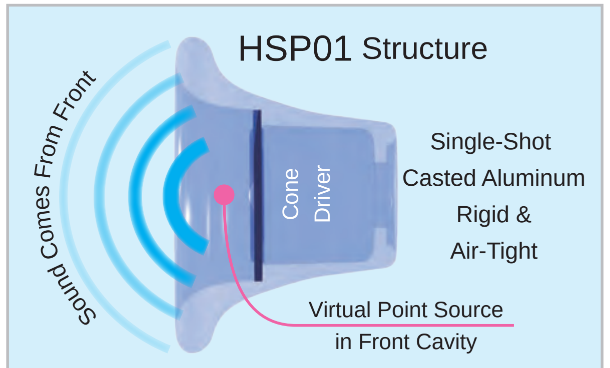
Elastic Suspention : Repulsion Effect



Enclosure Design of Clear Focus Speaker

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

HSP01 Structure



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

HSP01's enclosure shape is designed to have a Virtual Point Source in the cavity in front of a Cone Unit. Audiences feel that all the sound is radiated from that one point.

It is formed as one-shot monolithic casting of Aluminum, to make it highly rigid and air-tight, and its cavity shape is designed to minimize the cavity space noise. The 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 hermetic seal to shut off the leakage. This unit is highly responsive, covering full music bandwidth of 30~30 kHz. To enhance the strength of deep bass sound, larger format with 5-inch cone is also provided as **HSP01-L** model.

HSP01 Product Line

- One-shot casted Aluminum enclosure is air-tight, shaped as free from harmful resonances and vibrations. The silence and the dynamism of sound is realized by this new common design concept.
- There are three types of product, -S, -M and -L, to satisfy audiophiles' taste about various music sources.
- HDSA01 is installed with the preset parameter library of the signal processing, optimized for each type.
- The 3.5-inch small cone driver of -S and -M is light weight and highly rigid. It plays back the full music bandwidth, and is controlled optimally to give the precise large scale spacial expression.
- A larger format model Type-L is more focussed on the expression of grand bass instruments. It has a 5" cone driver matching with the sturdy enclosure. It is fundamentally different from the conventional speakers with a large size woofer that adds un-natural bass sounds without articulation.

HDSA01 Phase Control System Amp

HDSA01 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 HSP01, it realizes the next generation audio system, realizing
the truely authentic sound to the original input waveform.

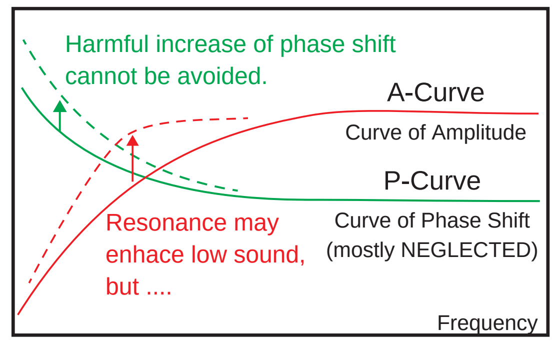
WANTED "True Bass Sound" Overcoming the Physical Limit of Speaker Unit

Generally speaking, the sound of dynamic speaker gets weaker toward the lower frequency, and gets more phase shift. This tendency is inevitable because of the physical limitations.

To overcome this limitations, various counter measures have been taken, such as enlarging the speaker cabinet. These days, lots of smaller models are announced also boasting the bass enhancement, but by using the resonance.

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

Consequently, though the resonance provides more bass volume, but it contaminates the waveform, and the result sound gets blurred and loses the impact.

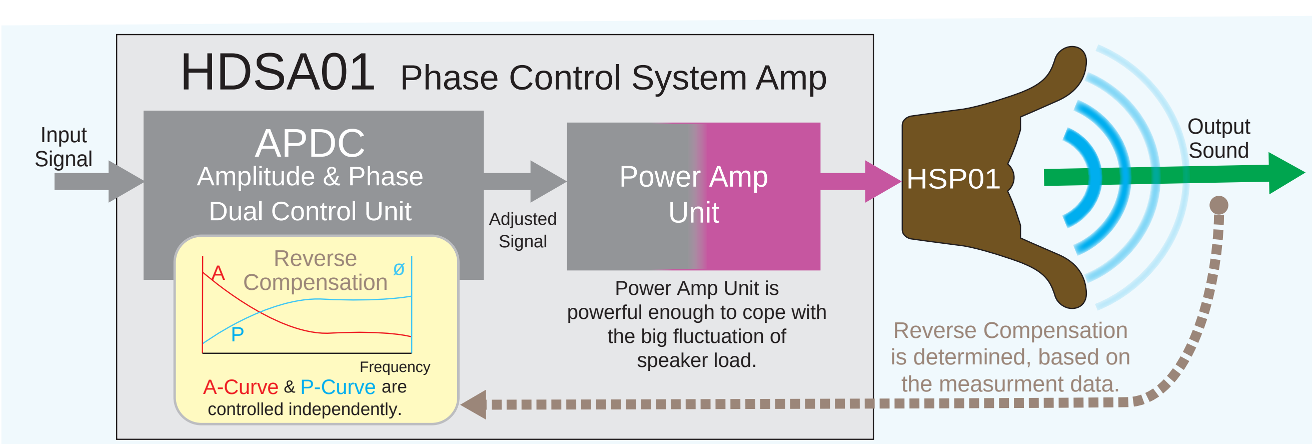


Bass enhancement by resonance
is accompanied by harmful phase shift.

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

In order to preserve the waveform, the Frequency Analysis Theory demands that both of 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 about the phase shift control, is impractically strict for the analog technology that depends on the physical phenomena to express the information. The phase shift control is about handling the time that is not altered arbitrarily by the analog technology. The only solution for this phase problem is in the digital signal processing technology.

Anyway, the waveform preservation is not an easy problem. It needs deep understanding on both analog and digital technologies, backed up by the skilful implimentation. Haniwa's APDC (Amplitude & Phase Dual Control) technology is the result of our efforts, that controls A-Curve and P-Curve independently, and is implemented in HDSA01 "Phase Control System Amp". This unique system amp is the core of Haniwa REAL 3D AUDIO.



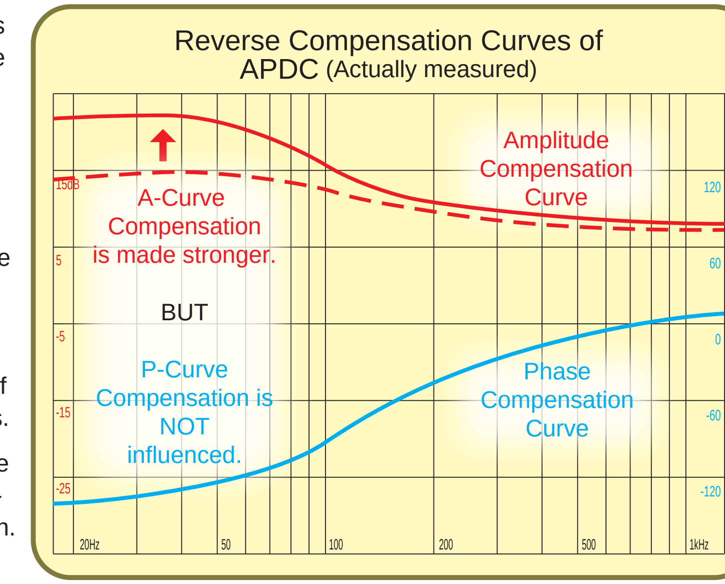
APDC Amplitude and Phase Dual Control

The Figure on the right shows the A- and P- Compensation curves used in APDC, to compensate HSP01's sound output. It should be noted that the P-Compensation curve is not influenced while A-Compensation curve is made stronger. It has been impossible with the analog technologies to control A- and P-Curves mutually independent.

There are, so called, equalizers that control the frequency response intending to improve the sound quality. Most of them are based on the analog technology, and recently Digital Graphic Equalizers are also announced. However, they control only A-Curve, and the induced change in P-Curve is neglected, even though the degradation of P-Curve seriouly damages the quality of sound. Eventually, many users tend to switch off those equalizers.

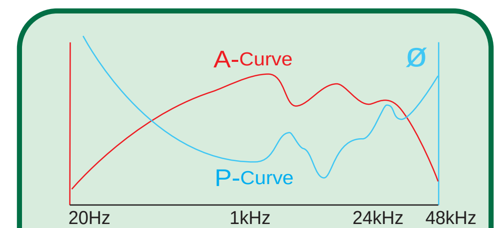
To the contrary, APDC is totally different and revolutional, because it utilizes the digital technology to its limit for realizing the independent control of amplitude profile from the phase profile modification.

It opens the possibility of more straightforward audio technology.

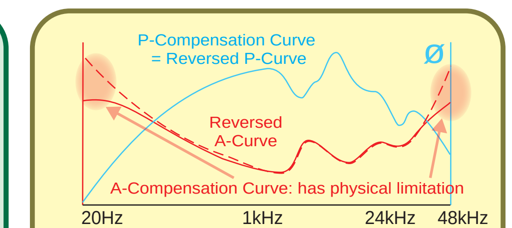


APDC Effect Shown in Frequency Domain

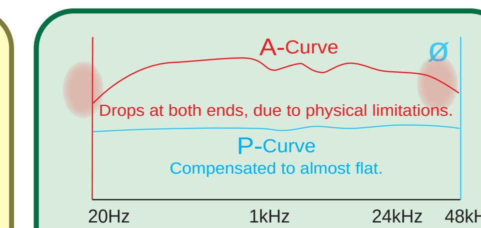
APDC (Amplitude and Phase Dual Control) uses DSP (Digital Signal Processor) to keep the input waveform in the output sound.



Frequency Response of
HSP01 **without APDC**



Reverse Compensation Curve
of APDC



HDSA01 + HSP01
Total Frequency Response with APDC

HSP01 by itself shows the typical single cone speaker characteristics. Its frequency response of gain (A-Curve) drops gradually below 1kHz. To the contrary, its phase lag characteristics (P-Curve) increases significantly toward the lower frequency.

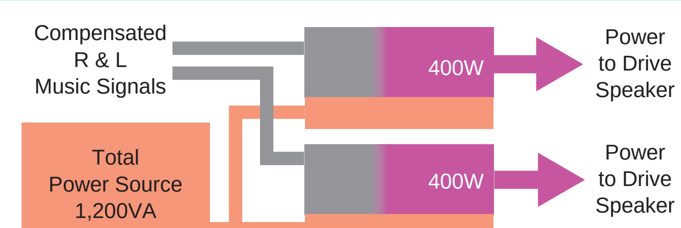
Although its mid to high sound is smooth and charming, the volume of bass sound cannot but be insufficient, due to the physical limitation of small full range unit.

Theoretically, A- and P-Curve can be compensated to be flat by applying the inversed A- and P-Curve for the reverse compensation. This raises the problem of deep compensation at lower than 100Hz.

This deep bass enhancement is necessary to keep the dynamism of bass instruments performances. So, the power amp should provide sufficient current, and speaker system should have rigid enclosure and unit to cope with this deep compensation.

By pre-adjusting the music signal with ADPC to compensate the HSP01's physical characteristics, the frequency response of the total system is made close to be flat toward low frequency. This is impossible with conventional speaker technologies that depend on the physical phenomena without close control of phase shift. The use of digital technology is mandatory to control the phase shift.

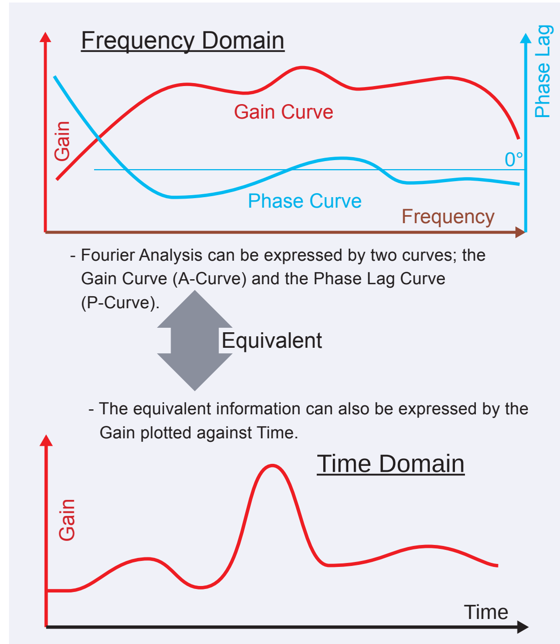
Power Amp Unit Highly Responsive to the Extreme Fluctuation of Speaker-Load



Two channels of 400W high power amp are suported by an ample power source of 1,200VA only found in huge systems. The output impedance of 400W power amp is low enough to match HSP01 with very low inpedance speaker unit, and large varience of speaker load. This power amp performance is indispensable for reproduction of powerful bass instruments.

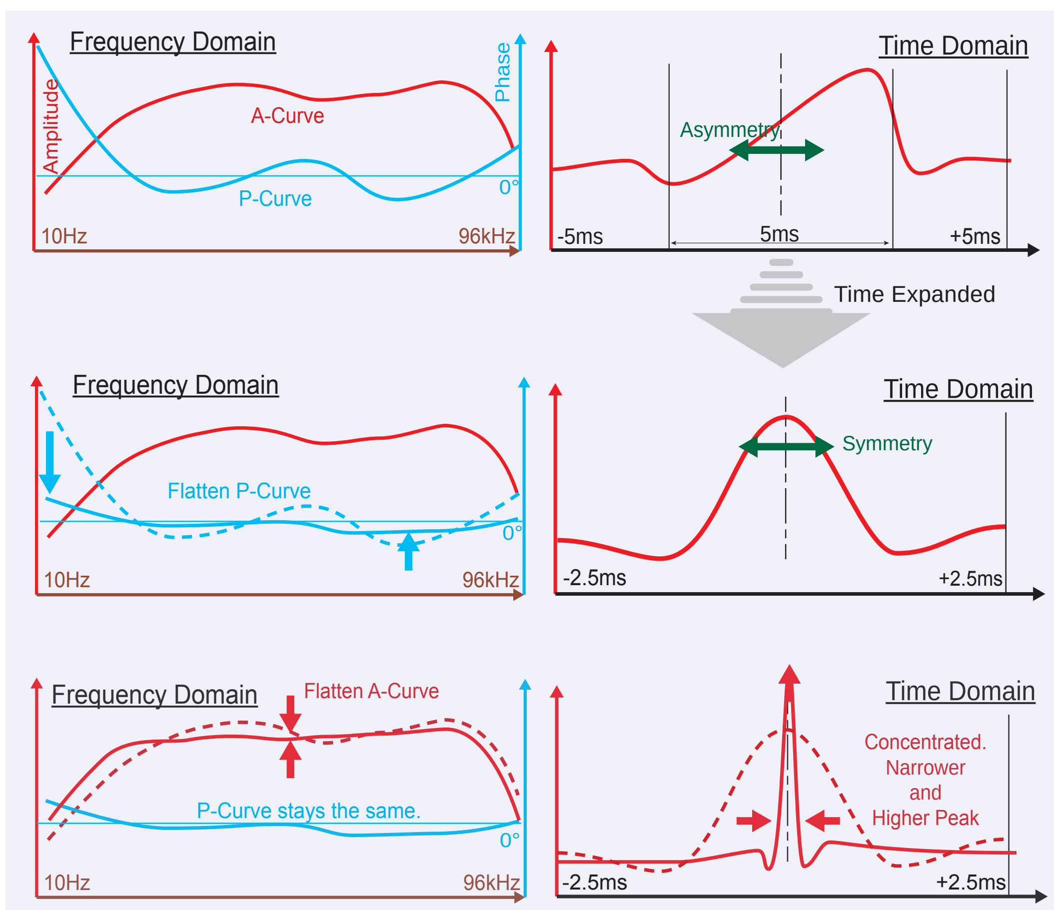
6. Relationship between Frequency Domain and Time Domain

- Fourier Analysis is a useful tool to understand the system behavior. By observing the response to the input, various characteristics of the system are obtained.
- There are two types of expression about the analysis data : the Frequency Domain and the Time Domain. The Frequency Domain data is plotted against the frequency, and the Time Domain data is plotted against time.
- These two ways of expression show the result of the same Fourier Analysis.
- To show the equivalent amount of information in these two domains, the gain and the timing characteristics are necessary.
- In the Frequency Domain, the gain is expressed by Gain Curve, and the timing is expressed by Phase Curve plotted against frequency.
- These two curves together, they carry the equivalent amount of information to the gain curve in the Time Domain.
- In the audio industry, Gain Curve is measured and published as “the Frequency Response”. But the Phase Curve, about timing characteristics, is not usually shown. Systems are evaluated only by Gain Curve, and the “waveform”, that carries important music information along the time, is neglected.
- ***Haniwa Audio recognized the importance of Phase Characteristics from the very beginning, and has established the technology to control Gain Curve and Phase Curve mutually independent. Based on this radical technology, new generation products are successfuly launched to the market.***



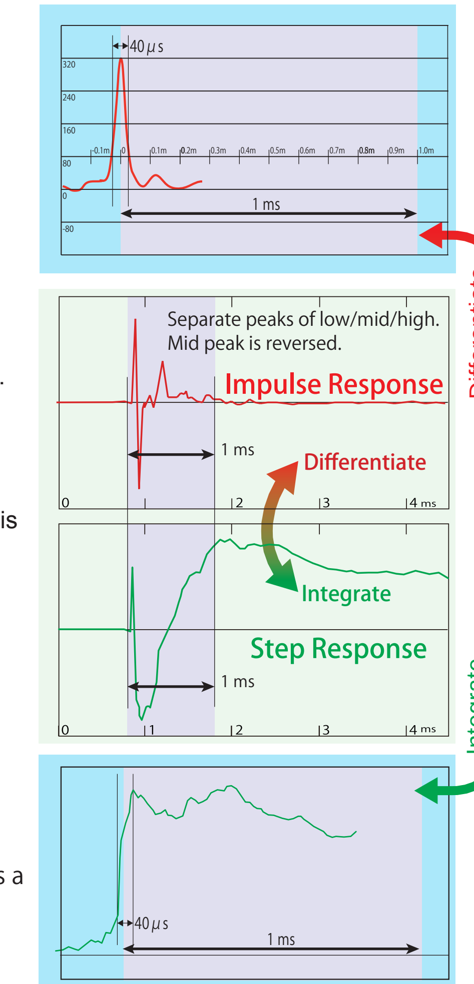
Improvement Process : Example of Audio System Development

- In the audio industry, the gain is called Amplitude.
- The music information is spread over 10Hz ~ 30kHz. To obtain precise information, HANIWA measures at 192kHz sampling, and analyzes up to 96kHz. In Time Domain, 10ms region is closely observed.
- APDC (Amplitude & Phase Dual Control) is the method to **flatten both of A- and P-Curves**.
- The followings shows how the Time Domain expression is changed by improving Curves in Frequency Domain.
- First of all, P-Curve is flattened by ADPC, then in Time Domain, all the response of full bandwidth appears at the same timing, and the Curve shape becomes symmetrical along time.
- Then, as bumps of A-Curve are decreased, the response gets concentrated.
- As A- and P-Curves are flattened, response of all frequency cancels each other to make the skirt part flat, leaving only one central peak.



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

- Figure on the right shows the HANIWA System performance, and in the Time Domain, the response is concentrated within ±20μsec range. This is 8 clock width at 192kHz sampling, that is the measurement limit. This level of sharpness of response has not been achieved.
- This situation is called “**The impulse responce in the Time Domain is close to 1.**”



High-End Multiway System

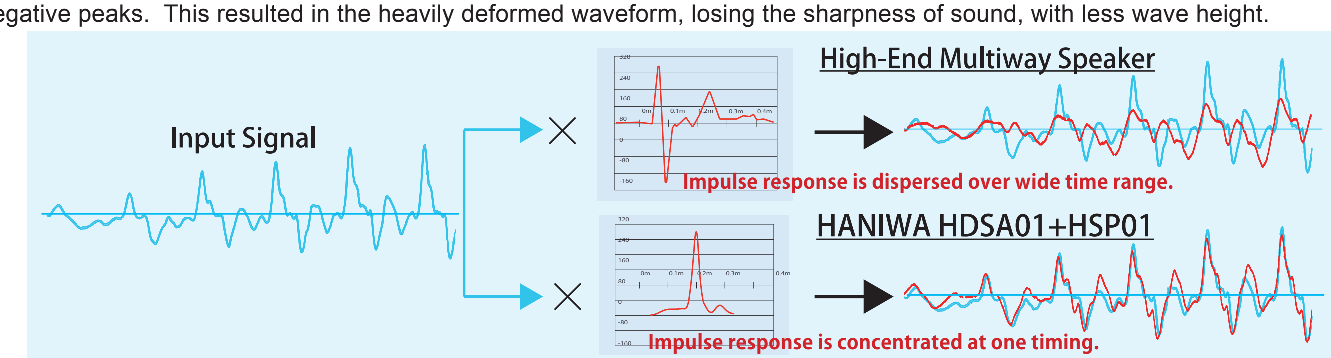
- Step response of aformentioned high-end system is published as the time domain data.
- Impulse response is suitable for observing mid to high range behavior, but not for low range observation. The low range response is distributed over wide range of time.
- The integral of impulse response is the step response, and vise 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 outstanding.
- High, mid and low range peaks are appearing multiple times and with the positive and negative polarity.

HANIWA HDSA01 + HSP01

- Step response of HSP01 sound is shown on the right.
- It rises up within very short time of 40 μ sec , and then falls down slowly.
- As shown above, its impulse response is rising and falling within 20 μ sec each, and has a single and sharp peak.
- This results in the clean step shape response, without the separation of high, mid and low ranges.

Relation between Impulse Response and Output Waveform

- Impulse response shows how the input signal is affected by the system.
- The 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 the sound by the system with the impulse response. If the impulse response is close to 1, the output sound waveform gets closer to the that of input signal, and gets almost identical.
- The actual waveform comparison is shown below, using actually measured waveform data.
- Conventional high-end multiway speaker has the impulse response wpread over the wide timing range, also with positive and negative peaks. This resulted in the heavily deformed waveform, losing the sharpness of sound, with less 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 virtuoso's performances are clearly appreciated with their spatial positioning.
- That is the essense of **REAL 3D Audio**, bringing the reality one step higher than the conventional systems.

Haniwa
Integrated REAL 3D AUDIO SYSTEM

This product aims at the most authentic sound reproduction from the input music signals.
For that end, we optimized it as one integrated audio system, with the small speaker system that shuts out any noise other than the input signal, and the system amp utilizing the highest digital technologies.

HDSA01 Digital Phase Control System Amp

HDSA01 (Digital Phase Control System Amp) drives HSP01 (Small Full Range Speakers) with ample power, after precisely and independently compensating both gain and phase characteristics, in 192kHz/24bit format. This is the standard data format of this versatile system amp to which variety of input signals are converted.

Input	**Analog**	RCA : 2ch (stereo) ±20dB Variable Gain
	Digital	Optical : 2ch / Coax : 2ch / AES : 1ch
Output	**Analog**	RCA : 1ch (stereo) for external power amp
		Speaker Driver : 400W (at 4Ω)
USB Port		USB 2.0 (Bi-directional Connection with PC)
APDC Unit		Independent Control of Gain and Phase Curves
Dimension (WHD)		316 x 95 x 370mm
Weight		5.5 kg

HSP01 Full Range Clear Focus Speakers

HSP01 is a family of speakers that features the followings.

- It has small Aluminum mono-casted structure that is small, perfectly air-tight, and free from the resonance of any kind.
- The shape of the part releasing the sound is designed to form a virtual point source of sound that keeps the sound focus sharpest to eliminate the blurring of sound presentation.
- The speaker unit is newly developed with specificaitons beyond the common sense. They emit the high audio energy with sufficient response speed. These units are driven by HDSA01 accurately and powerfully, from the delicate pianissimo to the grand fortessimo.

These features realize the amazingly realistic and impactful sound.
There are three variations of models with following characteristics.

- Type M: The general purpose model for most of the uses.
- Type S: Acurate and precise 3D space presentation provides the charm of musical conversation among players.
- Type L: Suitable for the grand music built on the power of base instruments.

	Type S	Type M	Type L
Full Range Unit	3.5” Aluminum Cone	3.5” Aluminum Cone	5.0” Aluminum Cone
Band Width	35Hz - 32kHz (-10dB)	30Hz - 30kHz (-10dB)	28Hz - 26kHz (-10dB)
Impedance	1.5Ω at 100Hz	1.5Ω at 100Hz	1.7Ω at 100Hz
Inductance	2.0mH at 100Hz	2.0mH at 100Hz	2.0mH at 100Hz
Dimension (WHD)	400 X 1,140 X 400 mm	450 X 1,045 X 450 mm	450 X 1,060 X 450 mm
Hone Size (Φ, D)	240 X 167 mm	270 X 202 mm	290 X 210 mm
Weight	9.5kg	12.0 kg	14.5 kg

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