

# SPEAKER SYSTEM BASED ON TIMEDOMAIN THEORY

## INTRODUCTION

50 years has passed since “Acoustics (Beranek)” issued, and 35years has passed since “Music, Physics and Engineering (Olson)” issued.

Many acoustic components were produced with electric sound reproduction technology, and countless numbers of components were popularized.

But basic quality, such as “Good sound”, or “High Fidelity” has progressed very slowly. We cannot say the up-to-date speaker system or amplifier is better than the system 20 or 30 years before.

This typical phenomenon is applied only in audio engineering area, not in another engineering area such as computer engineering or Car technology engineering.

It is doubtful that the harmonic distortion or frequency response is proportional to “Sound goodness”; in spite of the harmonic distortion or frequency response is the

basics of sound technology.

We cannot tell which is better the vintage 0.1%-distortion amplifier and up-to-date 0.001%-distortion amplifier. (There used be a amplifier company called “Point-One”, titled by the amplifier’s distortion) In fact, we can tell which is better only after listening.

In electric-acoustic-reproduction world, audio R&D has been done based on the distortion as a key parameter, which results almost no difference in listening (in spite of 100 times difference in distortion).

On the other hand, in car engineering, it is no doubt that the acceleration and speed will change dramatically if the engine output changes 100 times.

As the same, we have many doubtful points in frequency response. But here we focus on distortion, and we will discuss about frequency response another chance.

## BACKGROUND

In 1980, I was much moved and inspired I had never experienced when I listened to the music in the famous hall in Europe. (Picture.1) I found many people also affected who were not interested in music itself.

I found “Sound goodness” is important factor, i.e., necessary condition for musical emotion. Besides music piece or performance.

After I had experienced this ultimate emotion, I was inspired to deliver this kind of musical emotion to many people using electric-acoustic reproduction system. So I started researching and developing this kind of reproduction system.

In conventional audio, audio system was evaluated in frequency domain (F-domain). Sound wave is expressed by the collection of sine waves; reproduction system should express all of the sine waves exactly.

Mathematics taught us the sound wave is expressed by the collection of the sine waves, and using the analysis-measurement tool, we can see the collection of the sine waves for the sound, so we were apt to think “Sound” = “collection of the sine waves”. But there is difference between “sound wave is expressed by the collection of the sine waves” and “sound wave is made of the collection of the sine waves”.

Fig.2 top is the tone-burst. This signal is

Fourier transformed to Fig.2 center signal (collection of sine waves). Fig.2 bottom signal is the summation of Fig.2 center signals. We can see the difference between Fig.2 top signal and Fig.2 bottom signal. But if we sum up to infinity frequency sine waves, Fig.2 top should be the same as Fig.2 bottom.

If we sum up left half parts, it should be the mute sound. Also if we sum up the right half parts, it should be the 8 sine waves.

In other words, “Mute” is “8 sine waves”!!

It seems to be very strange.

In Timedomain audio, sound wave is treated in time domain. Sound is the pressure change

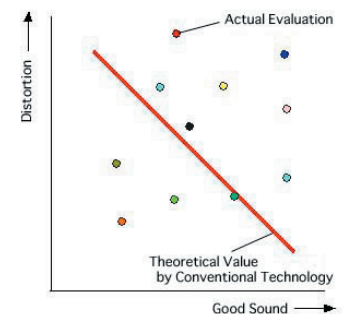


Fig.1  
Distortion vs. Sound goodness relationship



Picture.1  
Staatsooper(Wien)



**TIMEDOMAIN**  
Timedomain Corporation

in time domain, which is sensed by the ear. The faithful reproduction means, "Faithful sound-pressure wave reproduction" in Timedomain audio.

Let's discuss about the tone-burst wave as mentioned before. The left half is mute part, which lead to no reproduction (no sine wave). Only we have to reproduce the right half 8-cycle sine waves faithfully.

In short, high fidelity means faithful frequency reproduction in conventional F-domain way of thinking, on the other hand in Timedomain way of thinking, high fidelity means faithful sound-shape reproduction.

We can apply Timedomain theory to anything about sound. So we will apply Timedomain theory to all audio components. Now we have released the amplifier and speaker system. So next we explain about speaker system.

## CONVENTIONAL SPEAKER CONSTRUCTION

Typical conventional speaker system is made by the rectangular wood box, woofer, squawker, tweeter, and dividing network which divide the amplifier output to Low range, Mid-range, and High-range.

But in principle, it is impossible to reproduce the original sound using this kind of system. Of course, it is possible to reproduce original signal synthesizing each signal part mathematically or electrically. But acoustically it is impossible.

One example, if you input the same level of sound in reverse phase, output should be zero mathematically or electrically. But in reality, we can hear reverse phase sound from 2-speaker unit acoustically.

Also in F-domain way of thinking, basic idea is sine wave, which is the fundamental element of F-domain sound.

In natural world, simply repeated, continuous sound like sine wave does not exist. We are apt to mislead the result in F-domain way of thinking.

Fig.3 indicates the concept of the conventional speaker system, which is inputted sine wave. If its frequency is 1kHz, 1kHz sine wave is radiated from tweeter, and woofer, even though the level or phase is different. The enclosure box is vibrated at 1kHz, 1kHz signal is radiated from the panel of the enclosure box, and even some amount of harmonic distortion is increased.

It should be pure 1kHz signal both by measurement and by listening. Also theoretically, it should be 1kHz even added different level of sine waves or different phase of sine waves.

Fig 4 is the concept schematic of impulse signal input to the system.

Impulse signal has the same nature as the natural sound. Impulse signal is radiated from both tweeter and woofer. But their addition is not the same as the original impulse signal. The result is different from the case of sine wave. The more number of speaker units, the result will be much more different from the original impulse signal.

The enclosure box is vibrated by the impulse signal. But its vibration will remain after the input signal disappeared. This remaining vibration sound has no relation to the original signal. If we call this remaining sound "distortion", the distortion ratio is infinity. (Because the original signal is zero)

## SPEAKER SYSTEM BASED ON TIMEDOMAIN THEORY

We will show the basics of the dynamic speaker in Fig.5. Voice coil current in the magnetic field generates the sound force. This force vibrates the cone diaphragm, so this vibration according to the sound signal shape generates the sound, and then the sound transmits to the space.

### <VIRTUAL GROUND>

For the accurate cone diaphragm movement, the basic still-ground point, magnetic circuit must not be moved. Thinking about conventional speaker system, the magnetic circuit is fixed by the frame at the enclosure

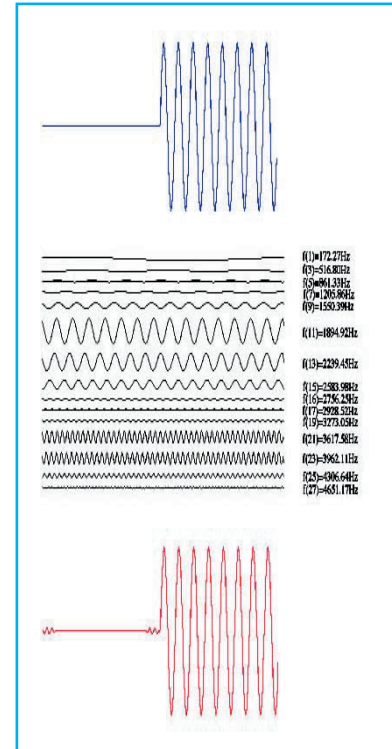


Fig.2  
Tone burst signal and Fourier transformation

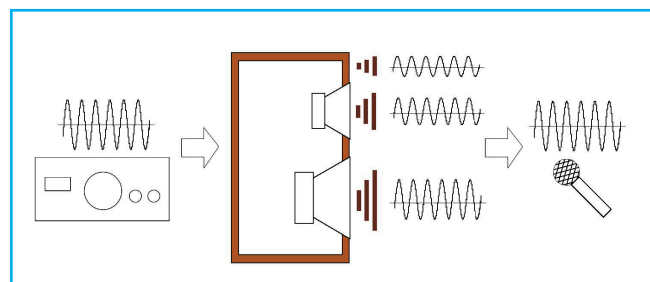


Fig.3  
Sine wave reproduction model  
: Sine wave is sine wave even if any same frequency sine wave synthesized

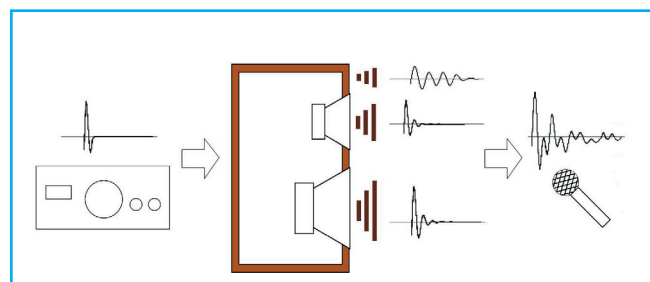


Fig.4  
Impulse response reproduction model  
: Music signal is similar to impulse signal. Synthesized music signal is different from original.

box panel. The voice coil current vibrates the speaker box and frame. In this case, the box and frame are not the still point. The sound from the cone diaphragm is not pure original signal in this case. (Fig.6)

Ideally speaking, the magnetic circuit should be fixed at the ground. But in real case, the voice coil current induces the vibration to the ground and the con-rod, which connect between unit and ground, so the ideal case will not come true.

In our Timedomain way, the magnetic circuit is fixed at the virtual ground (Fig.7) Virtual ground is the metal shaft, which mass is 1000 times heavy as the diaphragm mass. This virtual ground is supported by the GEL material, which does not transmit the vibration. So this virtual ground is the ideal ground: basic still point.

#### <SMALL DIAMETER, SINGLE UNIT>

For accurate reproduction of the sound pressure waveform, speaker unit should be the small diameter, single unit. The original waveform is not reproduced by the synthesis from the conventional multi-way units. The large diaphragm will induce the divided vibration; also induce the inaccurate movement because of the heavy mass.

Our pipe type “Yoshii9” adopted the 5.5cm-diameter diaphragm, which mass is only 1.4g. This mass (1.4g) is less than 1/10 of the 20cm units. It is the same case such that the car weight is 10 times heavier, quick start and stop will be difficult even how this car has strong engine and brakes.

#### <PIPE TYPE>

The conventional enclosure is composed of the panel, so the enclosure box has its own rigid body vibration. This vibration cannot be stopped how it is reinforced. As

we mentioned before, this kind of vibration generates the disturbing sound. On the other hand our enclosure is “EGG-SHAPE” or “PIPE TYPE” using our Timedomain method. These shapes seems to be unique, also they are so rigid as you know. Even if the disturbing sound generates from these enclosure, only little part of the sound reaches to the listener.

Here we explain our PIPE-TYPE SPEAKER called “Yoshii9” as a typical example of our Timedomain method.

Pipe has the similar nature to the exhaust pipe of the automobile, we prefer to call it “Flow-Pipe” rather than enclosure.

This pipe supports the speaker unit with the virtual ground. The speaker unit is separated by the GEL material from the pipe, so the vibration does not transmit to the pipe.

The pipe is made of the aluminum. The aluminum surface is the hard alumite layer after honing treatment. Pipe shape is originally rigid, so it does not vibrate by the inner sound pressure.

The sound pressure wave from the backside of the speaker unit transmits through pipe decreasingly, and then goes out from the bottom.

Inside the conventional enclosure box, there is full of dirty sound such as standing wave sound, diffraction sound, reflection sound, and the sound through absorbing materials. You can recognize this dirty sound if you hear the microphone sound or stethoscope sound set inside the box. This undesirable sound transmits through the speaker diaphragm and reaches to listeners. You can hear this sound if you put the sound-making substance inside the speaker box. You can recognize that the speaker diaphragm has almost no sound-shield performance.

Yoshii9, which frame construction and pipe

coupling is well considered that the backside sound pressure wave from the unit transmits quite smoothly. So the backside sound itself is good enough, and never goes back to the front side.

As a result, the sound becomes quite real, so we can hear the very small sound that is never heard from the conventional system. Also we can hear the delicate sound expression.

If the speaker unit is mounted to the conventional enclosure box,  $F_0$  goes up because of the box-air spring addition to the original  $F_0$  value.

Using the Timedomain pipe, air volume moves in pistonically as the speaker diaphragm moves. So the air mass in the pipe is added to

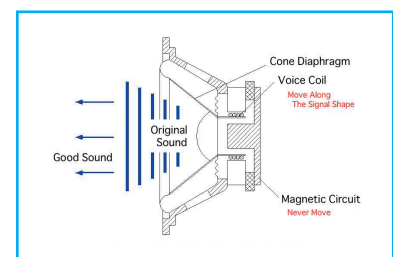


Fig.5  
Concept schematic of dynamic speaker

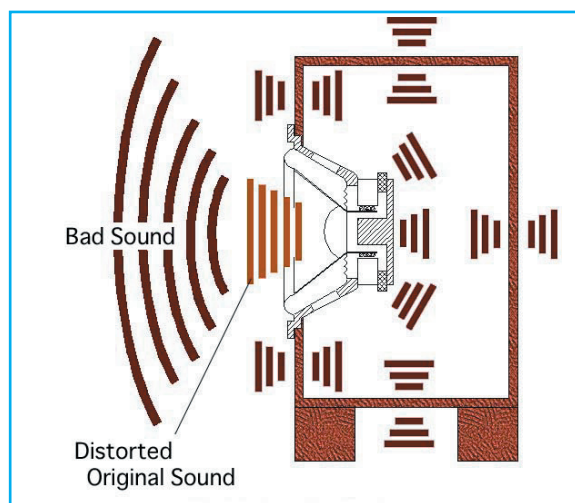


Fig.6  
Concept schematic of conventional speaker system

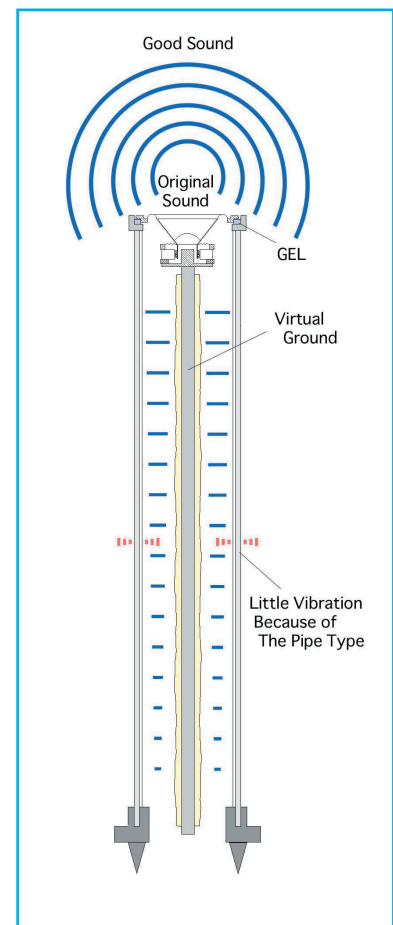


Fig.7  
Concept schematic of Timedomain Speaker system

the  $F_0$ ,  $F_0$  value goes down from the original  $F_0$  value. And the pipe controls the airflow, so Yoshii9 has superior bass expression that never heard from the conventional speaker system.

#### < TIME - AXIS DISTORTION ; COLLUPTION>

Thus speaker system's reproduction capability and expression capability are increased, the same level quality is necessary for driving amplifier.

Conventional F-domain way-distortion, i.e. linear distortion (Frequency response) and non-linear distortion (harmonic distortion) is compensatable, but time-based distortion, which Timedomain proposed, is impossible to equalize or compensate.

As this type of distortion is impossible to recover, it is better to call it "diffusion", "collapse", or "loss", than "distortion". So to speak, it is increase of entropy.

Entropy increase occurs anywhere in the amplifier. Most of the increasing point in electric-mechanical part than the electric circuit.

We notice much of the sound part is lost when we hear the sound using conventional amplifier. And we have impression that all the sound is dull.

The amplifier of Yoshii9 is designed not to increase entropy as possible.

We will describe about this amplifier another chance.

#### SYSTEM SOUND BY TIMEDOMAIN THEORY

Speaker system designed by Timedomain theory is all horn type; GS-1 (Picture.2) by ONKYO Corporation, egg type system bundled with FUJITSU-PC, egg type system by FUJITSU TEN Limited. egg type; "TIMEDOMAIN mini (Picture.3)" and "TIMEDOMAIN light" by our company, pipe type system by UCHIDA YOKO Co., Ltd., pipe type system by M.E.T. Japan Co., Ltd., pipe type system by SANYO Electric Co., Ltd. and pipe type; "Yoshii9" (Picture.4) also by our company. Their outlooks are quite different, but their sounds are commonly quite natural.

They will reproduce real sound image and real sound stage. So they will even reproduce atmosphere.

Furthermore they will reproduce the figure of the sound, the sound is hard to collapse, and so it is easy to find out one instrument sound in the noise. And their reproduced sound reaches far away.

We come to notice that the recorded sound has full of information even in old time recording, or any media, just we could not reproduce it as if we thought it was the limitation of the recording, limitation of the media, or reproduction tool.

We summarize the sound feature in Table.1

#### FUTURE PRODUCT STRATEGY

Our Timedomain sound is admired, of course by musician, and also by the people who are not interested in audio or music. "Reproduced music emotion for all the people" is our desire. Timedomain Theory is applicable to all things about sound; we would like to deliver reproduced good sound and reproduced emotion for all over the world.

(TIMEDOMAIN CORPORATION)



Picture.2  
The first product based on Timedomain Theory: "GS-1" by Onkyo Corporation (1983 released)



Picture.3  
Amplifier installed, egg-type, Timedomain Speaker: "TIMEDOMAIN mini" by Timedomain Corporation (May/2001 released)



Picture.4  
Yoshii9 (Tube-type Timedomain Speaker system and Timedomain Amplifier) by Timedomain Corporation (July/2000 released)

I feel so natural.  
I never feel tired even after long time listening.

Sound image is real.  
It seems to be actual musician existence.  
Time Domain system describes sound distance, sound width, and even sound height.

Deep sound stage.  
I can feel the recording atmosphere.

Sound image is completely separated from the speaker system.  
I do not feel the sound is generated from the speaker system.  
I feel like that the sound is generated from the space.

The sound does not collapse even at the long-distance from the speakers.  
The sound reaches long distance as if sound pressure does not decrease.

I can hear without collapse even in small sound.  
I can hear the sound clearly separated from the noise.

Sound separation between sound-images is good.  
I can hear various sounds in the orchestra.

I feel quite real about the background noise.  
The acoustic instrument sound is quite real.  
So I can feel delicate expression of the music.  
I can hear the conversation so clearly.  
I can hear English pronunciation clearly.

I can recognize the lip shape, tongue movement, even teeth movement.

No exaggeration of the consonant sound, also no booming sound.

I can hear semitone of the bass instrument so clearly.  
Also I can recognize the variation of the various touches or bowing of the bass instruments, and variation of the music tone.

I can easily hear the small background noise of the music performance reproduction such as audience whispering.

I was very surprised at the real, high fidelity sound from the old recording, LP, compact cassette, and TV sound.

The fundamental tone starts just from the sound or voice beginning, so the sound is very natural.

TABLE-1 Timedomain system's sound feature (impression by listeners)



技術と理論

ライセンス商品

雑誌掲載技術文献

NEW APPROACH HOW TO  
REPRODUCE HIGH-  
FAITHFUL SOUND (1)  
[NEW APPROACH HOW TO  
REPRODUCE HIGH-  
FAITHFUL SOUND \(2\)](#)  
[NEW APPROACH HOW TO  
REPRODUCE HIGH-  
FAITHFUL SOUND \(3\)](#)  
[NEW APPROACH HOW TO  
REPRODUCE HIGH-  
FAITHFUL SOUND \(4\)](#)

TimeDomain Corporation has been in research to looking for good sound since its foundation. One of the basic technological theory is "time domain".

The series of articles were appeared in "Radio Technology" magazine from July to October in 1983.

The concept of "Time domain" is explained very clearly to understand in these articles.

So, please enjoy reading to help your understanding.

Although it had been written more than 19 years ago, its theory and technology hold good even today as long as you consider the change of environmental situation.

We will link new material and information from that time down to this day into the main body additionally without changing the original article.(980904)

## NEW APPROACH HOW TO REPRODUCE HIGH-FAITHFUL SOUND (1)

**Why physical characteristics do not match with the sense of hearing?**

- [Distortion relating to time axis](#)
- [Proposal on Multipath-ghost distortion](#)
- [The reason of discrepancy between conventional frequency characteristics and the sense of hearing.](#)
- [Distortion relating to amplitude](#)
- [Operational difference between horn-type and corn-type](#)
- [Less Distortion in Horn-Speaker](#)

Nowadays, many people consider that in most cases frequency and distortion characteristics are not match with the sense of hearing by conventional methods. Are there any mistakes in the way of detection? Or, do we need another approach? If we pay attention to the basic issues, what kind of new aspects will appear? Nowadays, many people consider that in most cases frequency and distortion characteristics are not match with the sense of hearing by conventional methods. Are there any mistakes in the way of detection? Or, do we need another approach? If we pay attention to the basic issues, what kind of new aspects will appear? These are main subjects in this article. Especially, we have to pay a big attention to the time axis distortion in the future. (Editor)

New audio products are coming out into the market one after another. Always with copy "how to playback faithful sound ". Most cases only mentioning "make frequency response flat and better to reduce harmonic distortion", not about its technique and theory.

But, we can not playback music sound faithfully with only frequency response and distortion.

Those are important factors for sound playback, but not enough if other important factors are inferior.

I would like to present some important subjects which have been neglected. Let's think together, and I would like to accomplish an audio technique which satisfy us all in music playback. The neck is a speaker system as everybody admits. So in this article, I will carry on with this subject. Speaker system has been remaining as it is for 10 years, 20 years, even more than 50 years in this rapid progressing society. It is impossible for other industrial products to stay in the market like this way.

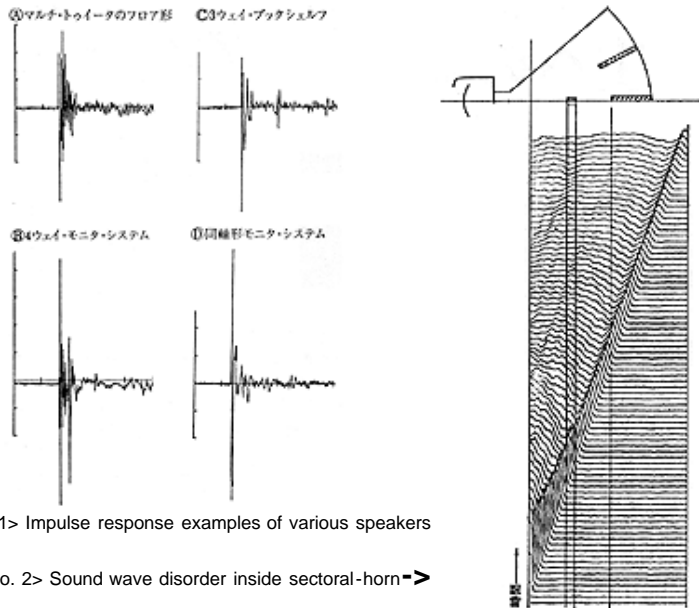
If you can understand music well, it will be possible to create excellent sound speaker-system by making the best use of high technique, engineering method, and new way of thinking.

### Distortion relating to time axis

#### Proposal on Multipath-ghost distortion

Played sound has very unique tone that everybody can recognize as " played sound". If you input impulse signal to the system, it would sound like "crack" and "slap". (Fig. No. 1) Looks like some reflection is main reason for causing overlapped sound of the impulse which should sound like "puff" .

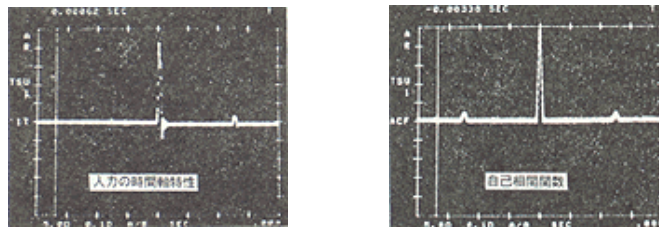
You can get the pattern on Fig. No. 2 , if you move a microphone inside a horn-type-speaker, and change arrangement of the impulse response data obtained at 50 points on the center axis by computer.



<Fig. No. 1> Impulse response examples of various speakers

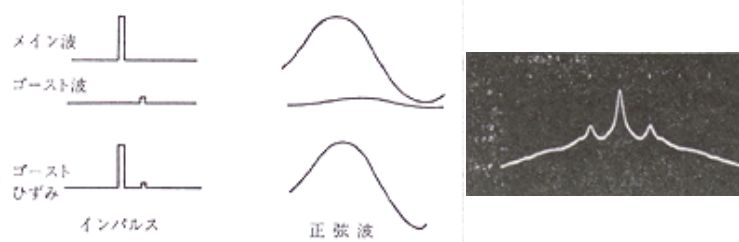
<Fig. No. 2> Sound wave disorder inside sectoral-horn=>

It shows very clearly the way of traveling of the impulse wave inside horn according to time. The impulse wave which hits a prop or a separator will reflect and return. The music sound will be changed by this phenomenon. For example, high-level area will become noisy, sounds poor, very hard to tell the nuance of the string and vocal, or sounds gorgeous and pleasant, etc. We would like to call these phenomena as "multipath-ghost distortion", since it is very similar to the ghost on TV. We can learn about small reflection by using self-correlative function and the technique of "power-capstrum". The higher correlation between two sound signals which is apart t:time in the time axis data, the higher reflection sound existence probability.(Fig.No. 3). In this figure, you can recognize that multipath-ghost distortion is 10%.



<Fig. No. 3> Impulse accompanying -20dB reflection delay and its self-correlative function

By the way, it is very difficult to find out multipath-ghost distortion if you use sine wave, even though you listen into very carefully and try to measure it out. Sine wave of 1kHz is a pure sine wave of 1kHz even you put many sine waves one over another (Fig. No. 4). When you sweep frequency, you can find unevenness in frequency response by its inter ference. But it will be under 0.1dB in case of 1% ghost distortion, and not reach to the width of the pen for recording machine. Sound has been changed without distortion (harmonic distortion) - thus we found out another distortion which disturbs a clear music playback although we cannot figure out in conventional physical characteristics. It has never been checked since this kind of distortion does not directly lead to unpleasant feeling. If you use self-correlative function, you can find out the existence of the distortion only from output signal even you have no information about input signal. There is an example of the measurement using one of the program sounds in Fig. No. 5. To detect the distortion only from output signal, this is very similar to what human beings are doing with their ears, isn't it?



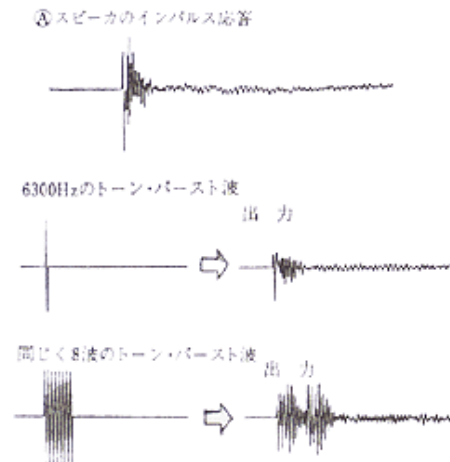
<Fig. No. 4> Distortion not clear from sine wave

<Fig. No. 5> Self-correlative function (average of 1000 times for 6 minutes) for FM broadcast disk-jockey which has gone through the signal system with multipath-ghost distortion.

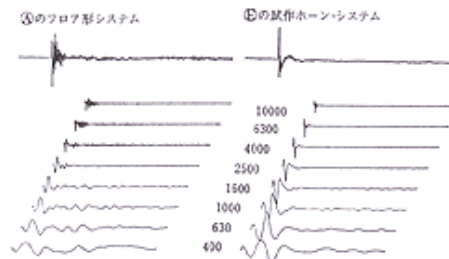
### The reason of discrepancy between conventional frequency characteristics and the sense of hearing.

If you check using this kind of measurement method, it will be clear that ghost distortion is generated not only from sound lens or some unevenness on the wall but also from previously mentioned prop and separator in the horn-type-speaker. Also this distortion will be generated at the corner from the wall point which abruptly starts expansion in a radial horn. Most horns in the market are not good regarding this point.

So we developed new type horn without multipath-ghost distortion. Fig. No. 7 and No. 8 show its characteristics.(I will explain about the details in another paper.) You can get tone-burst response by convoluting impulse response and tone-burst wave using computer.



<Fig. No. 6> Examples of impulse response and tone-burst responses



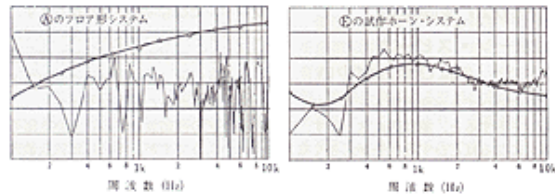
<Fig. No. 7>> Examples of impulse response and Wave-1 tone-burst response

Fig. No. 6 is the tone-burst response of the speaker which is on Fig. No. 1(A).

Burst wave is doubly multiplied by the multi-path ghost distortion. In case of the same amplitude of the wave, it should be recognized differently between two different number of the waves. Wave-1 seems to be bigger regarding input/output ratio considering from amplitude and the number of waves.

You can get Fig. No. 7, if you calculate Wave-1 tone-burst response for each frequency. Also the response wave power is calculated, then normalized by the input power. It is plotted by round marking in Fig.No.8. These given characteristics are recognized as frequency response to the shortest signal on the sense of hearing .

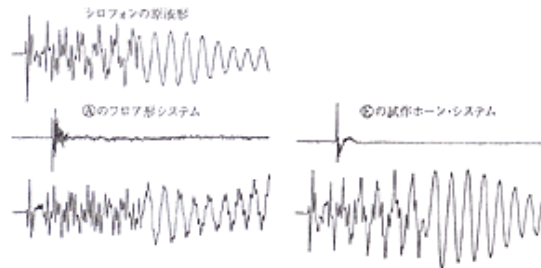
In an ideal system, this will match with conventional frequency response by static sine wave. However, sound-tone must be different between transient and continuous sound in the system like (A), so at actual source, sound will be heard high-raised tone on average. We found another reason for the mismatch between conventional physical characteristic (frequency response) and the sense of hearing.



<Fig. No. 8> The difference of the frequency characteristics against long signal and short signal.

If you convolute the impulse response and music sound together, you can get the equivalent sound with microphone output which is same as when you play the instruments using its system. The better point than an experiment by actual microphone output is we can repeat exactly the same sound as many times as we need.

See Fig. No. 9. It shows the experiment using xylophone. The experimental system shows all of its musical signal characteristics, generation of the sine waveform from the impactive noisy sound which is produced by hitting xylophone with mallet, and also envelope of music sound. On the other hand, it looks like many musical information would be lost in the system (A). However it seems to sounds very gorgeous, so maybe some people consider it is a good sound.



<Fig. No. 9> Impulse response and the response to the sound of xylophone

## Distortion relating to amplitude

### Operational difference between horn-type and corn-type

At present audio market in which frequency response is regarded as very important factor, 3 - 5 ways direct radiation type systems, namely corn-type /dome-type /flat-type etc. , are main trend. On the other hand, some maniacs prefers the special sound of horn-type, however generally speaking, its sound is treated specially as a peculiar tone or considered as not suitable to playback high-faithful sound.

The peculiarity at the opening edge of the horn is very popular, and many researches have been done.

However, as mentioned in the part of multipath-ghost distortion, there was also another distortion in the high frequency region and forming its own characterized tone.

If you take away these peculiarities, a good factor on the horn will be activated.

Actually there are many excellent points in this factor, and it is impossible to activate by another method. Let's think about one of these points, that is distortion.

Harmonic distortion which means "conventional physical characteristic" in this report will be connecting to high quality sound, if another factors have improved. However we have to evaluate not only quantity , but also the quality of the sound for music playback.

There is a possibility to create less distortion speaker in the horn-system than amplifier-system.

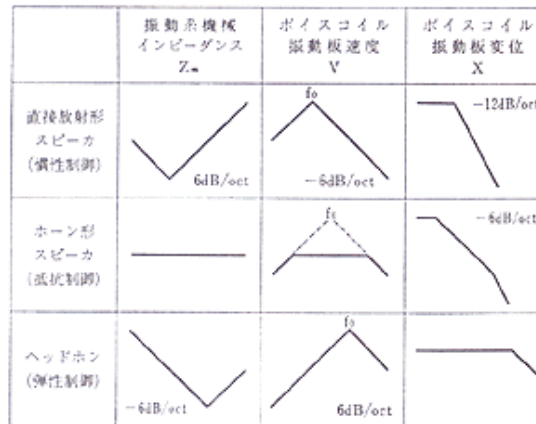
If you classify the electric to acoustic transformer in view of controlling the diaphragm, it will be classified into three systems; elastic control system like head-phone, resistance control system like horn-type speaker, and inertial control system like direct radiation type speaker like corn-type /dome type /flat-panel type.

These three systems looks like the same from the point of sound producing method by vibrating the diaphragm, however the truth is different. Each has unique characteristics from the point of distortion generation. See Fig No. 10.

As the mass is the major factor in direct radiation type, diaphragm is harder to move in the higher frequency. Therefore, the velocity characteristic will become right-side down. On the contrary, in case of head-phone, air density between ear and headphone will control the vibrating-part impedance, and getting become easier to move in higher frequency. In Horn type, vibrating-part is controlled by resistance in a certain frequency range where has enough electromagnetic

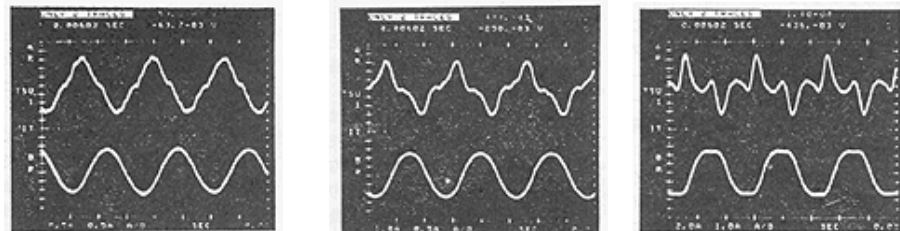


regulation and horn load. And the velocity will be constant despite the different frequency. Next, let's think about dislocation of the voice-coil/diaphragm. Since the amount of dislocation is the integration of the velocity, the amount of the dislocation of the corn type is -12dB/oct and right-side down, that of horn type is -6dB/oct and right-side down, and that of head-phone is constant. Frequency response of sound pressure is always become flat, despite frequency response of velocity and the amount of dislocation are different. That is to say, in corn type when frequency goes up, the velocity goes down, but the corn diameter is relatively larger than its wavelength, which radiation efficiency goes up. Also velocity and radiation efficiency will be constant for horn-type. For headphone, the amount of the dislocation is constant, and also the sound pressure level is constant.



<Fig. No. 10> Three kinds of frequency characteristics on electric, sound transformer.

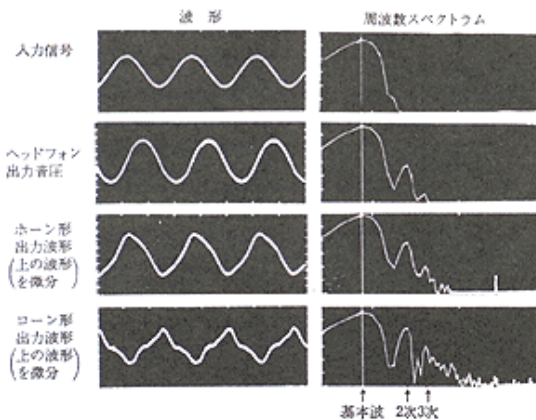
Few people understand that there are differences in diaphragm dislocation behavior in each system. Also looks like many people consider that the amount of the dislocation of the diaphragm and sound pressure waveform are similar in spite of the difference in frequency response. Even it is the same shape in sine wave, in music signal it will be different according to each system. In Fig. No. 11, you can see some sound pressure waveforms varing the corn-type speaker output from small volume to large volume until the sound is distorted. Maybe some people cannot recognize the distortion waveform (above), however if you make a transformed waveform through integrator, you can understand that itself is a saturated waveform. In case of the corn-type speaker, you can get the dislocation waveform through equalizing using -6dB/oct for velocity waveform, which means integration. (c.f. Fig. No. 10).



<Fig. No. 11> Output sound pressure (above) and transporting point on the diaphragm. Upper sound pressure is less.

### Less Distortion in Horn-Speaker

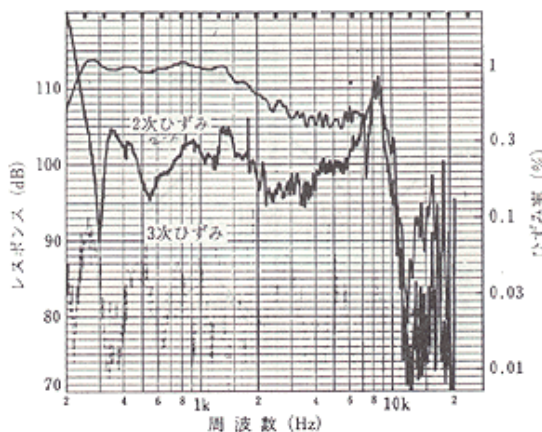
You can get Fig. No. 12 upon simulation in case that the distortion is generated from dislocation. The distortion difference becomes bigger for higher-order distortion. It is well-known that for music playback, the odd-order harmonic distortion is harmful, and also higher order distortion is harmful even if it is a small one. These things make us convince the reasons for fairly good sounds from cheap head-phone, and horn-speaker sounds pretty good even though it contains some peculiarity. We calculated the difference of distortion between corn type(direct radiation type) and horn-type in Table No. 1. This difference will be getting bigger if you consider that dislocation distortion of horn type is less due to the narrow amplitude comparing to direct radiation-type. The distortion quantity could be less in addition to the good quality.



<Fig. No. 12> Distortion waveform and spectrum of each system

Fig No. 13 shows the distortion characteristics for the experimental horn-speaker. This has been measured at sound pressure 110dB as the distortion is under noise level of measuring system. If you measure at the normal level, the value will be much less. You do not have to concern about the second harmonic distortion. Regarding harmful third distortion, in normal listening level it will be under 0 dB and it cannot be heard.

It is reported that the harmonic distortion of the amplifier is about 0.003%. But this value is measured in maximum output using sine wave. Please check the distortion data of amplifier. In the condition of under the normal listening level 1W, this amplifier is inferior to our experimental speaker. Especially, the speaker will be superior in case of delicate sound as the speaker distortion will decrease, on the contrary to the increase of the amplifier distortion. Also as the quality of distortion is better in horn-speaker (high-order distortion is less), the amplifier will be inferior if absolute distortion volume is the same. As a matter of fact, if you listen into this type speaker, you will find out that most distortion is generated from the cartridge or amplifier. ([To be continued.](#))



<Fig. No. 13> Distortion rate characteristics on experimental horn-speaker

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## NEW APPROACH HOW TO REPRODUCE HIGH-FAITHFUL SOUND (2)

Phase difference will be a subject in case of consideration from time window aspect

by Hiroyuki Yoshii

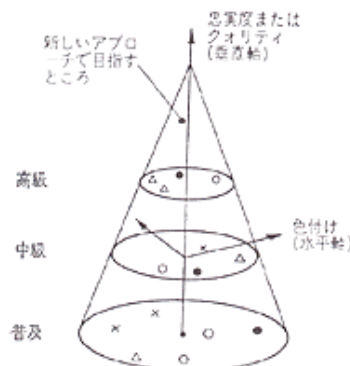
- [Sound Evaluator or Music Evaluator ?](#)
- [Impulse responses take the place of sine waves](#)
- [Distortion regarding phase and time](#)
- [Sound will be changed even that frequency response is flat](#)
- [Natural sound and high-class music](#)
- [12dB/oct Reverse phase connection is good](#)
- [Slow roll-off frequency response characteristics and sharp cut-off frequency response characteristics](#)

### Sound Evaluator or Music Evaluator ?

Looks like there have been two big main streams on how to deal with music playback. In one stream, physical characteristics and measured data are mainly concerned, and in the other, music nature and the sense of hearing are main subjects.

The former means to playback faithfully recorded signals in the record, and to avoid colorization. Most Japanese manufactures are belonging to this stream. If you regard frequency response as important, 3 - 4 ways which are using corn-type, dome, or flat unit will be main. Those data are fairly good, however when it comes to music playback, something is missing. It is right way to pursue faithfulness, however to playback music, its faithfulness is not enough yet. You can call it as "Sound Evaluator".

@The latter will be a evaluator not concerning about recorded signals itself. Music itself and feeling faithfulness will take priority over physical faithfulness. As not persisting in frequency response or distortion, there will be various unique methods. There is a well-balanced colorization including addition of room echo, consciously or unconsciously. Sometimes there is a compensation of weak points of recorded signals. Foreign products are looks like belonging to this stream. Its music sound pretty good in spite of weakness in frequency and distortion characteristics. We can call this belonging to "Music Evaluator". Due to the lack of the faithfulness, there will remain some dissatisfaction in live music playback.



<Fig. No. 1> The way of thinking to playback faithful sound

In new approach, we are trying to playback music itself faithfully. We are trying to playback music itself by the playback with faithfulness of playing method, sound image, sound-stage, and balance of music instruments.

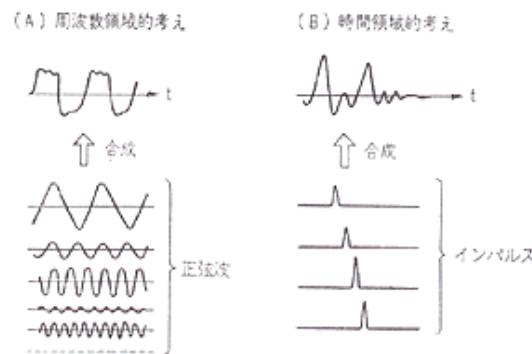
Playback music which is stereophonic recorded will not be handicapped to compare with live music. Listen into live music as many times as possible, and found out necessary and indispensable factors which is required to playback live music. We are thinking about the reason and doing experiment/verification, then trying to realize into the system.

In most conventional researches, the beginning point of the system was signal generator, and the ending point was measurement equipment connected to a microphone. In the new system, the beginning point is music, and the ending point is human heart. It will be a hard work since we are trying to replace live music with audio system which is big different from actual form. We have to research from music aspect to human data processing system and psychological field.

### **Impulse responses take the place of sine waves**

They used to use frequency domain physical data mainly such as sine wave. As every signals are consisted of various sine waves, they consider that if they could playback sine waves properly, everything goes well.

Measurement should be simple, however not enough for the evaluation of the transient phenomena like music signals. In our new approach, impulse wave will be more important than sine wave as we have to consider and examine mainly in time domain. We consider that all signals are the collections of impulse (Fig. No. 2). If we could playback impulse properly, it is possible to playback all signals including transient phenomena. It will be a big mistake, if we consider only from frequency domain aspect.



<Fig. No. 2> Collections of sine waves and collections of impulse

In the listening room, some people use pink noise as a playback signal and receive it by uni-directional microphone, then make it tuning to flat by using graphic equalizer. This is typical extreme example for frequency domain evaluator.

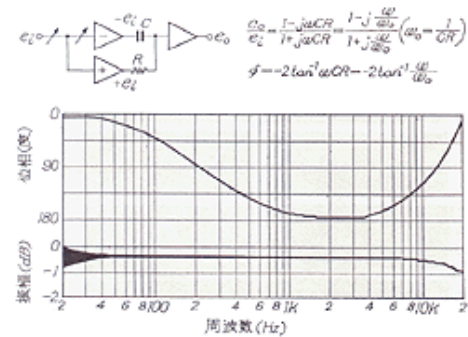
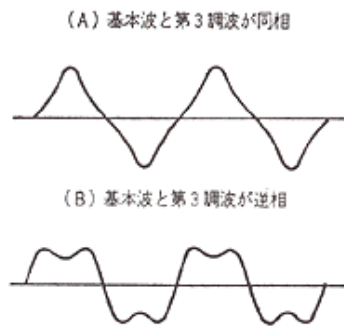
If they do not understand all aspects well enough, something goes wrong. This is an example of ignoring time axis, as the sound is equalized including the remaining sound as room echo and the direct speaker output sound. Also in this kind of idea, the sound direction is neglected because it is equalized including forward sound and backward sound. This example is extreme sample, but this kind of idea always exist in frequency domain consideration. The followings are examples.

### **Distortion regarding phase and time**

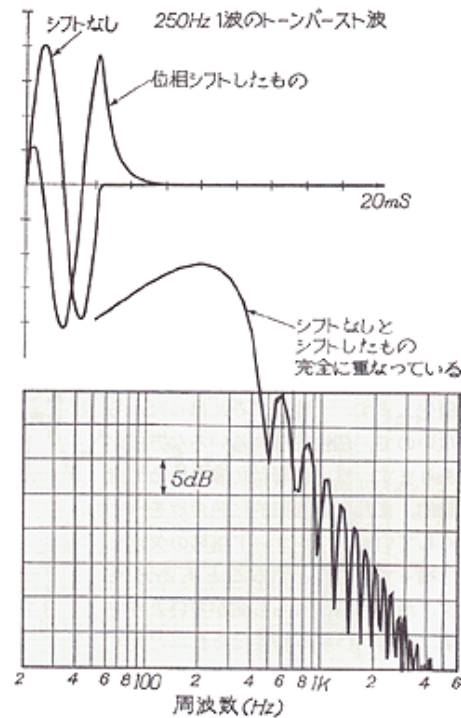
Helmholtz described that "Music tone will be decided by its amplitude which consist of sounds, and has nothing to do with phase between constituents", since then people have been considering that phase has nothing to do with sound tone.

Actually, there is a research that there is not so big difference even between extreme signals depicted in Fig. No.3. In case of our additional test we recognized slight difference, however maybe this was telling the non-linear distortion difference on ears or playback equipment caused from the waveform difference, not detecting the phase difference. Judging from these facts, conventional faithful playback method, we left phase distortion unquestioned, however when we consider about music playback seriously , we found out that this will be very important subject.





<Fig. No. 3> Reversed phase relationship signals <Fig. No. 4> Phase shifter used in experiment and its characteristics



<Fig. No. 5> Waveform and frequency response of burst wave which is phase shifted.

### Sound will be changed even that frequency response is flat

We made a phase shifter as in Fig. No. 4. Amplitude characteristics (commonly called "frequency response") has been unchanged, and only phase characteristics will continuously changed to 180 degree.

Using this shifter, we listened into various sounds. As a matter of fact, there is no difference using sine waves. Also there is small difference for simple compound wave as in Fig. No. 3, however there will be a big difference between transient sounds. It is easy to understand with not only for impulse but also for a simple one envelope of tone burst wave. In general speaker system, phase distortion exists in speaker system itself, so we will drew the conclusion that there is no difference from like this experiment.

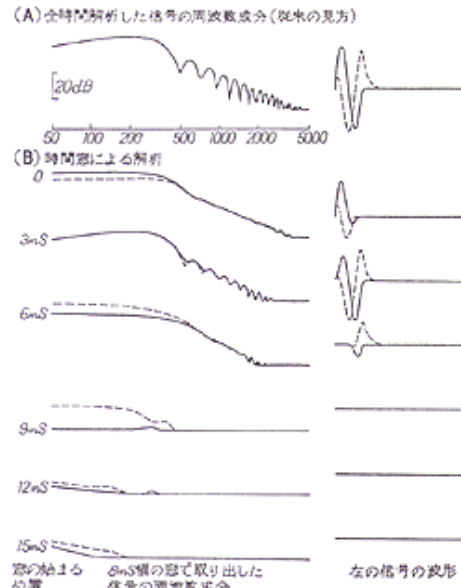
As in Fig. No. 5, you can see no difference between original signal and phase shifted signal after Fourier transform.

And yet why we hear the sound differently?

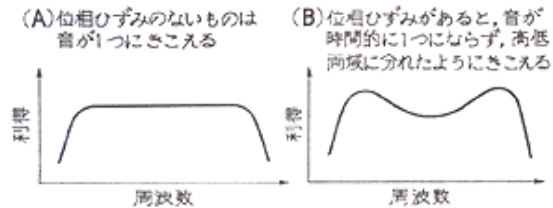
There is a key issue in time domain See Fig. No. 6. In upper figure, it is made in total time, and there is no difference like 0.1dB in each frequency elements. However, we can consider that tone is changing as time goes by even in a short period. So it will be clear that reasonable sound difference, a few dB in average and 20 dB at most, will be in frequency elements at each timing made from time window aspects.

It actually sounds as shown in Fig. No. 7, and this fits well with the result. According to this analysis, it looks like high sound in advance and low sound later to human ears from a viewpoint

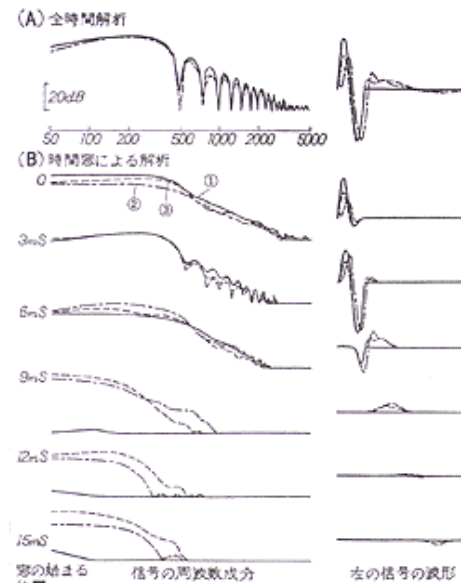
of time. Fig. No. 8 is a typical system as known frequency response is fairly good, but the same phenomenon as Fig. No. 7 is seen due to phase distortion. Now we can understand the reason why it sound so different in spite of almost same frequency response between two speaker systems. Thus we can call it a momentary tone which is limited in a short period.



<Fig. No. 6> Time observation of phase shifted signals. Many new aspects will come out from observation using new methods.



<Fig. No. 7> The way of sound through phase shifter



<Fig. No. 8> Time observation through conventional SP system.

### Natural sound and high-class music

As in the previous example, if you listened into the music through shifter, it sounds unnatural, high and low emphasized sound.

You cannot fix this phenomenon even modified with graphic equalizer. For example, if you listen into female vocal, sounds starting with "S" are emphasized, and not feminine low sounds are

stressed also. These are typical characteristics of played sounds, aren't they?

Music sounds very cheap if high-tone sound comes first then lower-sound follows. High-class music accompany with lower sound elements from the starting point, and if it is delayed the sound will become very cheap.

For example, the first class singer uses diaphragm regardless east or west, so it sounds very strong with accompanying fundamental low tone. But, an amateur sings from throat and lungs squeezing air, so it sounds very unstable due to preceding high-tone sound.

The first-class player produces high-class sounds using diaphragm and by tonguing in case of woodwind and brass instruments. In case of string instruments, low sound is mixed from starting point as playing quickly after a bow sticking to the string enough and stocking enough pressure. In case of poor player or cheap bow, sticking is not enough and as a result we cannot expect this kind of sound. Generally speaking, cheap instruments has a tendency to emphasize high-tone, and not accompanying with low sound, even so sounds not so crispy and clear. If there is phase distortion, it means not only causing time difference in frequency elements and sounds unreal due to an momentary tone difference, but also very hard to get high-class music. Time delay (group delay characteristics) is in proportion to phase slope, so phase slope should be slow and constant.

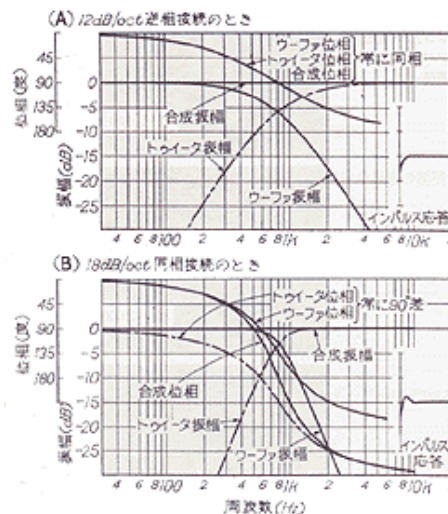
### 12dB/oct Reverse phase connection is good

In multi-way speaker system, this phenomenon will occur in dividing filter. The characteristics of dividing method is shown in Fig. No. 9. The sound will not be connected smoothly each other when listening to electrically composite sound through head-phone due to a large phase slope of composite characteristics at 18 dB/oct.

In case of sound composition in air-space using speaker, the connection will be worsen in addition to the phase difference between unit phase difference 90 degree.

In case of 6 dB/oct, composed phase is flat, but actual connection is not always good due to unit phase difference 45degree. Also it is not suitable for high-class audio system due to bad outside band decay characteristics.

From the consideration of outside band decay characteristics, phase difference between units, and composed phase characteristics, 12 dB/oct will be the best.



<Fig. No. 9> Characteristics and impulse responses of two filters.

According to this idea, the method to make it multi-way using speaker-unit for making frequency characteristics flat and wide, is not suitable for playback music faithfully. Since it is impossible to connect correctly in time domain, even smooth connection in frequency response.

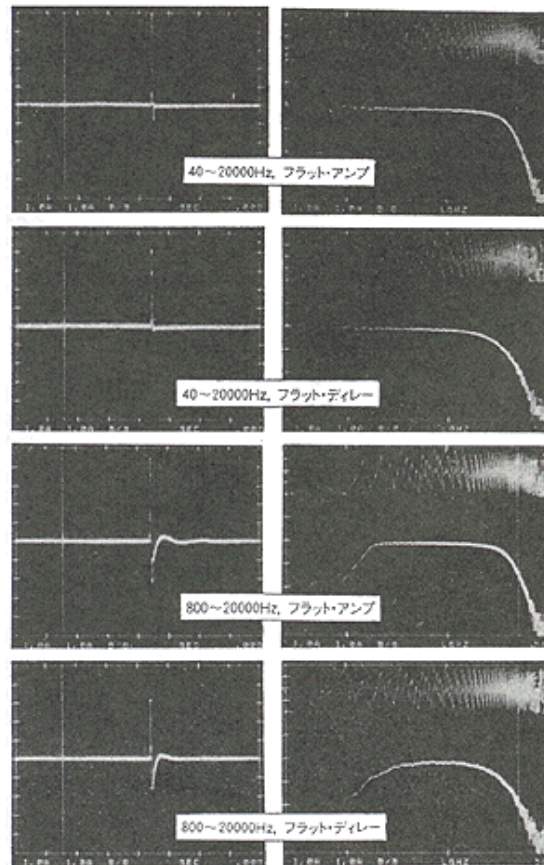
Even if it is possible to be connect correctly in calculating with electric signals, but it is impossible to synthesize two waves in short distance. So we came to the conclusion that "2 way" which connects one point with long wavelength will be the best.

### Slow roll-off frequency response characteristics and sharp cut-off frequency response characteristics

@We can say that 12dB/oct is superior to 18dB/oct from the consideration not only amplitude characteristic but also time, phase, also sound connection of composed characteristics. Generally speaking, better characteristics of amplitude characteristics are not directly connected to better characteristics of transient characteristics. This is not a case only for dividing filter. It is not suitable filter for audio-frequency band that is considering only amplitude and has flatness to its maximum. Maybe it is suitable for another purpose.

Gentle slope filter has better characteristics such as less phase shift, and better group-delay. In case of the same slope filter characteristics, the filter which has slow roll-off characteristics in shoulder area exhibits the better time-phase characteristics. Fig. No. 10 shows impulse responses which has the flattest amplitude characteristics 24 dB/oct filter and the flattest group delay characteristics 24 dB/oct filters .

Listening into various kind of music, it sounds natural and high-faithful when the filter shape is slow roll-off and group delay flat. It is natural that the speaker system which shoulder is too overhang to extend the lower range doesn't sound good, since it is a system to deal with transient signals. ([To be continued.](#))



<Fig. No. 10> The relationship between sharp cut-off frequency response and slow roll-off frequency response using impulse signal.



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## NEW APPROACH HOW TO REPRODUCE HIGH-FAITHFUL SOUND (3)

**A proposal of New Woofer aiming full-range resistance-control**

by Hiroyuki Yoshii

- [Force, stronger is better](#)
- [A proposal of new woofer system](#)
- [Reverb Distortion from board vibration](#)
- [How to get rid of reverb distortion](#)
- [The results of new system](#)

### Force, stronger is better

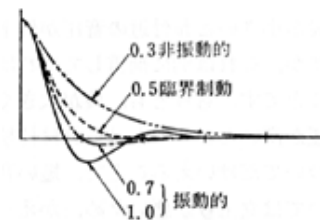
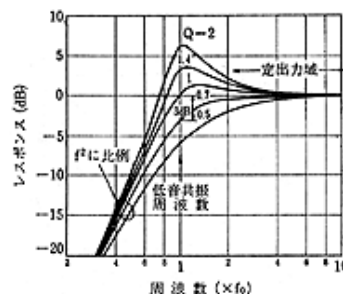
I studied very hard about low tone domain when I developed sub-woofer called "SL-1".

I actually got two aspects to learn after its release. First aspect is that Japanese (Oriental) are not so interested in low range. In music listening, they are listening mainly into high range of melodic part, not listening into low range such as thoroughbass. They can understand well how necessary and wonderful in low range sound if they listen into high quality sound, but it looks like they do not pursue low range quality particularly.

Secondly there are few good low range specified system to utilize sub-woofer. Originally we can expect a wonderful effect when connecting sub-woofer to good low range specified system.

However, most people seems to be satisfied with the fact that subsonic sound reproduced.

Now when we consider conventional woofer system from the viewpoint of extra high faithful sound playback, we can not satisfy especially with transient characteristics. Let's think about closed box system known as good transient characteristics. The low range characteristics of the system is determined by Q-factor, however it has been recommended about 0.7 ~ 0.9 from the point of frequency response flatness (Fig. No. 1).



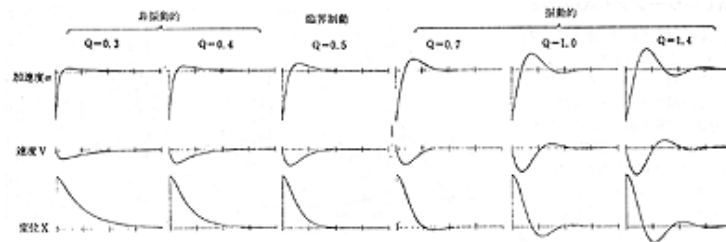
<Fig. No.1> Woofer Q-value and frequency response <Fig. No.2> Conventional explanation of Q and transient characteristics

Please see the Fig. No.2 for explanation,  $Q = 0.5$  is the critical control point, and if  $Q$  is bigger than that, it will be more vibratile. However from the point of frequency response and vibratile convergence, it has been considered that the ideal value for  $Q$  is 0.7 ~ 0.9.

It has been explained that if magnetic force is stronger than above value,  $Q$  dropped too much and become over-damped, and as a result the response will be small. Conventional system has been designed based on this theory, however this is a problem.

First of all, my intuition told me it is strange to say that the strong magnetic force is no good.

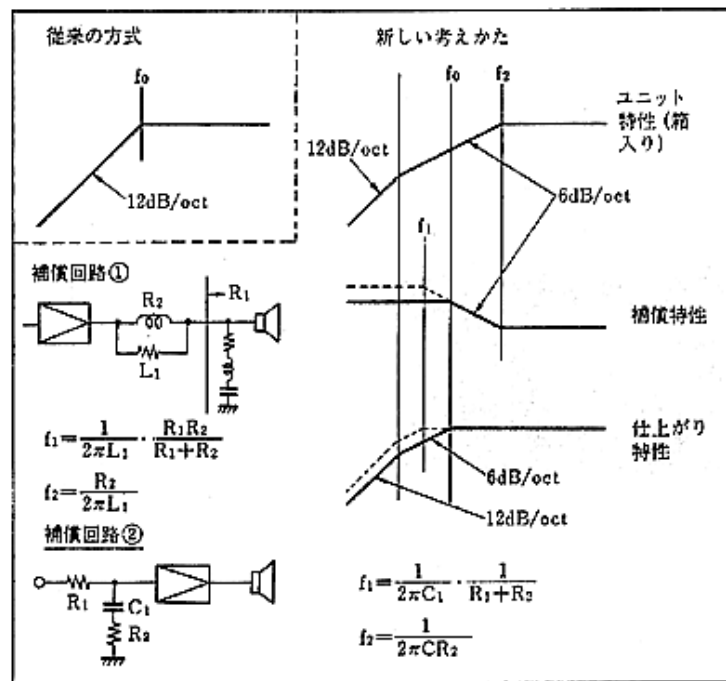
Also it looks like there is some misunderstanding that the movement of corn-diaphragm is equal to sound pressure (please refer to Fig. No. 10 in June issue). You can get displacement, velocity, and acceleration as shown in Fig. 3, after solving the differential equation of the movement. It looks like 0.7 is suitable for displacement convergence, but not for the acceleration and velocity which is equivalent to sound pressure. The smaller Q seems to have the better transient characteristics.



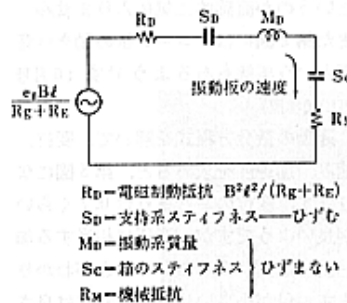
<Fig. No. 3> Proper way of thinking for woofer Q and transient characteristics

If Q is small, the sound pressure around  $f_0$  will become lower. But this can be solved by compensating separately. However it is only true for continuous sound that the bigger Q-system has the higher sound pressure. Probably it will sound smaller for momentary sound since rise-up is not so good.

#### A proposal of new woofer system



<Fig. No.4> Principles of new way of thinking for woofer.

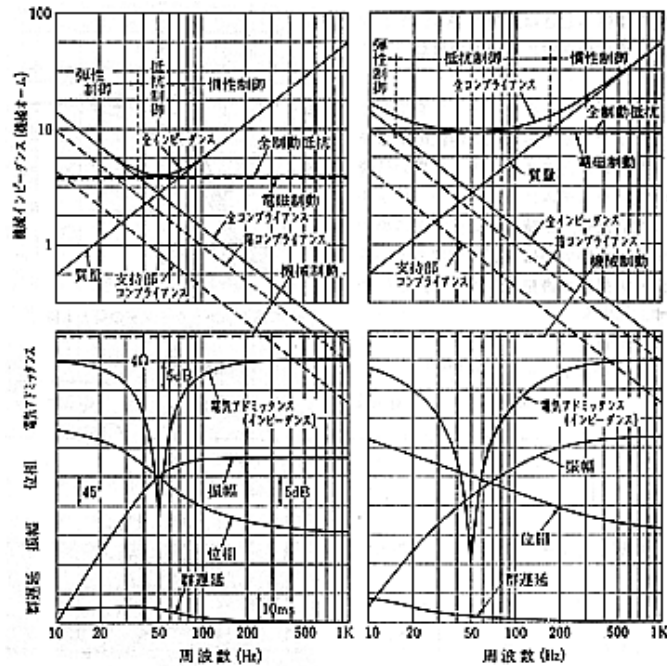


<Fig. No.5> If we make the electromagnetic control resistance R large enough, then distortion will become smaller and transient characteristics will be improved.

The way of thinking for new system is shown in Fig. No. 4. If the magnetic force become stronger, not only the transient characteristics will be getting better but also conventional physical distortion will be reduced (Fig. No. 5).

Most distortion will generated from non-linearity of the suspension. If you make the magnetic

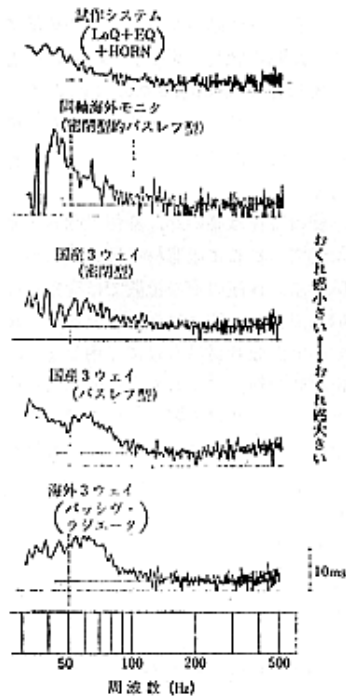
As impedance is pretty big due to electromagnetic control, in case of 10% distortion in supporting part at  $f_0$ , it will cause only 1% distortion. However it will cause 3% distortion with conventional method as shown in Fig. No. 6.



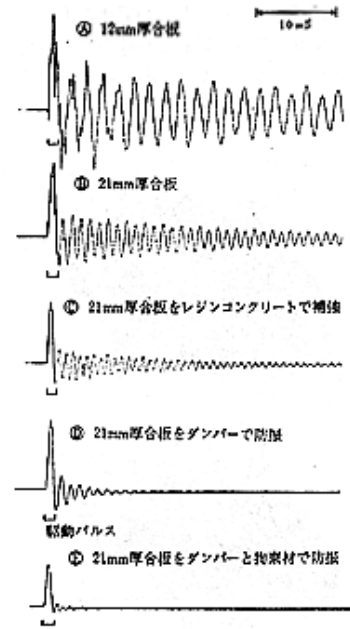
<Fig. No.7> Various woofer characteristics at  $Q = 0.3$ .

Also you can understand that with new method, resistance control domain with more than 4 octave range will exist although with conventional direct radiation type, such domain never exist. So we can expect this system can playback quite different sound to compare with conventional system due to highly improved phase characteristics and group delay characteristics, etc. In experimental system, we modified the system from mass-control to resistance control by installing short horn load, as a result, resistance control range covers all woofer range. We have listened carefully, low range sound like quite different as we expected. Every low range instrument sounds real. Each instrument sounds differently, for example, contrabass, cello, bassdrum has different character to compare with the conventional tone which use to sound similarly. It is so differently perceived by every string sound of contrabass. More objectively, it becomes possible to perceive the sound more clearly and separately like "dah!dah!dah!dah!" by the spiccato of contrabass, which used to be perceived like the sound like harmonica. As I mentioned in the previous issue, of course it is true that the sound will be grade up.

We listen into music by paying attention to the delay of low range, and compared with the conventional system according to the sense of hearing (Fig. No. 8). We got the feeling of equal interval each other, however when it comes to our experimental system the story is different. There is no feeling of delay especially. If we get the group delay characteristics from the impulse response, it will be consistent with the sense of hearing perfectly.



<Fig. No.8> Delay feeling of low range and group delay property.



<Fig. No.9> Effectiveness of panel vibration and preventing vibration effect.

### Reverb Distortion from board vibration

After the adoption of much better unit and method with excellent transient response, the next issue will be the materials for the box and horn.

Vibration will not cause distortion in frequency domain but in time domain, and as a result, it will cause bad influence to playback of music.

We would like to name it as "reverb distortion" since it will linger as shown in Fig. No. 9. It is very hard to detect the reverb distortion with conventional sine-wave measurement method even though it comes out a little bit to frequency response in case of large reverb distortion, however it won't be detected as harmonic distortion.

It is very similar to the echo in the room. And we think we can call it echo as same as conventional way since we can clearly tell the difference from reverb distortion according to the following two reasons. One is the big time difference to compare with the original signals, and the other is the reverb generation from different direction with main signals. So, we would like to treat as distortion which is generated in the system and difficult to tell the difference with main signals. Reverb distortion will cause bad influence to the sound of music as same as multipath ghost distortion which is defined in June issue. Bad influence means worsen sound separation, less tone difference, and foul sound with inter-modulation distortion, worsen sound image localization, etc. In the conventional system, it was not a serious problem since reverb distortion was not unpleasant to the ears. On the contrary, it used to be utilized appropriately like poor "Karaoke", which need some reverb.

### How to get rid of reverb distortion

#### (1) Prevent vibration

The biggest distortion will generated in the box. Conventionally, by the reinforcement or adjustment of the materials and shape, they put in shape from the sense of hearing. As a result, generally speaking, the sound from the box was -15dB~25dB to compare with the direct sound from speaker.

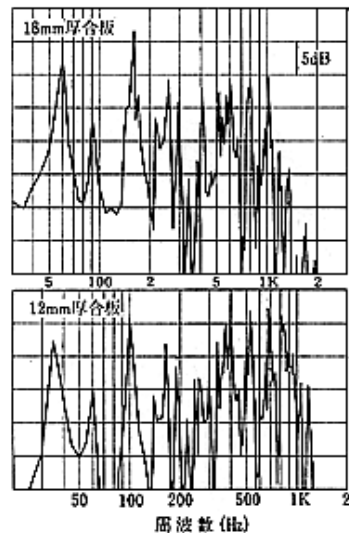
The reverb distortion will be 5~20%, but the distortion exists in the part where the signals had disappeared, so it means more bigger sound in point of hearing sense. If you reinforce, when this side is pushed then that side is up, and energy will not be absorbed so it looks like a continuing seesaw game. Even you can adjust the vibration, it is impossible to get rid of reverb distortion.



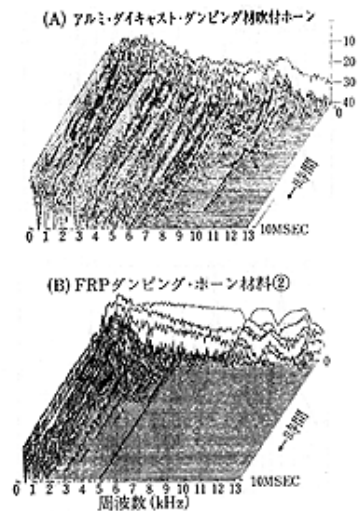
Fig No. 10 shows an example of changing board thickness. It is possible to adjust by shifting frequency response pattern, but this cannot be a fundamental countermeasure also. For super high-faithful sound playback, there will be extra requirements regarding reverb distortion.

We have to find out a new method to absorb vibrating energy, nothing to do with conventional method.

See the Fig. No. 9. (A)(B)(C) are conventional conditions and we can say that (C) is the extreme case. Only a super-maniac can do (C) measure. (D) is the case to consider the absorption of vibrating energy and then put together with damping material which has been developed newly. Still, we can not satisfied although both level and time difference are greatly improved. (E) is the case to add rest ricting material like an iron plate for damping material. This is perfect. We got a hint from the preventing method of vibration for a pole of iron bridge, etc. This is dramatically improved to compare with the conventional data of (A) - (C).



<Fig. No.10> Board vibration of the box. Board thickness will be 1.5 times, also frequency response pattern will deviate 1.5 times toward higher side.



<Fig. No.11> Careful selection of horn material according to accumulated spectrum.

Horn is also prevented from vibration by the same method.

Fig. No. 11 is an example of the improved horn.

We consider it's easy for you to understand that even frequency response( $t=0$ ) is the same in total time base, the transient sound will be different. Accumulated spectrum means something arranged in the three dimensional way which remained after cutting each signal's peak in turn. Consequently, frequency response at  $t=0$  means the frequency response at total time base, not momentary frequency response at  $t=0$ . Accordingly, the element which did not generated at  $t=0$  will be shown in the curve at  $t=0$ . We are likely to misunderstand that the sound was existed at  $t=0$ , so please be careful (Fig. No. 12).



<Fig. No.12> Data results are considerably different according to the difference of time-axis analysis.<

## (2) Absorbing sound in the box

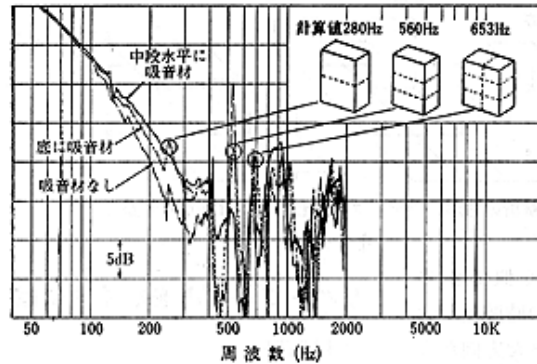
So, we cannot be satisfied with the absorbing sound in a speaker box as it used be.

If you calculate the standing wave mode and frequency, it is well-matched with the measurement value.(Fig No. 13). If you can find out the harmful element and its mode, you can erase it by using absorbing sound material.

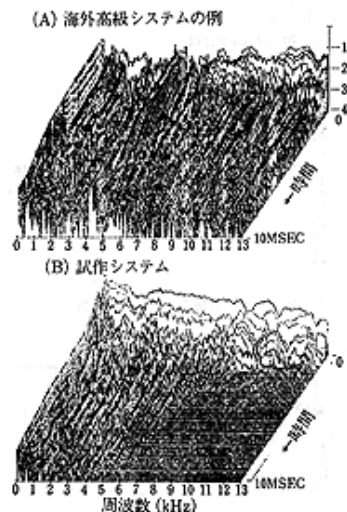
Conventionally, we have been attached absorbing sound material to the inside wall of the box, however this is not effective enough for absorbing sound. If we compare the effectiveness for absorbing sound between two cases, one is pasting one piece of absorbing sound mat to wall and the other is locating in the center (Fig. No. 13), the former case is not effective at all, however the latter is very effective to the mode which position will be a node point.

The sound pressure near the wall is strong, but the air doesn't move, so it is for nothing to put absorbing material on the wall.

There is no sound pressure at the position of the node point on sound pressure, however the air is moving strongly, so it works effectively to the absorbing sound material. To put the absorbing sound materials at the most smallest point which is the node point of standing wave is very rational although it is against the common sense. Of course, to make the box filled with absorbing sound material is the best. Also as an effective method to prevent standing wave, we should adopt one method to make a wall on a slant likewise a listening room.



<Fig.No.13> Air vibration mode inside box and absorbing sound effect.



<Fig.No.14> The difference of reverb distortion.

### The results of new system

The Fig. No.14 shows an example of the system which have been considered regarding like this time domain.

If you listen into music through this kind of system, you will deeply realize how the conventional system is lacking in faithfulness. After all hazy distortion like multipath ghost and reverb distortion have disappeared, so far hidden music will come out clearly.

When it comes to sound playback, we always have been considering that speaker system is a neck, but the story will be different from now on. @You can listen to the sound as it is in case you change a cartridge or change an amplifier. You will clearly realize the difference between digital and analog, also compact disk is still far away from we can call it as "ideal" or "dream". Also we will see the defects of source which has been unidentified until now. According to the same reason, we will improve our thought on the limits of audio after realizing what an excellent music was recorded in a disk. [\(To be continued.\)](#)

技術と理論

ライセンス商品

雑誌掲載技術文献

[NEW APPROACH HOW TO  
REPRODUCE HIGH-  
FAITHFUL SOUND \(1\)](#)  
[NEW APPROACH HOW TO  
REPRODUCE HIGH-  
FAITHFUL SOUND \(2\)](#)  
[NEW APPROACH HOW TO  
REPRODUCE HIGH-  
FAITHFUL SOUND \(3\)](#)

NEW APPROACH HOW TO  
REPRODUCE HIGH-  
FAITHFUL SOUND (4)

TimeDomain Corporation has been in research to looking for good sound since its foundation. One of the basic technological theory is "time domain".

The series of articles were appeared in "Radio Technology" magazine from July to October in 1983.

The concept of "Time domain" is explained very clearly to understand in these articles.

So, please enjoy reading to help your understanding.

Although it had been written more than 19 years ago, its theory and technology hold good even today as long as you consider the change of environmental situation.

We will link new material and information from that time down to this day into the main body additionally without changing the original article.(980904)

## NEW APPROACH HOW TO REPRODUCE HIGH-FAITHFUL SOUND (4)

**What is space distortion in stereo playback**

by Hiroyuki Yoshii

- [Space distortion is also important in stereo playback](#)
- [Competition between SPL and time difference for sound image localization](#)
- [How to think about directivity](#)
- [S/N and the sense of hearing](#)
- [About listening](#)

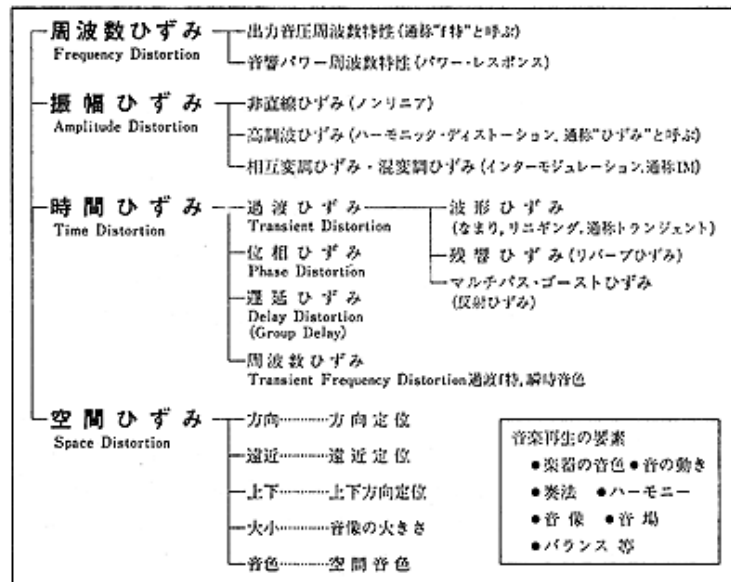
### Space distortion is also important in stereo playback

It has been very hard for us to satisfy with those speakers, one is a speaker with physical property which is not so good at music and made based on conventional physical property, and the other is music based speaker which is on the contrary not depending on physical property at all. We have been aiming real music playback. Firstly, we started from listening music carefully, then review the required conditions for music playback, and audio theory, technique.

It looks like we took quite different course, however probably this supposed to be the course which we should take. Although we cannot say that frequency response and distortion from sine wave were wrong way, those were necessary conditions but not enough conditions.

If frequency response and distortion were ultimate ideal characteristics, namely amplitude/phase perfect flat and harmonic distortion zero, then time domain characteristics will also become ideal. This means even climbing roads are different we will reach the same top. However we cannot reach the ideal. In audio history, there are many cases that the subject we considered no problem turned out to be a critical issue later. We are always on the road to look for the essence of audio. It is very important for us on which road we are now and which road we should choose. We classified harmful distortion for music playback including conventional physical property in case of stereo playback (Table No. 1). There is another way for classification, also another distortions which are not in the list but very common. This table is to explain an outline of the concept of distortion. We try to make it looks as neat as possible since some of them are repeating and similar, and some others are in the relation of cause and effect.

Frequency and amplitude distortion are very popular research subjects, and we often find articles in audio magazines. Most people consider those are well-understood subjects, so on the contrary sometimes we find misunderstanding conclusion and bad effect.



<Table No. 1> Various distortions in the process of stereo playback(Mainly in the speaker system)

There are not so many researches regarding time distortion and space distortion. However, those are very important factors for music playback. Until the last issue, we showed new way of thinking on time distortion and proposed reverb distortion, mutipath distortion and momentary tone, etc.

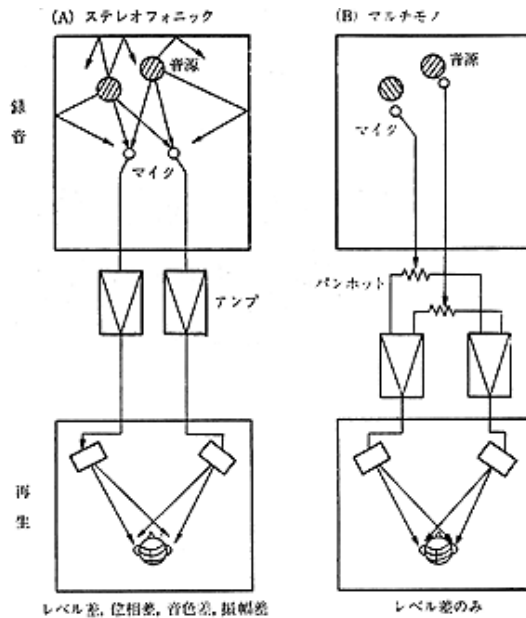
However it's very hard to find an article about space distortion in audio magazines except listening evaluation of good or bad point on localization. Since space distortion will be the most important factor for the advanced expression of music , this time we would like to check carefully about this.

#### Competition between SPL and time difference for sound image localization

Generally speaking, the definition of the localization in conventional stereo is the direction localization in most cases. Those are produced from the sound pressure sound level difference between right and left speaker outputs. In the playback system, there is a balance control according to this principle. And in the recording console table, panpot, which is to determine the direction localization after distributing signals which are recorded in monaural to right and left channels, is the main function.

We cannot satisfy with only like this direction localization from the level difference (Fig. No. 1B). It was O.K. in the conventional system, since it has not enough faithfulness in time domain, however when the faithfulness become high-level likewise our experimental system, we have to consider about localization including phase and position.





<Fig. No. 1> Typical model sample for recording and playback in stereo system.

We will playback the impulse at the same time from right and left using our experimental speaker. If you listen to the sound at the top of an isosceles triangle, the impulse will reach to right and left ears, the sound will be localized in the center. If we delay the right impulse by using a delay device, the sound will be closer to left. It sounds like the sound is coming from the first reached position.

Next, we will put only level difference without time difference. Even you put the level difference like 2dB, 3dB, the sound will come out from the center. This is the value which sound travels clearly in a normal condition, however in this case the impulse will reach to right and left ears at the same time, the sound will come out from the center. If you set the difference to 5dB, 6dB, the sound will go away from one side to the other starting from a certain level. Since only one side sound is possible to hear. Time difference is controlling and looks like nothing to do with level difference.

Then how about in the case of transient signals like music and announce. These are moving from one side to the other side even though they have level difference. But also moving by the time difference. We can get more clear sound image by the time difference than level difference. There are some competition between level difference and time difference. If you make the right signal delayed about 600mS, the sound image will move to the left speaker. If you raise the right speaker level, it will return to the center as it used to be around at 5dB. Since it is the center localization as a result of indicating the contrast position of time and level information, the quality of the localization is not so good. This means, on the contrary, if both information are in harmony with each other, we can get good localization. In stereo playback, it is commonly said that the center localization is bad for female vocal and 6.3kHz band noise, however this speaker will make a high quality localization.

In the localization experiment by transient sound, the level difference localization seems to dominate in case of conventional system which is not so good in time faithfulness.

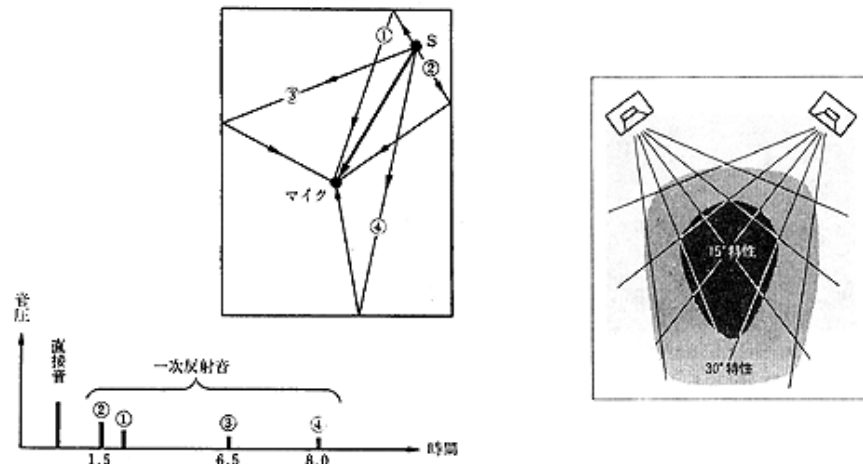
In case of continuous sound, level difference will be dominating. With level and time difference, only direction will be determined based on a theory, however, if you listen to music with good system, you will be able to get touch with depth and sound image. There are various views regarding to recognize perspective shown in Table No. 2. Anyway as these are very advanced and sensitive information, it is very difficult to playback without high-faithfulness.

	near	far
sound level	big	small
tone	vivid	dull
high-level at frequency characteristics	flat	drop
direct sound and reverberation	much direct sound	much reverberation

<Table No. 2> Perspective and physical property

If considering in time domain, it seems like we can playback the sound stage and sound image based on a theory like Fig. No. 2. Conventional research about sound source localization used to be considered with direction information in the frequency domain of head transmission function. On the contrary, Mr. Hiranaka of Tokyo University is considering in time domain of ears-relating response. He has been successful in the localization likewise up and down, right and left, front and back, and seems very reasonable.

If we interfere the wave surface by putting a small obstacle in our experimental horn system, then the sound will be getting closer to conventional sound due to multipath ghost and ghost distortion. At the same time with the deterioration in the quality of sound faithfulness, the localization factor will become worth. We can suppose that this way of thinking is correct, since not direction but perspective, width, reality of sound image and size are spoiled.



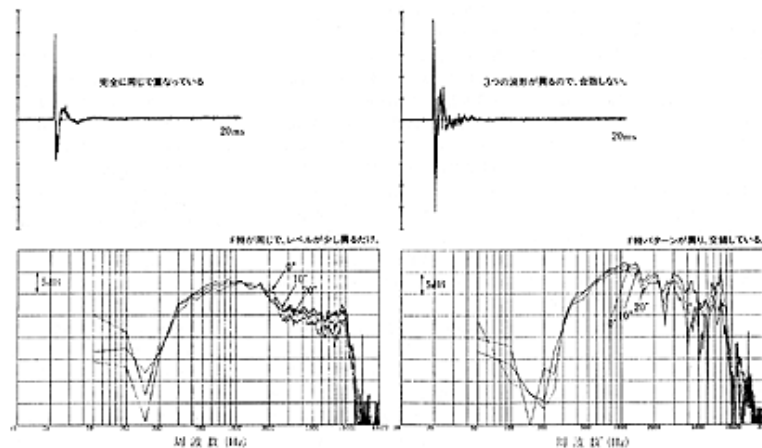
<Fig. No. 2> The condition of recording and perspective simulation. <Fig. No. 3> Directivity as a necessary condition.

### How to think about directivity

There is also some ideal image for directivity. We used to define "good" if it has a wide directivity, however width and good are different subject. It is clear that wider is better naturally for theater and PA, but it's not recommendable to apply to listening room as it is.

See Fig. No. 3. If you look for an excellent sound image localization with time information added, it is not recommendable to move away from the centerline. Regarding the level difference, we can utilize the balance control in case of off the center, but we cannot change the time difference.

Strict faithfulness is required until 15 degree. If we guarantee until 30 degree it will be enough. "Good" directivity means SPL is decayed more than 30 degree without any degradation. The element of more than 60 degree is harmful since it will produce strong first reflection sound which is less time difference at side wall. The same reflection sound is useful in the concert hall, however we should not confuse the situation since the mission and principle are different between concert hall and playback soundstage. This confusion is the case of audio equipments and materials including speaker also indoor sound effect, and we should be very careful. The experimental horn which results is shown in Fig. No. 4A acquired completely equivalent signals in the necessary angle at frequency and time domain. This is necessary condition to playback sensitive tone and real sound image at each moment in the space. After getting rid of multipath ghost distortion and reverb distortion, and controlling radiation angle in the wall design, then acquiring coherent wave.



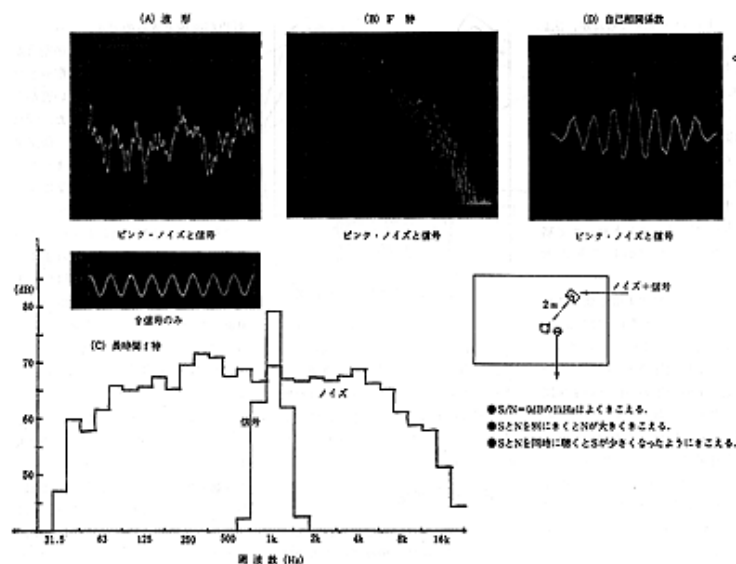
<Fig. No. 4A> Directive property of experimental horn, and <Fig. No. 4B> Directive property of horn made in abroad (with lens) impulse response.

### S/N and the sense of hearing

We can say that good system is superior in sound image localization, and also good at localization for sound image, so it will be possible to playback real sound.

Real sound could be played back since each element assembles to the original space, and no colorization exists. In the part of phase and time distortion in the August issue (New approach how to playback high-faithful sound (2)), we emphasized that correct sound will not be played back at any time if each element of the sound is distributed neglecting time factor even the proportion of elements were correct. It is also the same story for space. If adjacent two instruments sound were mixed, it will be fairly similar sound and color tone numbers will be decreased to compare with real music.

Human being can tell one voice from another among many conversations. We call this as "a cocktail party effect". They can distinguish in the level of S/N 0 dB by using excellent pattern recognition ability. If you could compose the correct elements in time and space, this ability will be improved more.



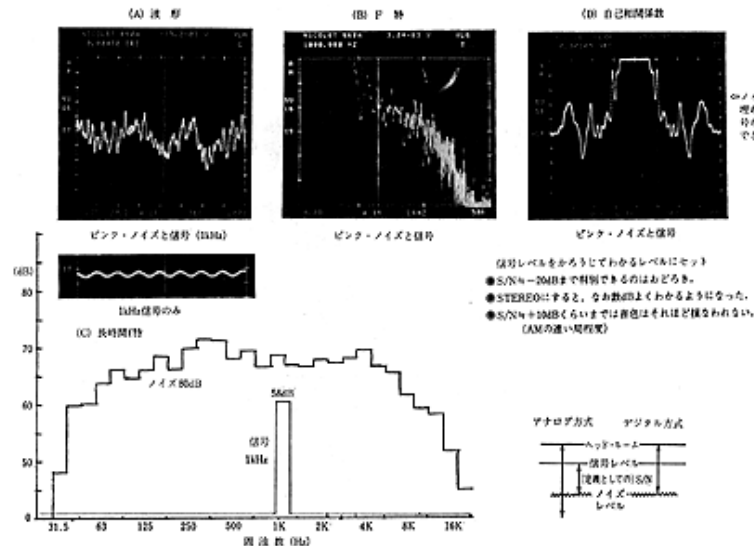
<Fig. No. 5> Various properties of 0 dB sound to compare with SN. With this condition, the contents of music is very clear.

We will show you one experimental example to indicate the difference between measurement and the sense of hearing.

We can make S/N 0dB sound with pink noise and 1kHz sine wave(Fig.No.5A). It is clear that the sound is 1kHz if you listen. Frequency response of short period of 20mS is B. Long period frequency response is C. Noise will be averaged, so it is clear than B. The reason for this maybe the value of S/N10dB around signal elements even though 0dB in case of total elements. Also as indicated in D, signals will be more clear if you take self correlation since the properties are different between signals and element nearby. It looks like the sense of hearing of human being

has like this function.

Then, we will reduce 1 kHz level until ears cannot recognize the sound (Fig. No. 6A). We can recognize until -20dB, the degree which can only tell the genre in music. We can recognize until some more dB worse, if we spread the noise over the space after making it to stereo sound(as all factors of realistic noise are random), and fix the sound at arbitrary position by panpot after making the signals to monaural.



<Fig. No. 6> The conditions seems like hearing possible limit. In case of music, to the extent we can tell the genre.

In case of low faithful system, the story will be different. However in high faithful system, all frequency, amplitude, time, space, will be possible to use until 20-30dB below from the measured value  $S/N = 0$  which has been defined.

This is the reason why we can perceive the sound with wide dynamic range in the new approaching system. Also this is the reason for playback of sensitive timbre, sound image, and necessary reason for the playback of faithfulness in time and space.

In addition to the subject of space timbre, sound image localization is attractive itself. We keenly understand that faithful playback of sound image will be the necessary condition if you can realize the realistic movement at the opera theatre and acting people, and can tell each voice and expression in the chorus, and can distinguish the numbers of strings and each timbre and performance in orchestra.

We will add some information to  $S/N$  issue.

We cannot discuss about the digital system which has no signals below noise level and analog system which has effective signal element below noise level at the equivalent condition. Also according to the same reason, the evaluation will be different depend on quality and level of the instrument which is used for evaluation.

Noise gate will be enough if we only prefer to raise dynamic range on the measurement data and some kind of sense of hearing. In this case, the another dynamic range will be lower in the sense of hearing. This is just for your information to give you a hint on the meaning of the measurement data and what is the sense of hearing.

Note) Noise Gate: A device to cut off the signals which is under a certain level to prevent mixing up the sounds of other instruments in case of multi-monaural recording.

### About listening

This research had started from listening to music and sound.

If you started from measurement and theory, it may have a big possibility to misunderstand the situation. Because there are still many mysterious points in theories and physical properties. However human ears and music have been existing for a long time. Please start to enjoy audio with your ears and your favorite music.

There are many opportunities for listening offered by academic circles and research institutes, and they seemed to have many issues as same as theory and measurement data. As we don't have enough space, I will list up issue points as followings.

1) Need to examine the purpose and condition of listening. It sounds like nonsense to try to get data regarding localization in the system which has bad localization characteristics, and try to get the detection limit in the system which distortion is not so good. Also we have to review the data which supposed to establish the theory long time ago by experiment and the quality of the instruments were not so good to compare with the present one. There are many cases which seem like to turn out different data and results if we have done additional experiment with high faithful system.

2) Systems to collect many panelists and hold a listening and get a result from a pair comparative method are authorized but some of them are not so effective. Professionals should evaluate the quality of high level sound such as other responding detection. In the industry of perfume and wine, professionals are evaluating precisely in high level which is impossible for amateur.

3) We have to use the data regarding limits after checking exactly the level for the purpose. Please understand it is not always suitable to apply average value for the general public as the basic data for equipments specialized to ultra-maniac.

4) We need more concrete and objective expression to evaluate sounds although some words are authorized already. For example, which part of drum is what, and which part of string is how , etc. We can not imagine actual condition of high level sound by using vague expression.

5) We always have to check the difference of quality and colorization to make a judgment on good and bad for audio instruments. In case of faithfulness has arisen, at the same time information volume will increase. In other words, the number of color will increase and it will be possible to tell the difference which used to be difficult to distinguish. It's the same with sound image localization. In case of colorization, information volume remains same or going to decrease even if it sounds favorable. So we can tell the definition of good from this point, if it's favorable from the way of colorization or from the consideration of quality . Finally, we cannot playback high level music without the improvement of quality.

Last, I would like to quote the words of C.J. Lebel from "Psychology of hearing" written by Mr. Souichiro Kuroki. "If it measures good, and sound bad, it is bad."

Please trust your ears and keep looking for the best sound in audio system.