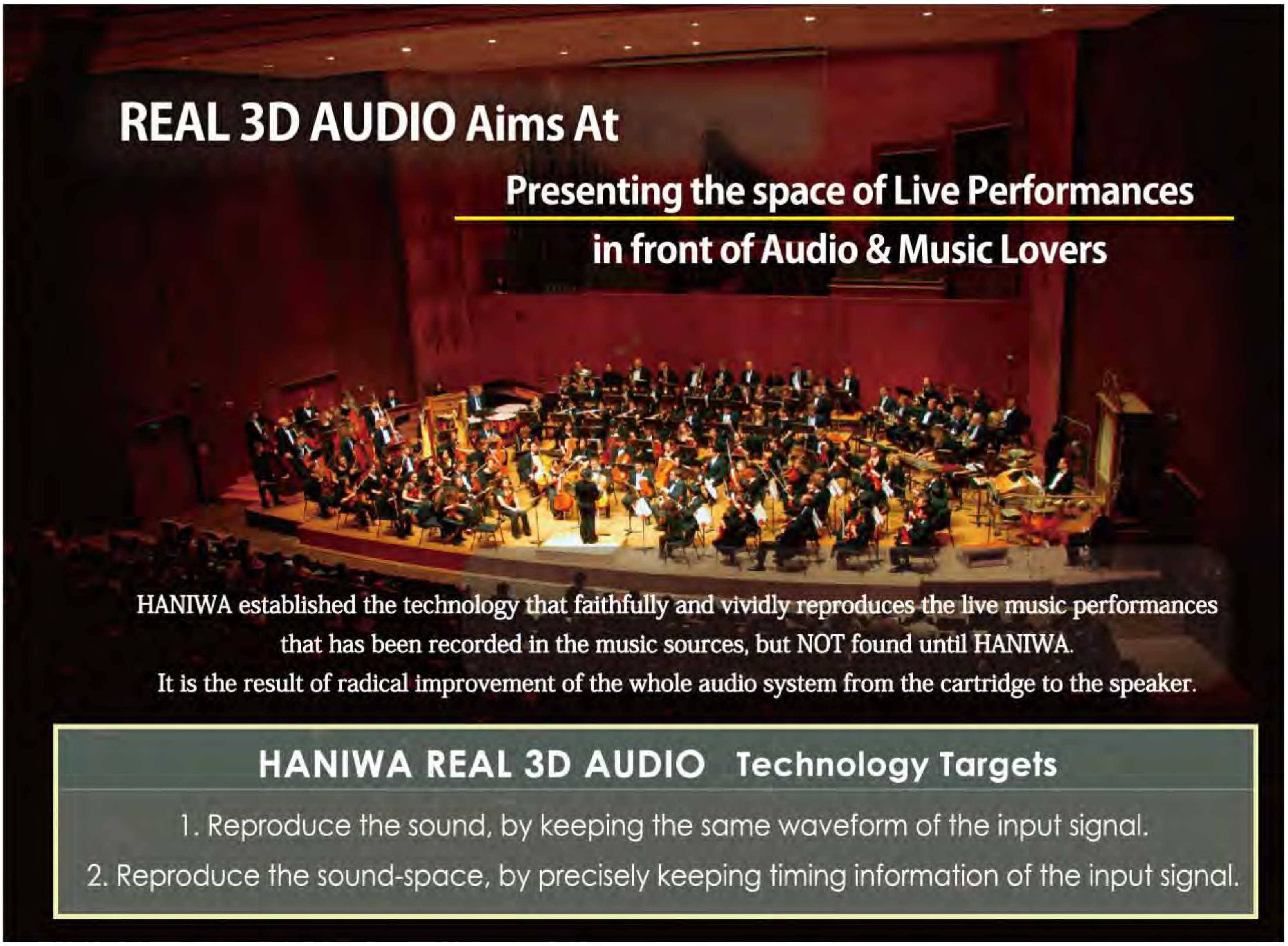

# FULL INTEGRATED REAL 3D AUDIO SYSTEM

HANIWA Audio



# REAL 3D AUDIO Aims At

## Presenting the space of Live Performances in front of Audio & Music Lovers

HANIWA established the technology that faithfully and vividly reproduces the live music performances that has been recorded in the music sources, but NOT found until HANIWA.
It is the result of radical improvement of the whole audio system from the cartridge to the speaker.

### HANIWA REAL 3D AUDIO Technology Targets

1. Reproduce the sound, by keeping the same waveform of the input signal.
2. Reproduce the sound-space, by precisely keeping timing information of the input signal.

## 1. Why NOT the Waveform?

"Audio System reproduces the music as it is recorded." This statement sounds clear, simple and practical. However, the current audio industry seems not to face this fundamental quest. Many of supplyers claim "high fidelity to original sound." But in reality, they want to sell their "own Good Sound" that is quite subjective. They chase after easier technology targets, based on misled technology theories. The objective and doubtlessly clear target of "preservation of the waveform" has been left untouched.

It is widely accepeted to use "Fourier Analysis" for the evaluation of audio equipments, though only the amplitude curve is used for ease of understainding. It is a part of common sense that "Flat gain curve of Fourier Analysis means the equipment keeps high fidelity to the input signal." However, this statement tells ONLY A HALF of the truth of fidelity. Because, if the timing for each frequency is not preserved, the waveform becomes quite different and of course the sound becomes different. Simply suppose that the low frequency portion of bass instruments arrives late to the higher frequency portion. You may not be able to accept this "late bass" to represent the sound of contrabass. It is also the truth of mathematics that in order to preserve waveform, both amplitude and phase curves should be flat. Then, why the flattening of a phase curve has not been taken as a serious technology agenda in Audio Industry?

A simple answer to this question is "It is prohibitively difficult to fflatten the phase curve." The conventional audio systems have been developed on the components that use "physical phenomena with which the phase (time) cannot be manipulated arbitrarily." Furthermore, these components are developed independently each other, using only amplitude curves for their mutual matching. For example, when a filter ciruit is designed to flatten the amplitude curve, that filter circuit itself changes the phase curve. Suppose that an audio system is composed of numerous components with individual phase disturbance, and you will instantly know the difficulty of handling this problem.

So, the industry simply could not but abandon to think about the preservation of waveform.

## Limitation of Analog Audio Technologies

"Direct use of physical phenomena to express the signal magnitude, and to convert the signal to the sound" is the technology used to realize electronic and electric circuits, speaker units, and enclosures. It depends on the natural physics, and is called ANALOG technology as a whole. Most of the contemporary audio equipments are based on this fundamental category of technology.

By the way, the physical phenomena is ruled by the non-stop TIME flow, that is uncontrollable by the physical phenomena by itself. It is also true that the physical amounts, such as voltage, current, resistance, capacitance, inductivity, etc., are off the rule of proportion as they go beyond some limit. For example, human audible frequency covers a wide range from few Hz to tens of kHz (wider variation than ten thousands times), and most of the physical phenomena cannot be proportional all through that wide range. This means that the signal range should be divided into several sub-ranges where the proportionality is assured in the practical allowance, and the results are summed back to handle the whole audible frequency range. That is, so many stages of analog processes are concatinated to realize the total audio system, through which a lot of distortions and time shifts are accumulated until the signal is finally converted to sound. No wonder why the waveform cannot but get confused.

The electronic circuit can handle the wide range of audible frequency rather easily, but for the process of "electric circuit" driving "speaker unit + enclosure," it is almost impossible to convert electricity to sound without changing the waveform. Actually, the distortion in this final process is the most evident among the whole audio process.

## Possibility of Digital Signal Processing Technology

On the other hand, the digital signal processing is discretely different from the analog technologies. As it represents the signal as the sequence of purely abstract numbers,

by the stable information storage, "the timing can be shifted" ,

by increasing the digits, large and small values can be handled with the same precision, and

all the process that is describable in mathematical formula can be executed without losing precision.

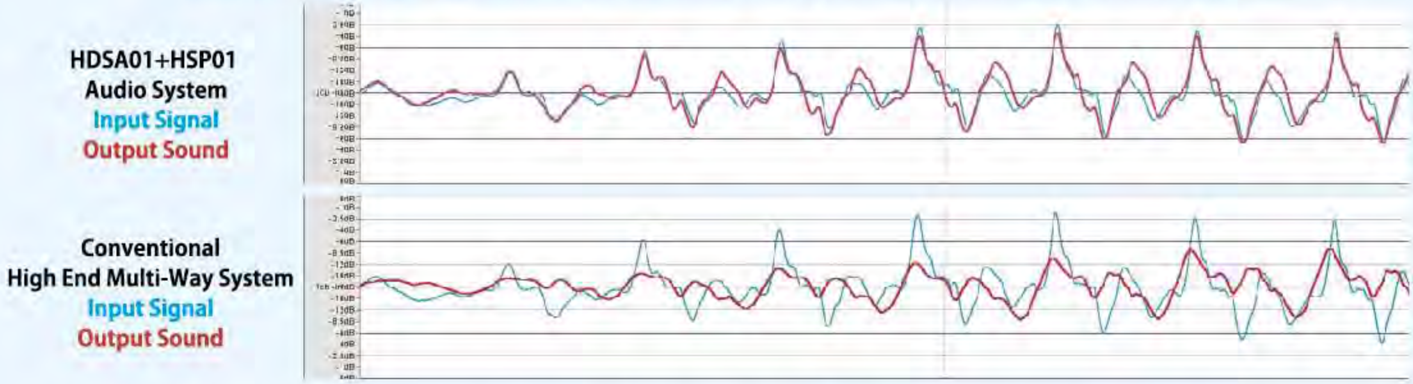
The limits are set by the memory capacity and processor speed, but the recent dramatic progress allows to process the signal of band width of 10Hz to 96kHz, through the huge storage. The information storage already has sufficient read/write speed and capacity, and the signal processor has sufficient speed, too. The progress of DSP (Digital Signal Processore) technology has made the timing-shift operation practical.

## HANIWA Technology, Today

### The Waveform by HANIWA : Total Solution of Analog & Digital Technologies

### HDSA01+HSP01 Audio System vs Conventional High End Multi-Way System

Blue Curve = Input Signal Red Curve = Output Sound (Playing : "Art Pepper meets The Rhythm Section")

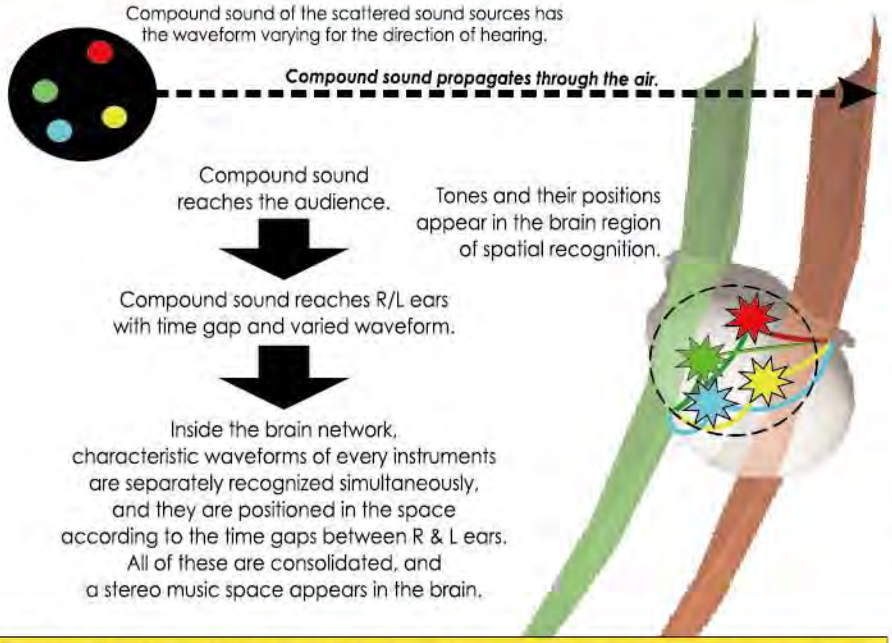



- Two graphs above illustrate the output sound waveform of two systems, playing "Art Pepper meets The Rhythm Section."
- The first graph shows HANIWA product, and the second shows the conventional high-end system. Their output are shown over the same input signal.
- Let' s see how the output sound waveforms are keeping the waveform of input signal.
- The common input signal has the waveform with sharp upward peaks, that appears typically at the start of plosive of the brass and saxophone. HANIWA system is preserving this feature well, keeping the timing and the height of peaks. To the contrary, the sound of conventional system loses the sharpness of peaks, and the waveform gets closer to symmetric up and down, losing the feature of saxophone. It starts slowly and the trangent is not explosive . Eventually, the sound gets dull and misses the articulation.
- This experiment proves that HANIWA keeps the input waveform far better than the conventional product, and HANIWA provides the charm of original sound well.

## 2. Reproducing the Music Space

When we recognize 3D space by hearing, the sound arrival timing gap between two ears plays the important role. When the music is played in a hall, various musical instruments are laid out on the stage, and if this delicate time gap is not precisely reproduced, the 3D space, including the hall, instruments and players, gets confused, and the reality of live atmosphere gets lost.

### The Process of 3D space Recognition by Hearing




The technology to accurately transfer waveform of each instrument is absolutely essential to present the music space with reality.

Hearing is an essential tool of survival for most of animals. It is necessary for them to avoid danger and to gain foods. It has been playing the important role of identifying calamities, or positioning enemies and games.

In order to reproduce the lively music performance, we must develop audio systems based on the deep understanding about this nature of hearing.

By the way, what is happening inside our brain when we are enjoying the live music?

The sound of instruments and the voice of singers arrive at us having small time gap between left and right ears. For example, if an instrument is played on the side closer to left ear, that sound arrives at left ear a bit earlier than at right ear.

As the hearing stimulus spreads over the brain network, multiple instruments with their own tones are identified real time. At the same time, right/left time gaps are also recognized for all the identified insruments. In this process, the delicate tone features of each sound source are extracted from the total compound sound, and the positions of each sound source is judged instantly.

Inside the spatial recognition region of the brain, all of these information are consolidated into the total image of the performance.
As a whole, the virtual experience of attending the concert is created in the brain, together with the sense of space about the hall.

### Only the sound with preserved waveform can induce the natural feeling of the whole detail of live performances.

By understanding the mechanism of hearing, it becomes evident that audio systems must transmit correct sound waveform. It should be noted that by preserving the features of waveform accurately, and by keeping the timing information correct, the sound becomes lively and real. The delicate touch of piano and violin, and the powerful impact of brass and percussion, these can be revived only by transmitting the accurate and precise waveform.

***The audio system should not add nor make up sound, other than the original. HANIWA insists on this statement.***

## 3. HANIWA Solution

### The accurate spatial expression can be realized only with a full range single unit speaker.

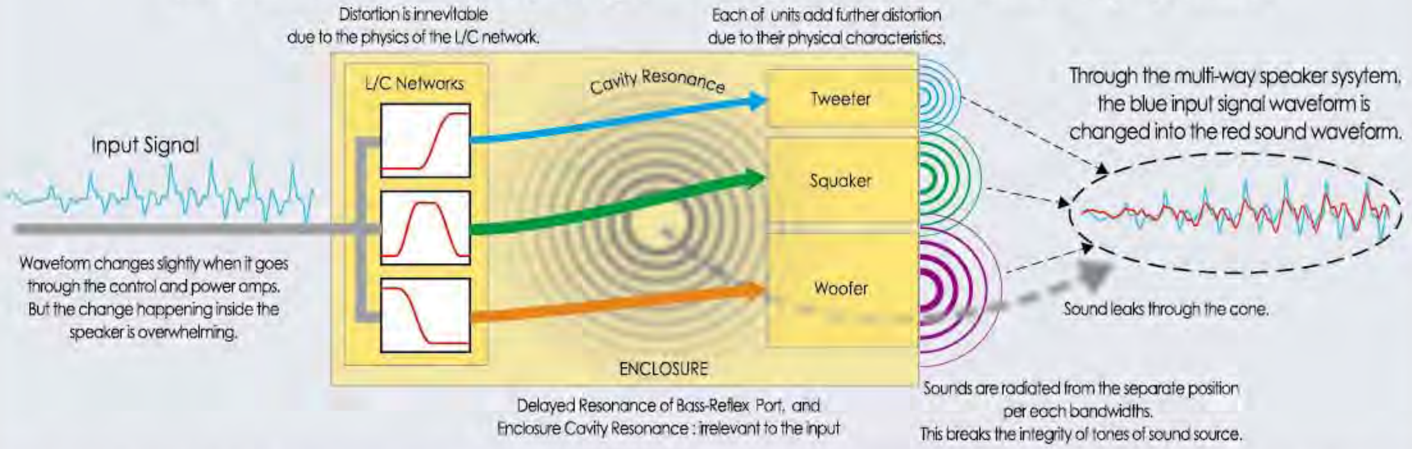
We have conducted the long and wide research about the best speaker configuration for reproducing waveform accurately, and for expressing the music space with reality. The conclusions are that the full-range single unit speaker is the best for this purpose, and the multi-way speaker is not appropriate especially for the spatial presentation.

Simple thoughts may conlude that the multi-way speaker is more expressive and powerful. It is widely believed that for the purpose of keeping the sound fidelity, it is necessary to separate the music bandwidth into three pieces of high, mid and low ranges, and the amplitude curves in each ranges should be adjusted flat. However, this solution is handling only the amplitude curves, and neglecting the disturbances in phase curves that causes the serious deformation of waveform along time axis.

The figure below shows how the waveform is deformed while the music signal is transferred to the output sound through a multi-way speaker system. The music signal passes through control amp and power amp without significant deformation of waveform. When it goes through L/C network inside the speaker, it is divided into multiple frequency ranges. At this point, signal waveforms of each frequency range are distorted due to the phase characteristics of L/C network. On top of these distorted waveform, more serious distortion by speaker units, tweeter, squaker and woofer, are added, and radiated from speaker units with timing lags and positional displacements. It should also be noted that an unnecessarily large enclosure itself is also the source of noise that is irrelevant to the input signal, but peculiar to the cabinet design, such as the resonance and standing wave of internal cavity, or the vibration of enclosure cabinet.

Consequently, the output sound gets highly contaminated, and the music space information cannot be presented to the listener.

### How the waveform of input signal changes through a multi-way speaker?

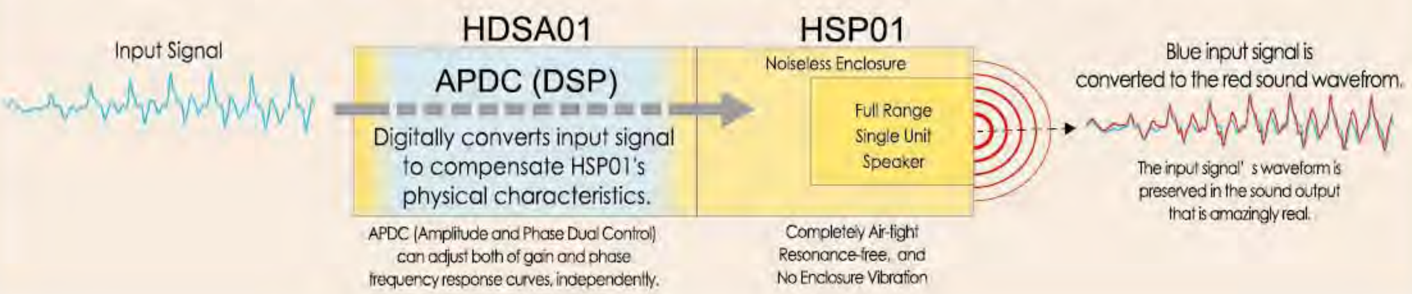



In case of HANIWA, the the simplest full range single unit speaker HSP01 is driven by HDSA01, that is equipped with the technology compensating HSP01's low frequency weakness and restoring the original input signal waveform.

HDSA01 (Digital Phase Control System Amp) is equipped with the APDC (Amplitude and Phase Dual Control) unit that utilizes the power of the latest DSP (Digital Signal Processor) to its limit. APDC adjusts the input signal so that the original input signal waveform is restored in the output sound. The adjustment is done based on the parameters actually measured beforehand on HSP01.

As whole design is intended to preserve the input waveform in the output sound, HANIWA system presents the cleanly focussed space image to the listener, and the original music space is authentically reproduced in his mind.

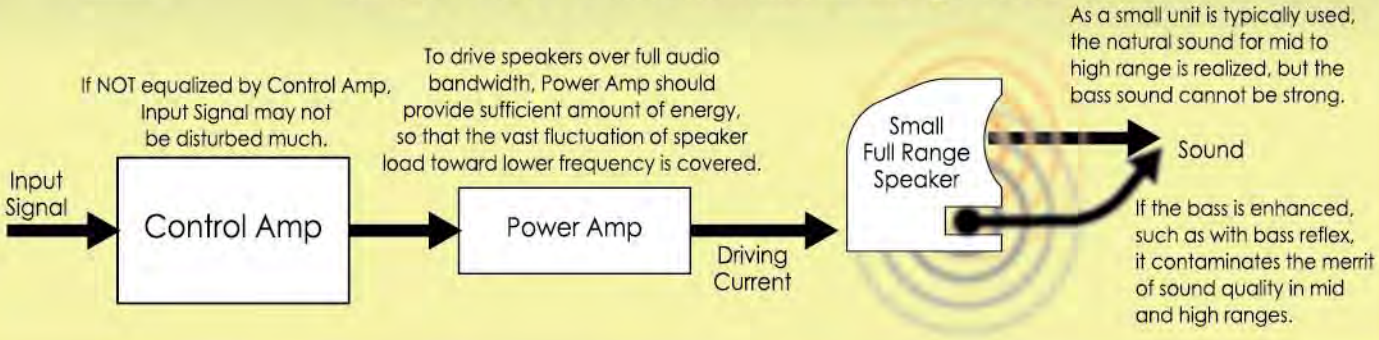
### How "HDSA01+HSP01" system transfers the input waveform to output sound?



# 4. Full Range, Clear Focus Speaker

# HSP01

## Weak Point of Conventional Full Range Audio System




## Elastic Suspension Technology

Even a small speaker unit can reproduce the punchy performance of bass instruments.

**"In order to output solid bass sound, a woofer diameter should be large."** This is a common statement about low sound reproduciton. However, there is some misunderstanding about why the big woofer can output low sound.

To begin with, the sound is the wave of cyclic change of air pressure, and that pressure is generated by the speed of reciprocal motion of a cone. The larger size of speaker cone usually means the longer stroke of cone motion. Eventually, even at low frequency, cone moves at sufficient speed to generate sufficient change of air pressure. This is how the low sound is generated by a large woofer, and it is not because of the cone diameter.

On the other hand, small full range speakers can reproduce the natural mid to high range sound. One of the reasons for this is that they have less divided vibration due to their small size. But at the same time, their bass sound cannot be but poor, because their stroke motion speed is not enough to generate the sufficient amount of pressure needed for strong low sound.

To overcome this defect of small full range speakers, HSP01 uses the unit that has double length of cone stroke so that the cone can move fast enough to generate pressure even at the low frequency. In addition to this, it uses the Elastic Suspention mechanism to accelerate the returning speed from the farthest end of the cone stroke. By these improvements, the bass sound is enhanced without the distortion of waveform.

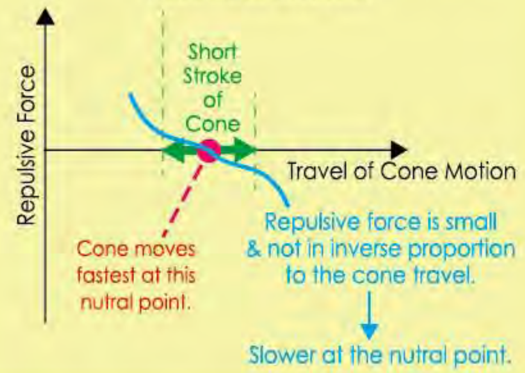
The voice coil of this unit is as large as 30mm diameter, with the impedance of less than 3.5-ohms. To drive this extremely designed unit, the power amp unit inside HDSA01 can provide up to 400W driving HSP01.

Consequently, the low frequency sound from this small speaker is not just low sound, but actually reproduces how the bass instruments are played.

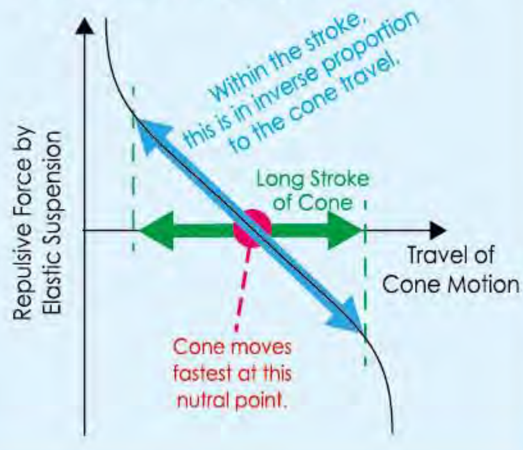
**Small Full Range Speaker**

Travel of cone motion and Repulsive Force

(Slope indicates Speed.)

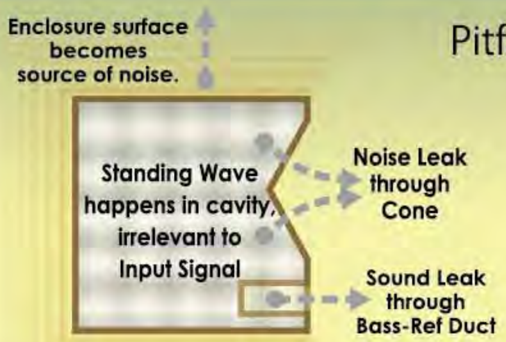



**HSP01 : Elastic Suspension Provides Strong Repulsive Force**



# Speaker of the Next Generation

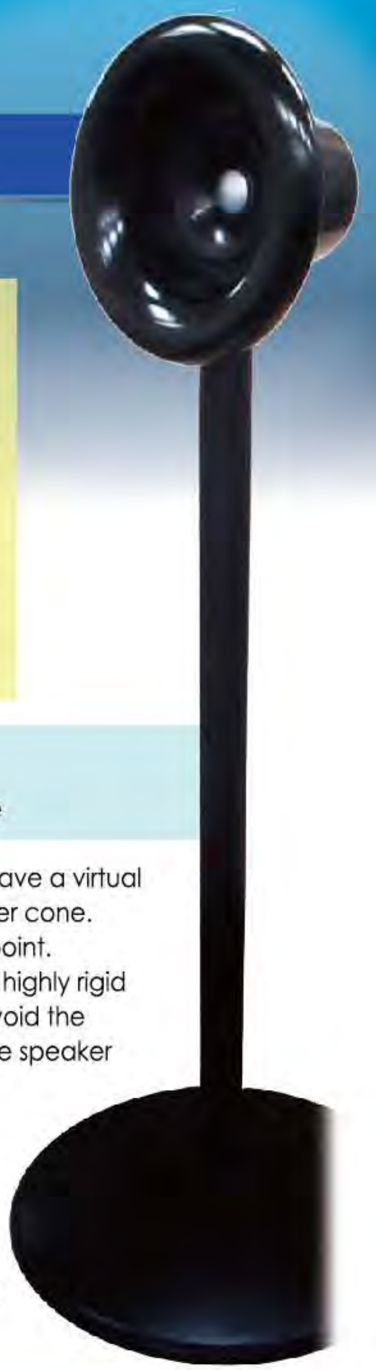
## Silent, Agile, Accurate and Powerful




### Pitfall of Enclosure Design

There is a type of noise comming from the speaker called **Box-Sound**, that is emitted from the surface of enclosure. When a speaker makes sound, enclosre itself vibrates and it becomes the source of noise. To prevent this noise, usually the enclosure is made stiff and still.

However, the cavity of the enclosure itself generates standing waves that is irrelevant to the input signal, and that is radiated forward through the cone. This noise is not negligible, but is not paid necessary attention.



## Enclosure Design of Clear Focus Speaker

### Mono-Cast Structure is Rigid and Air-Tight / Shutting Down Noise & Resonance


As full bandwidth sound comes from one virtual point source, the focuses of L/R speakers become clear, and the sound becomes sharp and shows vivid spatial expression.

The shape of HSP01 enclosure is designed to have a virtual point source in the cavity in front of the speaker cone. All the sound virtually comes from that single point. Its structure is the aluminum mono-cast, and is highly rigid and air-tight. Its internal cavity is shaped to avoid the standing wave. Any sound other than from the speaker cone is supressed.

The speaker unit has highly rigid Aluminum cone. It is mounted with tight seal, to shut down the sound leakage through the cone. This unit has ultra-high response speed and can endure the deep bass boost, to cover full range from 20Hz to 30kHz by itself.

## Features of HSP01

The enclosure is Aluminum casted as one piece, and is perfectly air-tight. It is shaped to supress resonance and vibration. This enclosure contributes to realize one step higher level of silence and dynamism of music scene reproduction.

The enclosure is held at appropriate height on top of the pole that has strong vibration absorbers. So, it is almost like a pure sound source is made floating in the air.

The small cone driver of 3.5" diameter is light but highly rigid, and reproduces the full audio bandwidth without crooked characteristics. So, it can be controlled precisely by HDSA01 to present the large scale spatial expression.

The concept of "virtual point source" positions sound sources accurately in the 3D audio space, without blur.

In order to express the sound of bass instruments without losing its impact, "Elastic Suspension" technology is applied. This is radically different from the conventional "bass enhancement with large woofer" that simply adds the amount of bass at wrong timing.

# 5. HDSA01 Phase Control System Amp

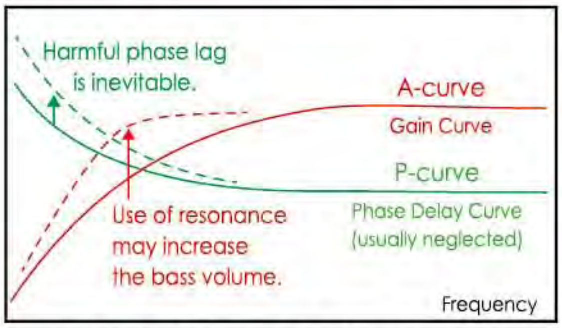
**HDSA01 is the core element of Hanawa REAL 3D AUDIO system, that unifies and controls the Haniwa Audio system, from the beginning to the end of play back.**
**It provides the unique phase control function based on the deepest use of digital technology, enabling the precise compensation of the speaker characteristics.**
**The combination with HSP01 realizes the sound output having the same waveform of input signal. This achievement pioneers the new audio technology frontier.**

## WANTED : Lifelike Bass Sound Can we break through the "physical limit of speaker" ?

The sound of ordinary dynamic speakers gets weaker toward the lower frequency of bass bandwidth, and is accompanied by the the increasing phase delay. This is the physical restriction that cannot be avoided.

There are various idea proposed to compensate this attenuation of bass sound, such as the use of bigger speakers and enclosures. Lately, the use of resonance has become the standard remedy for this symptom, declaring the rich bass sound from the smaller speakers.

But the use of resonance directly means that the added bass sound cannot happen at the same time as the original sound. It is like a shadow that cannot but follow after the original sound. So, the increased amount of bass volume blurs the original sound, and the whole sound gets dull and much less vivid.

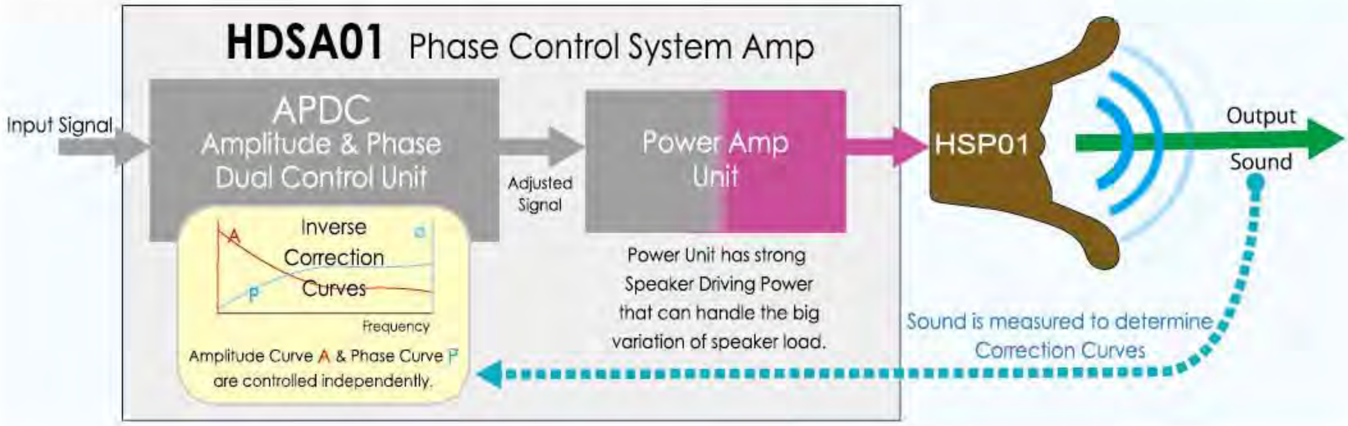



**Bass Enhancement by Resonance Adds Harmful Phase Lag**

## Waveform Reservation = Signal to Sound Conversion with FLAT A- and P- Curves

If the problem of "signal to sound conversion without the change of waveform" is stated from the method of Frequency Analysis as "the signal is transferred to the sound with its A- and P- curves are flat." This statement is simple, but is difficult to be realized by the analog technology, where the physical law rules. Only possibility is to use digital technology where the timing can be rearranged through the memory. However, there are practical problems to be solved, in order to actually develop products. The deepest understanding about analog and digital signal processing should be compiled, and the genius level of ideas and efforts for the implementation is necessary.

HANIWA has been working on this problem over a decade, and finally successfully developed the **APDC** (Amplitude & Phase Dual Control) technology, that can control A- and P-Curves independently. APDC is put it into the the product **HDSA01** (Phase Control System Amp) , as the significant first step of HANIWA REAL 3D AUDIO product line.



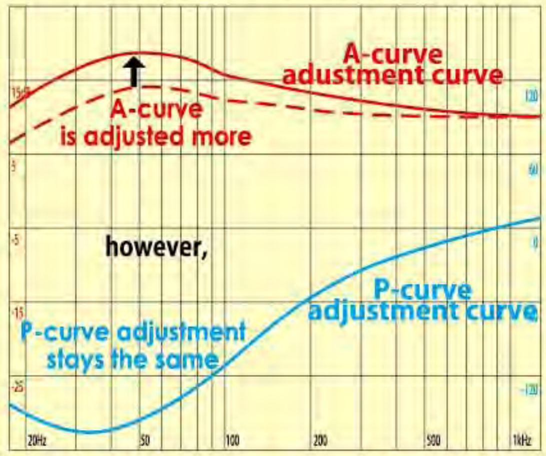
# APDC Amplitude and Phase Dual Control

The figure on the right shows the compensation curves used at HDSA01 to flatten A- and P-curves of HSP01 sound output. Note that the P-adjustment curve, that is to flatten P-curve of speaker output, stays the same even the adjustment of A-curve is enhanced. This independent control of A- and P-curves has not been realized before.

There is a technology called "equalizer" that controls the frequency characteristics to improve the sound quality. It has been developed with the analog technology, and recently, the "graphic equalizer" is introduced that uses digital technology. However, most of them controls only A-curve, and the deteriorated phase characteristics is left untouched. Though the amplitude is adjusted, the phase gets more confused, and many users get disappointed by the degraded sound, eventually stop using the equalizer.

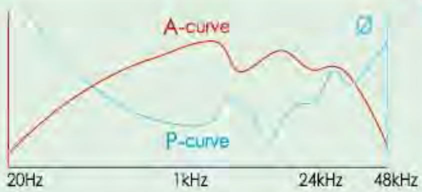
APDC is totally different from those conventional equalizer. The APDC is revolutionary as it is essential to keep the input waveform, but also as it enables the development of concise and honest audio system.

**Compensation Curves of APDC**
(measured in bass range)




## Improvement by APDC Amplitude & Phase Dual Control (shown in Frequency Domain)

APDC (Amplitude & Phase Dual Control) is the digital signal processing function, to make the waveform of output sound close to that of input signal. It is realized by intensive use of the DSP (Digital Signal Processor) capability.




HSP01 Frequency characteristics by itself
(No adjustment by APDC)


APDC Compensation Curves


HDSA01 + HSP01 Total Performances
(After adjusted by APDC)

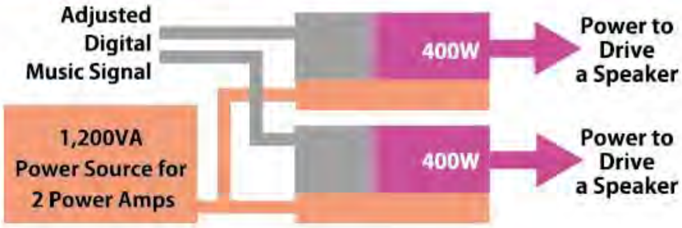
HSP01 has the characteristics of typical single cone speaker by itself. Its A-curve gradually falls in the bass bandwidth of lower than 1kHz. Its P-curve, on the other hand, rises rather quickly toward the bass end.

The small single cone speaker cannot but be physically restricted to have insufficient amount of bass sound, though it provides charming high quality sound in mid to high range.

In order to flatten A- and P-curves of a single cone speaker, the reverse curves may be applied for the compensation. However, it should be noted that the necessary boost in the bass sound requires to be extreme. This raises the agenda to be resolved on the speaker side, such as requirement for more toughness of whole structure and the agility of driving unit. It is also important that the power unit of the amplifier has enough amount of driving capacity. These technology elements should be coordinated well.

The characteristics of HSP01 are compensated by APDC with reverse curves, and both of A- and P-curves are extended flat toward the bass ends. This level of improvement has not been realised by the conventional speaker technologies, that have been designed within the limit of natural physics where the phase cannot be adjusted accurately. The outstanding feature of APDC is the "timing management" capability realized by the digital technology, and it enables the independent adjustment of A- and P-curves.

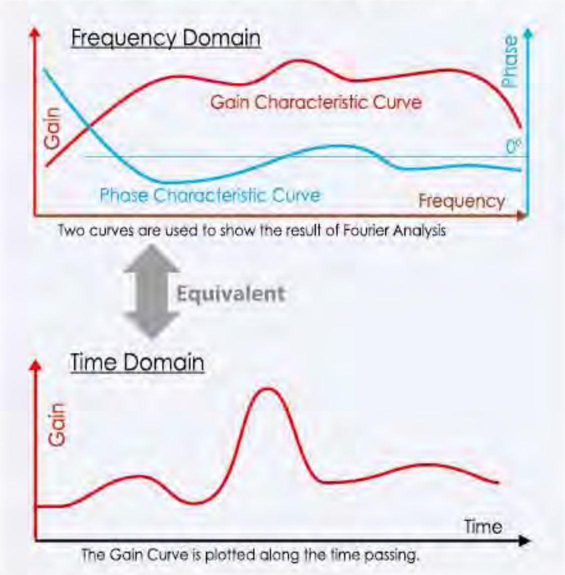
## Power Amp Unit Sufficient Power to Handle Wild Fluctuation of Speaker Load




1,200VA of big Power Source is equipped to supply for a pair of 400W Power Amps. The power amp units installed in HDSA01 are very compact, but can follow the wild load fluctuation of speakers that covers the full range of music. This is especially indispensable for the powerful presentation of bass instruments.

# 6. Frequency Domain & Time Domain

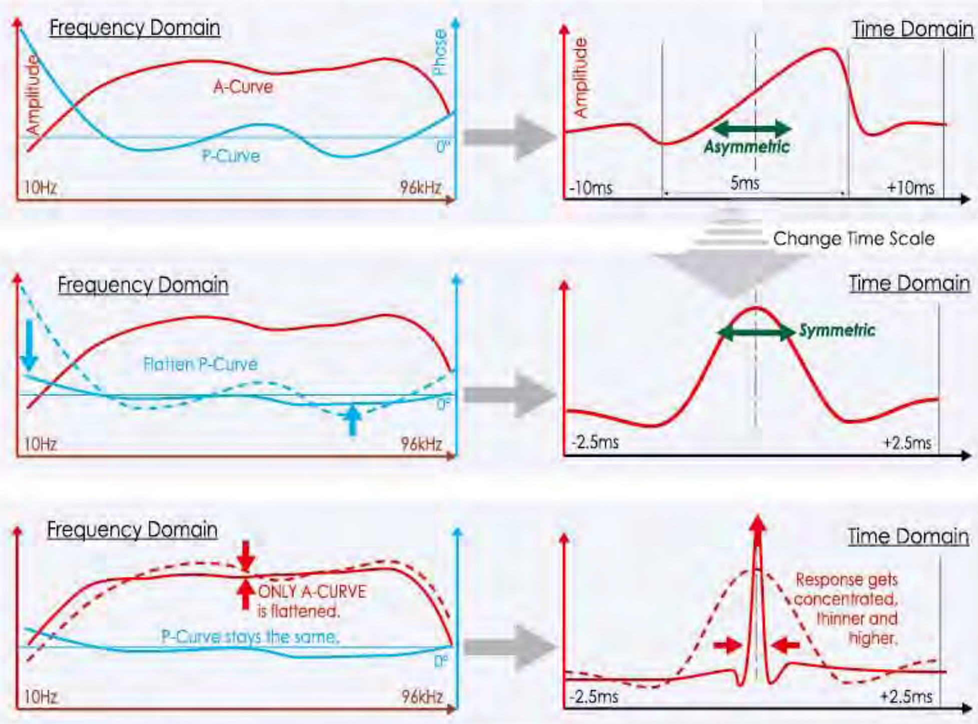
- Fourier Transform is a useful tool of signal analysis. By observing the Fourier response to the input, we can understand how the signal is processed. There are two formats of expressing the Fourier Analysis results. They are Frequency Domain and Time Domain. Information is plotted either against frequency or against time in each format. Difference of Domains is just the format of expressing the same Fourier Analysis.
- In order to express the equal information in these two domains, both of Gain and Timing information should be shown. In the Frequency Domain, Gain is shown by Gain Characteristic Curve, and the Timing is shown by Phase Curve. These two curves as a set, becomes equivalent to the Gain curve in the Time Domain.
- In the world of Audio System, the Gain Curve is commonly measured and publisized as the Frequency Characteristics, but the Phase Curve is not mentioned equally. In short, a system is evaluated only by the Gain-Frequency Curve, and the "waveform" (the most important music information including timing information) has been neglected.



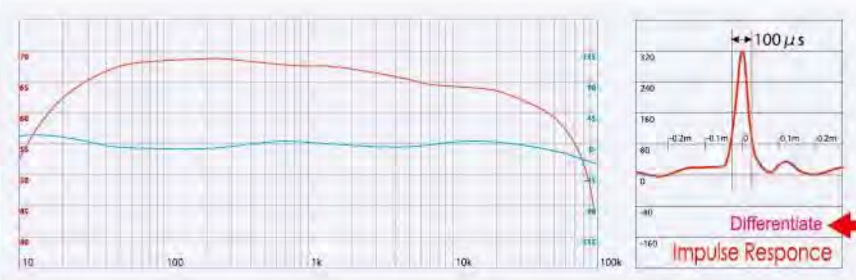

**HANIWA is developed with the equal respect for Gain and Phase Characteristics. Based on this new viewpoint, we established the way to control both of Gain and Phase characteristics mutually independent. HANIWA Audio provides the optimal system based on this technology breakthrough.**

## Actual Process of Improvement in case of Audio

- In the Audio world, the Gain and Phase characteristics are expressed in Amplitude Curve, and Phase Curve, respectively.
- The music information contains from 10Hz to a few tens of kHz. So, HANIWA development is based on the observation by using 192kHz sampling, and is analyzed with the high resolution, up to 96kHz. In Time Domain, observation is done in the time range of 10mSec on both side of a noticed timing.
- In Frequency Domain, A- and P- Curves are adjusted toward flat, by using APDC (Amplitude and Phase Dual Control) method.
- In the following, how the curve in Time Domain looks changed, as the two curves in Frequency Domain are modified.
- First of all, as the P-Curve is flattened in Frequency Domain, in the Time Domain, whole response over full range gets concentrated around the same timing, and the curve form becomes symmetric over the time.
- When the A-Curve is smoothed, the Time domain response gets more concentrated around the center.
- Finally, as both of A- and P- Curves are made almost flat, the waveform in Time Domain changes to show all frequency factors cancel each other around a highly concentrated peak.



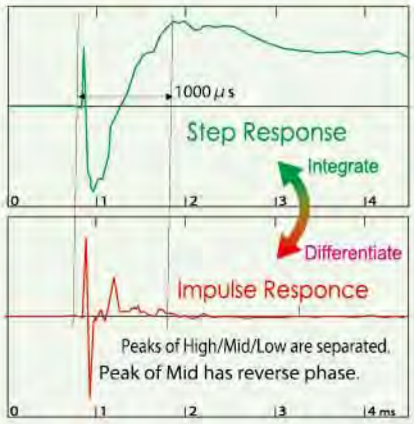
- The figure on the right shows the actually measured data of Haniwa System. The corresponding Time Domain Curve, on the far right, shows that the response is concentrated within ±50μsec range. This is equal to 20 clocks of 192kHz sampling, and is almot hitting the limit.
- This level of response is called ***"Impulse response is close to 1."***




# Comparison of HANIWA against a High-end Multi-way System, in Time Domain.

## High-end Multi-way System

- Generally speaking, the impulse response is suitable for observing mid to high range sound, but it is not necessarily good for low range observation, because the low range response is scattered wide in the impulse response.
- The step response is the integral of the impulse response. In reverse, the inpulse response is the differentiation of the step response.
- The step response of afore-mentioned high end system has been publicized. It has widely spread starting portion, 1000μs wide, and the form is not symmetric. The high, mid and low range have different step-up timings and peaks respectively. Especially the low range is obviously delayed, and the mid range has negative peak.

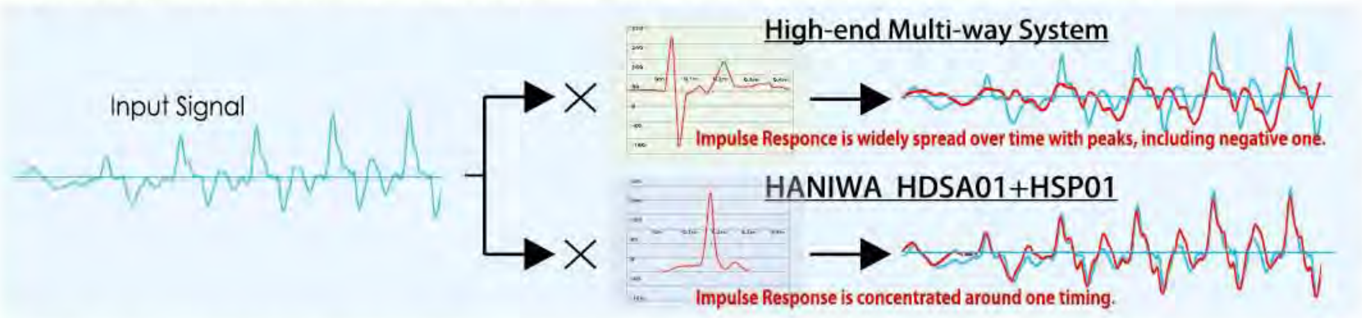



## HANIWA HDSA01 + HSP01

- The step response of HSP01 is shown on the right.
- It rises up quickly in 100μsec, and following the sharp step up, it decays gradually.
- As stated above, its impulse response has a sharp and high single peak, concentrated within a narrow width of 100μsec.
- Accordingly, its step shape is clean without the separation into high, mid and low peaks which is obvious in the conventional multi-way system shown above.


# How the Impulse Response affects the Waveform

- Impulse Response of a system shows how the input signal is deformed as it passes through that system.
- Input to the audio system is "waveform" that is the music information itself, and the output is the sound waveform coming out from the speaker. As stated at the top of this page, if the inpulse response is closer to "1", the output waveform will be closer to the input waveform, and will become almost the same. This principle is well illustrated by the examples of waveforms shown below.
- The Impulse Response of the conventional high-end multi-way speaker scattered over wider period with the peaks toward plus and minus, the waveform is heavily deformed, dull and with less articulation.




- Due to the accuracy of HANIWA sound, individual tone of many instruments are separable even when they are played together. So, listeners can easily point at players and can feel the whole stage in front of themselves.
- This is the REAL 3D Audio that makes audiences feel the reality almost touch-able.

HANIWA

# FULL INTEGRATED REAL 3D AUDIO SYSTEM

HANIWA has been working on realization of true sound fidelity closest to the original music performances that have been recorded on disks and other media.

For that end, we developed a small speaker system that does not emit sounds other than from the input signal, and is tightly coupled with a system amplifier that fully utilizes the latest digital technology. These two components are optimally tuned to work as one consolidated audio system.

HDSA01 is a system amplifier with the unique feature of built-in digital phase control funciton. It converts variety of the input signals to the common 192kHz/24 bit digital format, and drives HSP01, a pair of small full range speakers, after pre-adjusting the input signal for the compensation of both of the amplitude and phase Frequency Characteristics of this speaker pair.

HSP01 has the compact and rigid enclosure of Aluminum full-body casting, and is shaped uniquely realizing the virtual focal point of sound. This lean enclosure houses a small but extremely powerful full range unit, driven by HDSA01 to play whole music bandwidth from the deep bass to the highest tone, with full impact and energy. Eventually, the amazingly realistic sound coming out of this system presents almost touch-able music space in the listeners' audio rooms.

## HSP01

### Full Range Clear Focus Speaker

| | |
|---|---|
| **Full Range Unit** | 3.5" Aluminum cone |
| **Bandwidth** | 30Hz - 30kHz (-10dB) |
| **Impedance** | 3.5Ω at 100Hz |
| **Dimension (WHD)** | 400 X 1,090 X 400 mm |
| **Weight** | 10kg |

## HDSA01 Digital Phase Control Aystem Amp

| | | |
|---|---|---|
| **Input** | **Analog** | RCA : 2ch (stereo) |
| | **Digital** | Optical : 2ch / Coax : 2ch / AES : 1ch |
| **Output** | **Analog** | RCA : 1ch (stereo) |
| | | Speaker Output : 400W (at 4 ohm) |
| **USB Port** | | USB 2.0 : Bidirectional with PC |
| **APDC Unit** | | Independent control of Gain and Phase |
| **Dimension (WHD)** | | 316 x 95 x 370mm |
| **Weight** | | 5.2 kg |


By connecting various audio equipments to HDSA01, variety of music sources can be played with precision and richness. HDSA01 can also be connected with PC bidirectionally for digital recording and play back, without additional recording devices. Furthermore, through the connenected PC, users can receive the streamming data of music and can upload the live audio via the built-in high quality A/D converter. These should extend users' musical lives over the network.

As such, the system of HDSA01+PC can be the center of sophisticated digital music life, and will be the effective tool of opening a door to the one-step higher new audio world.

**GENERAL HANIWA BUSINESS CENTER**
**KUBOTEK USA Inc.**
2 Mount Royal Av. Suite 500, Marlborough, MA 01752, USA
Phone: 508-229-2020 Fax: 508-229-2121

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