About something an AR series requests. **2017. Aug.**

About 5 years pass after AR-1 is manufactured, and 2 years and a half have been passed from AR-2 planning.

For keeping my original intention, I should order my thoughts from various points of view.

It was 7 years ago that I decided to make an AR system.

It was a trigger that I heard the sound of the ECLIPSE series of Fujitsu Ten.

That gave me a clue of the performance of "reproduction of sound perspective".

There was a sensational loudspeaker system with the visual feature as combination of streamline cabinet with the stand like a pillar.

And it adopts a way of thinking of a ground anchor (virtual ground) and is using a cabinet made of light-weight resin as well.

In 1970s, I knew a way of thinking about a ground anchor for the first time.

That was the article which appeared on "Saburo Egawa laboratory" serialized by Mr. Saburo Egawa, the audio critic who has died in 2015. Mr. Egawa seems to have been obtaining a patent as "the structure to which it's connected mutually at the rear end between 2 units".

There are several use examples of a woofer about this structure. For example G2 GIYA of Vivid audio company. The structure called "RCCM reaction canceling compliant mount".

The sound pressure from both units can be effectively used by the low range with the long wavelength sufficiently to the distance between the units.

However, in an upper frequency band, the mutual interference between two units spoils localization of sound, so it's necessary to separate an acoustic radiation area of the units.

If the acoustic radiation is not separated, peak and dip are made in a frequency response and a phase response.

Therefore I recognize that a consumer product with a full-range unit isn't sold.

To every action there corresponds an equal opposite reaction.

A magnetic-circuit is moved by the reaction of a movement of diaphragm.

And it is perceived as the distortion.

You say "It's just inconceivable!"

That will be simulated below.

I supposed that an actual mass (inertial mass) of the vibration system is 20g, and that a mass of magnetic-circuit is 2kg.

When a diaphragm is moved 1 mm as an action, a magnetic-circuit is moved 10 μm in an opposite direction by reaction.

A frame to fix on a cabinet is vibrated in common mode with a magnetic-circuit. And this vibration is transmitted to a cabinet.

Vibration intensity level of a magnetic circuit and a frame is 1/100 (-40dB) of vibration intensity level of a diaphragm.

The vibration transmitted to a cabinet through a frame is intermodulated by the material of the frame and the shape of the frame.

And the phase of the vibration is reverse to a diaphragm.

So if all of the transmitting vibration was distortion, a rate of distortion is 1%.

I think it's the difficult value to declare that this is the small value.

The general way to protect from distortion is to raise the mass and stiffness of the frame and the cabinet.

A frame and a cabinet can be regarded as uni-structure.

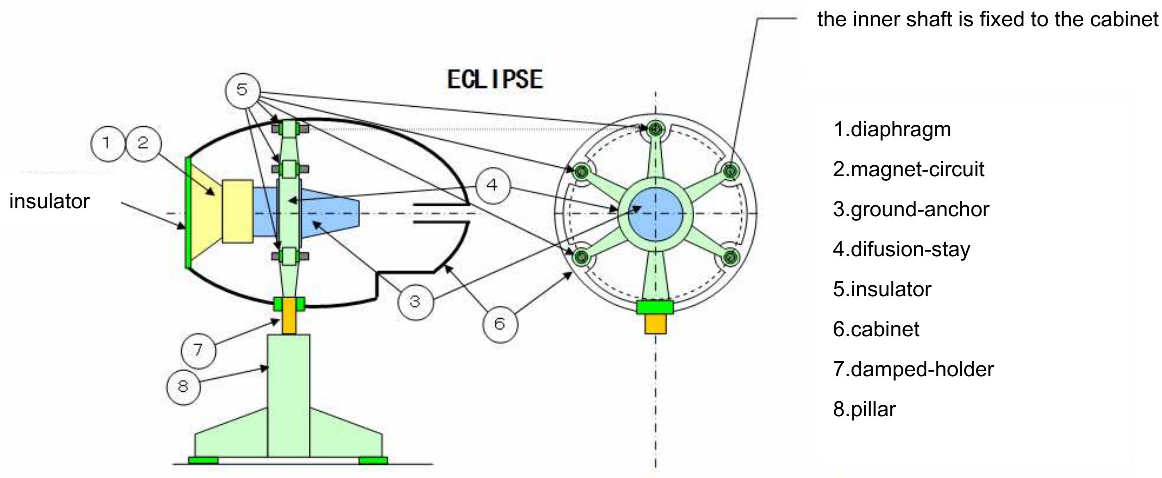
And the acceleration of the vibration can be suppressed small consequently.

The way adopted in ECLIPSE, it's so-called "the unit is floated from a cabinet".

When not making vibration be spread through a cabinet from a frame, the distortion doesn't clarify.

This idea was born by rejecting the stereotype "A frame exists as the means of fixing to a cabinet."

And how is the unit fixed?



In ECLIPSE, a magnetic-circuit is connected to a part like a big artillery shell called "ground anchor".

And the ground anchor has decided this location by a course to a floor via a diffusion stay, response control support structure (dumped holder) and a pillar.

From the direction of a vector in an inertial mass, a magnetic-circuit, a ground anchor and a diffusion stay seems to function as reaction of the vibration system. (Vibration control support structure is the major premise which isn't rigid.)

The structure with which a cabinet is supported through an insulator exists in an edge of the diffusion stay.

This structure is the elaborate structure depended on the ability of the mechanical engineer. It's important to achieve a performance in this part (a performance of insulation) sufficiently.

This is functioning sufficiently, so vibration isn't transmitted.

A cabinet should be a functional part which just decides the capacity, so I think it's OK made of the resin. I think it's OK when a cabinet is made with resin.

In PDF "About transformation to an electric system and the analysis by an analogy method", I indicate how these structures function.

Please refer to it.

In a catalog of ECLIPSE, it is indicated that all products conform to "time-domain theory".

Time domain theory was advocated by Dr. Hiro Yuki Yoshii after A.D. 2000. And that's just a way of thinking by time domain which isn't theory.

I doubted to declare that good sound was made with a flat response by the frequency domain. Therefore I approved a way of thinking of a time-domain.

I have the experience which has designed the unit.

When a frequency response wasn't flat, the audibility was often desirable.

So I sympathized with thought of time-domain.

I'm learning the hearing ability from the side of the physiology recently.

And I understood that hearing emphasizes information on time domain most in the sense for the outward recognition.(in front of the horizontal direction especially)

On the other hand a change in a frequency spectrum becomes important to recognition of the top and the bottom direction and a backward and forward direction.

Time domain and frequency domain are two sides of the same coin.

When doing Fourier transform of information on time domain, it'll be information on frequency domain. And when doing Fourier inverse transform of information on frequency domain, it'll be information on time domain.

There is just a way of thinking to analyze or a tool.

A problem is which domain is easier to analyze.

Anyway, "time-domain" isn't a word of special magic. And I can never say it is just enough to analyze by "the frequency domain".

Of course, the listener who doesn't have a measuring instrument has only the evaluation capacity as the ear.

He has no choice of method. And he decides about the sound between right and wrong by hearing.

Finally, evaluation becomes multifarious by a personal taste and the degree of the pleasant sensation.

While a measuring instruments can deal with only one of information, the hearing is excellent in the point that more than two information can be integrated and judged in the brain. But hearing tends to get tired and be tricked, and it seems to be a measuring instruments that is difficult to handle.

Something I esteem in stereo playback is the atmosphere of a concert hall and a live music club.

If microphone arrangement is "on" at a concert hall, the details of the sound should be reproduced clearly. And if microphone arrangement is "off", it's important to reproduce perspective of the sound accurately.

At the top of a home page, I notify the motto "Pursuit of real sound reproduction".

A stereophonic (2 channel) system is restricted by their own theory and physical limit.

So an engineer draws his playback image in each stage. And the work brought close to the image is carried out.

A recording engineer is making an effort to record the atmosphere (image) of the site using limited machinery and materials. Because the image can't be reproduced when the information isn't in sauce.

A studio engineer who took the work over makes an effort in order to reproduce a site (in his image) to mixing work without processing a source as far as it's possible.

It's because he knows to get away from the reality so that it's done when processing a source.

Aim of AR system

The purpose of a AR system is to realize the image using a possible way in playback stage. I'd like to complete the speaker system which can reproduce the image an engineer wanted to express as faithfully as possible.

Musical reproducibility is too wide, so I understand that reproduction of perspective isn't everything.

Someone has the following way of thinking.

It isn't necessary to participate in a recording process.

And as a result, he'd like to feel comfortable when the sound is playback.

A lot of people seem to hear music only with a headphone.

Though the headphone-unit is located in both ears, can real-perspective be reproduced correctly?

In the genre of the classical music, some persons like an ability to reproduce the strings beautifully, and another persons like a reproduced sound of the wind instrument comfortable.

"Reproduction of perspective" may be so-called complacence.

Which point of reproduced sound is emphasized depends on sensitivity of each listener.

Thus I understand that several people can accept my way of thinking.

Because the request sound of each listener is multifarious.

So something I don't like most is a condemnation and a scolding based on my own idea to opinion and advice.

I check it variously and ask accepted logical technical corroboration for it. But in the long run it may be the one I reach in complacence at the end.

About selection of loudspeaker unit.

A loudspeaker unit uses Lorentz force.

When an electric current is run into the conductor in the fixed magnetic field, Lorentz force is generated in fixed direction.

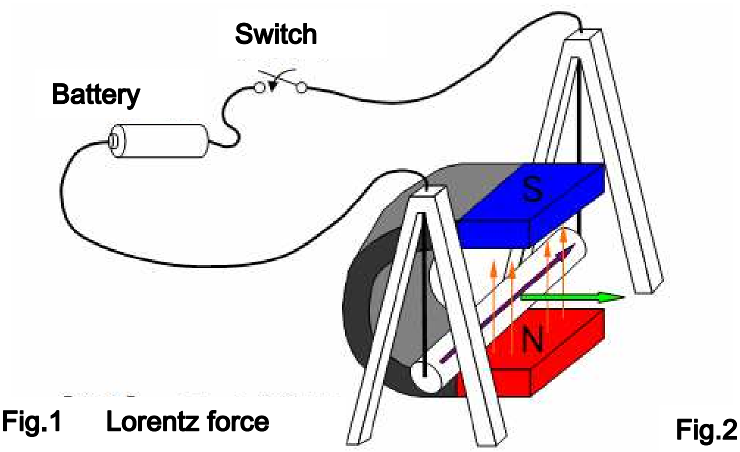


Fig.1 Lorentz force

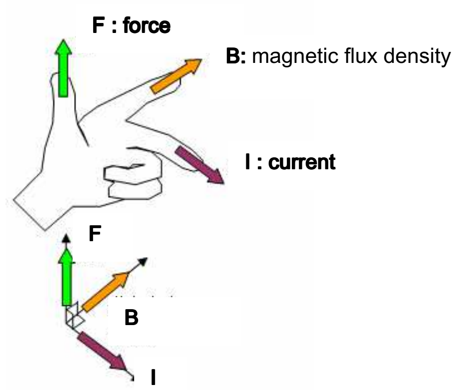


Fig.2 Fleming's rule (left hand)

A swing made of conductor puts between the N pole and the S pole of a magnet.

When the switch is closed and an electric current is running, a swing moves to the way of the green arrow.

Magnetic field line (magnetic flux Φ) exists to the S pole from N pole of a magnet, and Lorentz force F occurs to a swing of the conductor put in field by running electric current I.

Lorentz force F is expressed in following equation.

$$F = B\,I\,l \qquad \cdots \qquad (1)$$

B : magnetic flux density [Wb/m^2 or $N/A \cdot m$ or T (tesla)]

l : The length of the wire across the magnetic flux [m]　　I : current [A]

An electric current is direct current on Figure 1.

When an electric current complies with the audio signal, Lorentz force also complies with the audio signal. When the way of the electric current becomes reverse, the way of Lorentz force also becomes reverse.

It's important for the electric current, the magnetic flux and Lorentz force to be the vector quantity with the way information. Therefore when the direction of the electric current is fixed based on the direction of the magnetic flux, the direction of Lorentz force is fixed uniquely.

In case of the loudspeaker unit, conductor is rolled into cylinder as a voice coil(hereinafter, this is called "VC").

Magnet-circuit consists of top plate, magnet and center pole.

The magnetic flux radiate in a radial direction to S pole as a top plate from N pole as a center pole, and vice versa by the polar setting of magnet.

Please refer fig.3.

When it's made such structure, every part of a coil will turn toward the same direction in Lorentz force F which operates on a coil. (Force F goes to the front direction at all parts of a coil by a top view on figure 3.)

First the loudspeaker unit has to change the audio electric current to the pressure of the compression wave, so a conversion efficiency becomes important.

What should be changed to make Lorentz force F big without changing magnetic flux density B and electric current I?

From formula (1), it's necessary to **make a proportion constant *l* bigger**.

About VC, I defined a diameter as 2r, and total number of turns as N.

$l = 2\pi r N$ ・ ・ ・ (2)

And I defined number of turns per the unit width as n, and winding width of coil as w.

$l = 2\pi r n w$ ・ ・ ・ (3)

$N = n w$

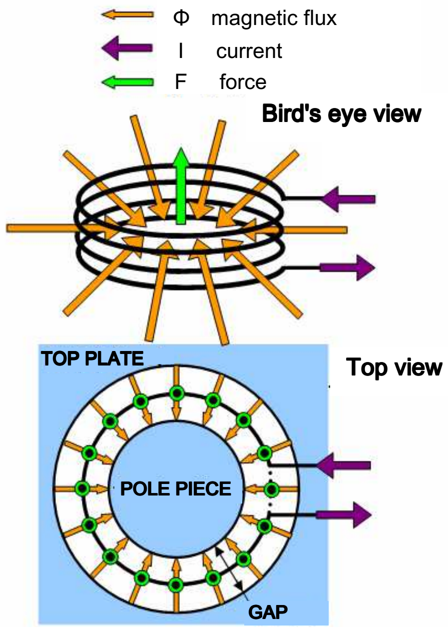


Fig.3 Lorentz force in V.Coil

From formula (3), when the magnetic flux density B is a uniform area, an important matter of a bigger conversion efficiency is that a VC diameter is big, "w" is long, "n" is big.

Actually, a uniform magnetic flux density area is restricted around the gap. Therefore a conversion efficiency is decreased by a long voice coil. The coil part which is sticking out of uniform magnetic field generates heat energy by its resistance.

There is a way to make wire wrap multiple and use a magnetic field efficiently.

When it's put into effect without expanding a gap, the wire diameter becomes thin. The resistance value rises consequently, so an electric current decreases and an electric energy convert to thermal energy as loss.

When expanding a gap, the magnetic flux density falls, and the efficiency falls consequently.

These mean that it's stuck with no way out.

In short it's important to consider the order of priority according to the purpose.

An outline is indicated below.

The unit which takes charge of medium high range doesn't require a big swing width.

Therefore it'll become a short voice coil. An effective mass and a gap width are able to be smaller, so it becomes more efficient.

It's necessary to make low frequency sound including a resonant frequency in case of a woofer. Therefore big swing of a diaphragm is necessary to that.

That's better for using a long voice coil.

For the priority of heat radiation, that's better for being equipped with a big VC diameter using the thick wire diameter and a heat-resistant bobbin.

Please remember a motion equation.

$$F = m a \qquad \cdots \quad (1)$$

m: inertial mass [kg or (g) : CGS]

a: acceleration [m/s]

This is applied to the vibration system of the loudspeaker.

In case of a woofer, an inertial mass of the VC becomes big ...like the above-mentioned.

The diaphragm diameter also becomes big, so the effective mass of the vibration system becomes big consequently. When the mass becomes big under the condition that Lorentz force which occurred to the vibration system doesn't change, the acceleration will be small.

The formula(1) indicates this.

The ways to suppress the delayed distortion are 2 ways, but the vibration system becomes heavy in case of a woofer, so It's necessary to have big Lorentz force to get enough acceleration.

Moreover when the efficiency is considered, I feel like making the magnetic-circuit strong and extending uniform magnetic gap territory.

When the purpose is clear, it can be treated by exclusive design.

When a designer can't narrow the purpose down, a noncommittal incomplete result is derived. A design of a full range unit has to cover a wide frequency band. So as a result, this design is most difficult.

Other elements were excluded and only a conversion efficiency to Lorentz force was considered in the above.

However, back electromotive force E caused by the self-inductance occurs as a result of Lorentz force actually.

It will be a problem that the time-lag distortion of the low frequency range occurs by back electromotive force. It is a problem that the time-lag distortion of the low frequency range occurs by back electromotive force.

To reduce this distortion, the vibration system is light and tough and its vibration convergence has to be good.

Please refer to PDF "dumping factor and back electromotive force" for a in-depth reason.

As far as I find a balance excellent in a full range unit design, Mark Audio is excellent.

Of course, all of the units made by Mark Audio is not in accord with the best condition.

It's necessary to satisfy the following 5 items concretely.

1. Use the CCAW (copper covering aluminium wire) for a VC wire to make the mass of the VC light to the limit.

2. By making the diaphragm the shallow shape, both of the strength and the directivity are satisfied consequently.

3. A diaphragm is made with the paper that has a moderate internal loss.

4. The linear working area is wide so as not to interfere to playback sound in low frequency band.

5. The effective diaphragm diameter is less than 10 cm (if possible, less than 8 cm) to make it do behavior similar to a simple sound source.

These are the reason to select Alpair-6P for AR-1.

F_0 : lowest resonance frequency （Hz）	74.308
Sd : Space of a diaphragm （cm2）	36.320
Vas : Vibration system equivalent capacity (Ltr)	3.696
Cms : mechanical system compliance (m/N)	1.973
Mmd : mass of a diaphragm (g)	2.199
Mms : Vibration system equivalent mass (g)	2.325
Qts : Totalized Q-factor	0.435
X-max : Maximum linear operational range (mm 1 way)	3.3

The equivalent diaphragm diameter is 6.8 cm, Mms is about 2.3 grams and X-max is 3.3mm. The performance is enough for a small full-range driver unit.

Isolation between unit and cabinet

There is a target of an AR system "Turn off its own existence."

While listening to music, I don't like sensing of "a sound is coming out from a loudspeaker system". And the fact that half of the product in a market is similar injures me.

I think that the cause is assumed as follows.

1. The distortion of the unit is big.
2. The intermodulation distortion goes out from a cabinet.
3. Diffraction effect by a baffle of a cabinet.
4. Sound leakage besides the resonance tone from the bass reflex duct.

Item 1 isn't necessary to talk on this.

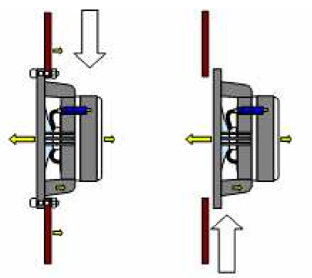
The dissociation between the unit and the cabinet is effective in item 2. (about ECLIPSE) See fig.4

About item 3, it's effective to choose the unit of the wide directivity and to devise the shape of the cabinet for avoiding a diffraction effect. See fig.5.

About item 4, it's a measure to put a downward duct on the cabinet.

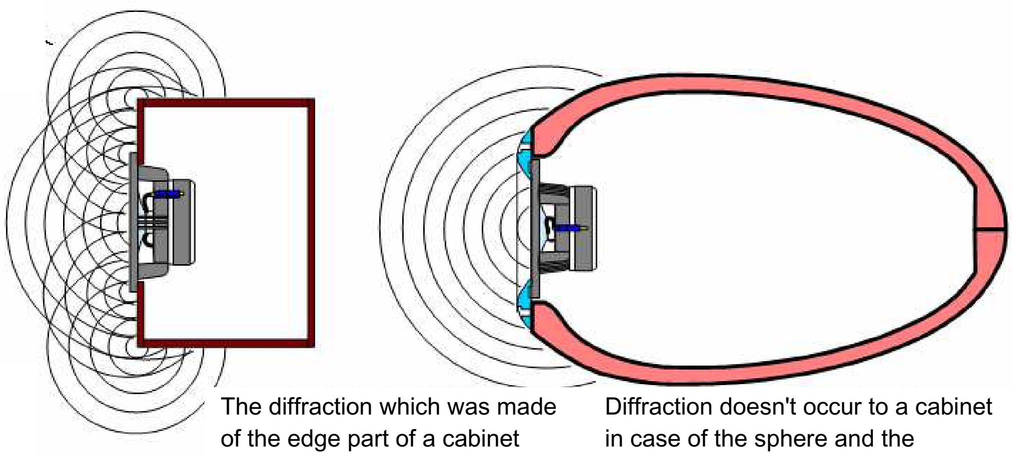
AR-1 showed that influence transcends my assumption.

When a diaphragm moves to the front, a frame and a magnetic-circuit move to reverse by the reaction.



When isolation between the cabinet and the unit is built in, the vibration isn't transmitted to the cabinet any more. It's necessary to prepare the air leakage countermeasure.

Fig.4 Action & Reaction

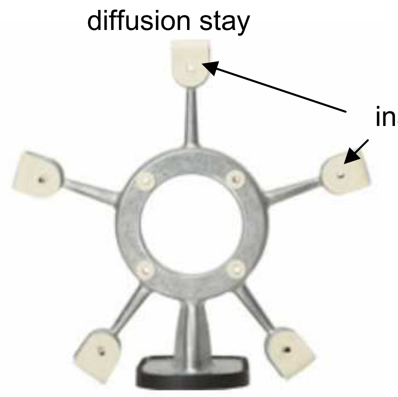


The diffraction which was made of the edge part of a cabinet interfere to the direct radiation.

Diffraction doesn't occur to a cabinet in case of the sphere and the streamline which have no edge.

Fig.5 Wave diffraction

In case of ECLIPSE, a "diffusion stay" keeps a cabinet by 5 absorbers which are located on radiation-line from center, so the physical relationship between the cabinet and the unit are decided. An absorber which also protects an air leakage is inserted into a gap between the rim of a frame of a unit and cabinet.



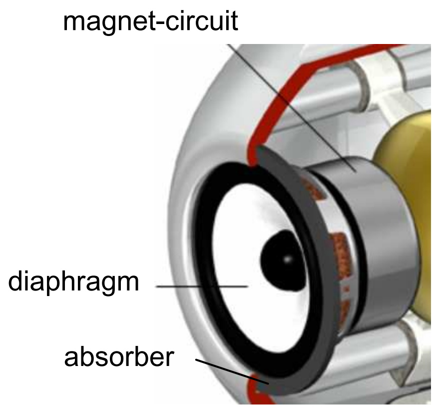


Fig.6 Construction of ECLIPSE

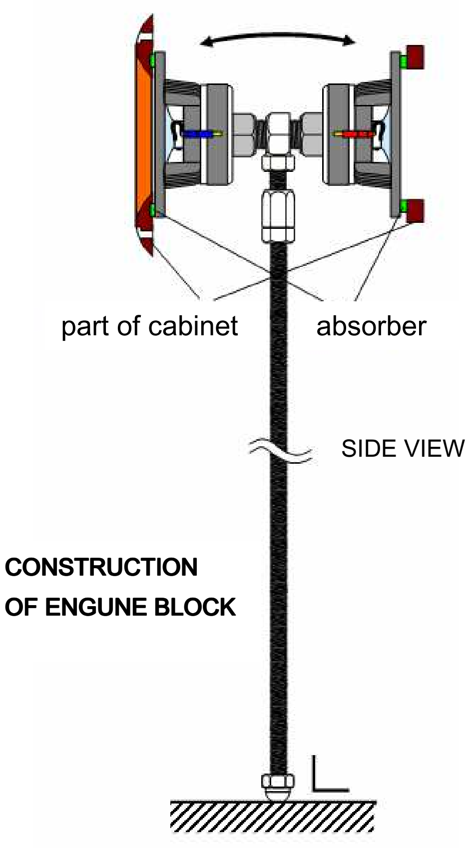
In case of the AR-1, an engine block is constructed of identical two units and "anchor shaft" connects the bottom of each unit.

The role of "anchor shaft (Φ20mm iron)" is canceling a dynamic force of each unit's reaction.

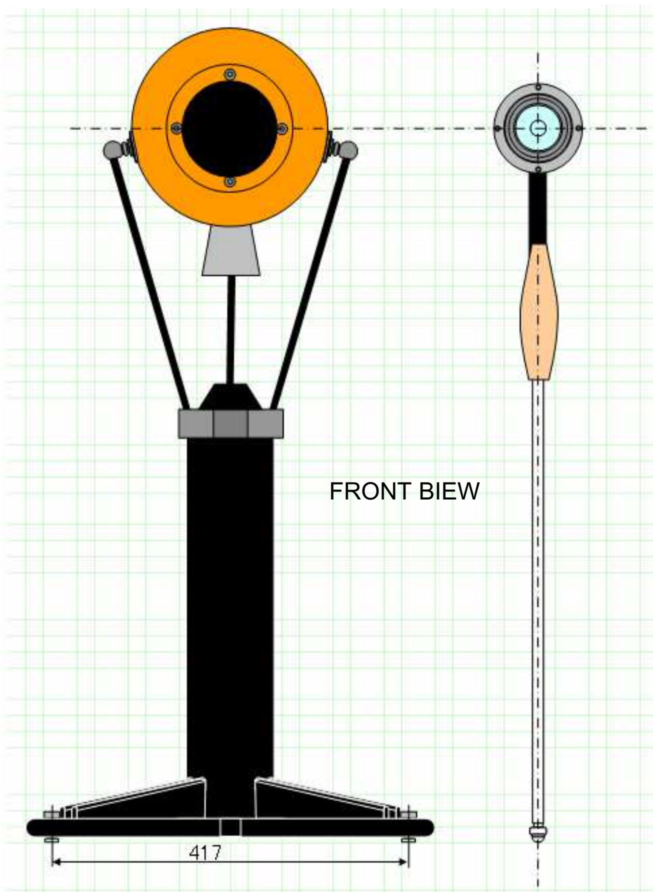
Mechanical center is a dynamic neutral point.

Between the unit and the cabinet, an absorber (Sorbothane) is inserted like ECLIPSE.

Fig.7 Construction of AR-1



CABINET AND ENGUNE BLOCK



When verticalness of a support shaft is maintained correctly and height management between the cabinet and the unit are accomplished correctly, an entropy maximal (stable) position is maintained without adding a stress to an absorber.

Refer to PDF "the AR system outline" for details.

The gravity always works on the midpoint of "anchor shaft".

The leakage vibrational energy by which a vector balance collapsed will be transmitted to the support shaft which stands on the floor vertically without cross modulation.

The vibration transmitted or reflected by a floor is common phase and is added simultaneously in two units via the support shaft.

Therefore behavior is symmetrical in both unit.

The way to hold a cabinet is different in ECLIPSE and AR-1.

AR-1 is different from ECLIPSE where a pillar is common impedance.

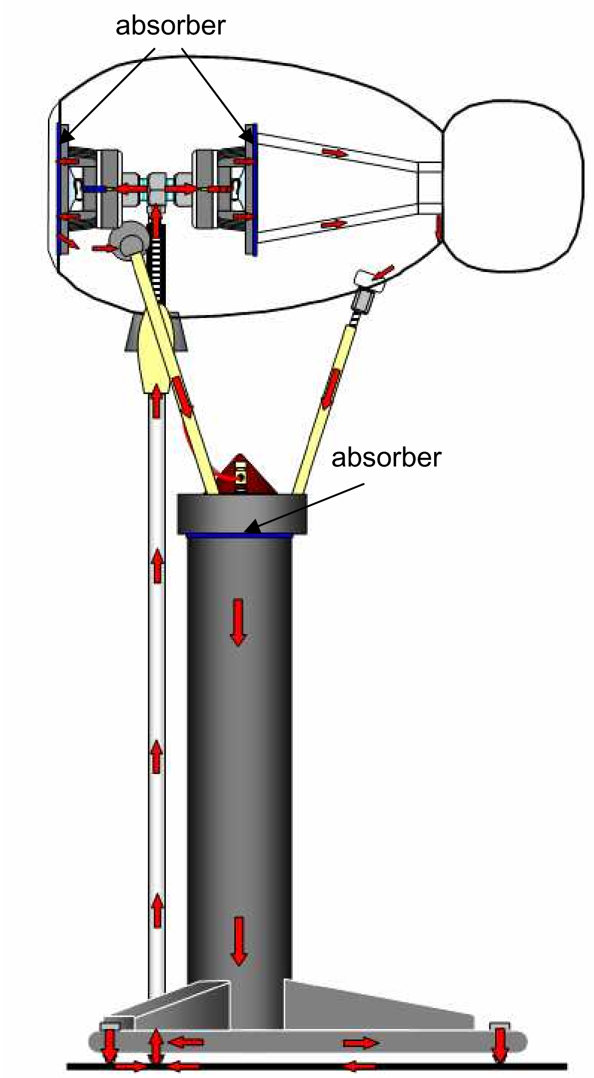


Fig.8 Transmission route of AR-1
Revolving loop

In case of AR-1, cabinet is supported by tri-shaft and the pillar without the other system which the engine is included.

If it is insufficient (in the state that a stress was added to Sorbothane) of isolation between the unit and the cabinet, a cabinet generates the non-linear distortion.

And a big revolving loop including a cabinet is made.

See fig. 7 for further details.

An absorber (Sorbothane) is also inserted during a pillar and tri-shaft holder which supports a cabinet with 3 shafts. When it isn't put in, S/N is aggravated.

I'm not experimenting, but I think a floor is better that its surface is made by the stone and the concrete block or a thick metal plate with big mass put on the earth directly. It's possible to stop a revolving loop so that it may be mentioned above.

It's difficult to achieve such situation, but a floor becomes near a true GND.

Consequently, AR-1 will be more excellent than ECLIPSE of "the structure by which a pillar will be common mechanical impedance". Under such condition, it seems not necessary to put an absorber under the "tri-holder".

It's necessary that the support shaft does not contact the other structure in case of AR-1.

And it's also necessary to put an opening on a cabinet consequently.

The opening is used for a bass-reflex duct. As a duct faced an opening, not straight, it was made the shape which becomes wide. A duct can change the resonant frequency by using a mutual material relation with the rectifier installed in the support shaft.

I couldn't get an expected performance as a result.

By an audition, I found out a distortion which was made with sound's leak from a duct.

That was bigger than assumption.

So I had no other choice but to wind sound-absorbing material around a duct opening.

When I think calmly, it takes the back pressure equivalent to 2 units for a duct, so the leakage distortion becomes high 6 dB (double).

The distortion increases, but the bass reflex effect also becomes this double.

A frequency response wasn't measured, but lack of a low area has not been felt.

Deep bass can't be expected of course, but....

Probrem

A problem was becoming clear by making of AR-1.

When usually using the unit of the plural, it's right to use by parallel connection.

In case of a CD receiver that I possessed, it could drive the units in parallel connection by small power. But when usual power was given to the units, the problem that a protection circuit functions occurred. Alpair-6P : nominal impedance 6 Ω (5.8 Ω), DCR 4 Ω (3.8 Ω)

When the sound quality confirmation by serial connection and parallel connection carried out inevitably, they were a merit and demerit.

The image in the sound quality is as follows roughly.

In case of parallel connection ; hard(attackable) pin-point position

In case of serial connection ; smooth wide 3D-perspective

I'm asking an expression of perspective, and I felt it's good at a series connection.

I thought a serial's playability is better than a parallel's.

For the time being, it was use by a series connection in AR-1.

I compromised on AR-1 including the surrounding circumstances.

However, correct drive isn't made with a series connection theoretically, so I can't accept.

I'll adopt an active loudspeaker system in AR-2 .

It'll be a form that one amplifier output is loaded in one speaker system.

(It'll be 1 against 1.)

The form which drives 2 units by 1 stereo amplifier equipped with L/R2ch amplifier. A stereo amplifier makes them function as a monaural twin amplifier with LR common input.

The next problem is unbalance of a stiffness.

A spatial stiffness in front of the front side unit and that of the back side unit are unbalanced. (the front space of the front side unit is opened, and the front space of the back side unit is closed)

Then the lowest resonant frequency will be different between front side unit and back side unit.

Therefore the canceling movement can not do ideally.

In produced AR-1, I filled an acoustic material with the reverse cone shaped space which is made in front of the rear unit. I expected that a multiple reflection at an inner cone wall and acoustic material in the space is attenuating wave in medium high range. But I found out that sound leakage from an air hole becomes a lot more than assumption. See Fig.8 left side.

The reverse cone shaped space seems to raise the sound pressure. When an ear will be brought near the vent hole, leakage of wind noise like the bellows and reproduced sound of medium high range is heard at the same time.

I noticed that it was necessary to keep enough distance from a wall behind AR-1.

When a system is set near the rear wall, the buzz strikes an ear, and S/N becomes bad.

Improvements are as follows.

① To spread a vent hole; suppress increasing of an acoustic pressure

② To make a chamber in the back; when a sound wave reach the chamber, acoustic pressure was reduced once suddenly

③ To make the route longer; decrease the sound level; (attenuation)

More attenuation in meddle high range can be expected.

④ To use a mechanical sound filter; (silencer)

use a cardboard with air holes (perforated plate) and fill the space with sound-absorbing material

The sound-absorbing material crammed into space in front of the rear unit was reduced to less than half, and then the stiffness was lowered. Addition of the item ① and ②.

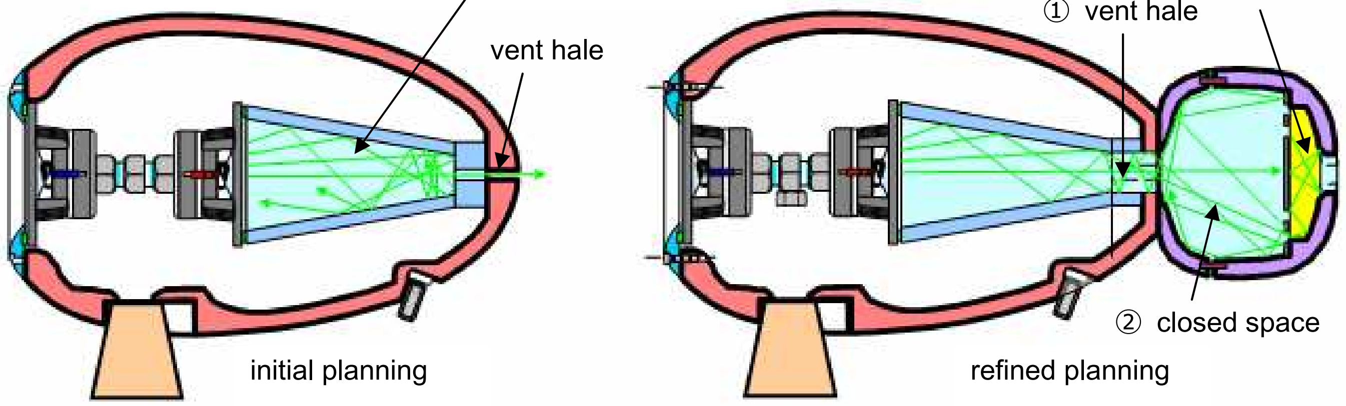


Fig.9 sound attenuation system

When it was auditioned actually, S/N went up compared with remodeling before, and prospects of the depth (perspective) became good.

The ooze which followed about the sound image disappears, and I heard the sound image is tighten.

A resonant frequency of the rear unit also seemed to go down the more or less by the stiffness reduction effect, and the low sound quality from the duct was also improved.

The duct still seems to be functioning by staggered bass reflex. It shows that the improvement degree doesn't have enough level.

Thereby I judged that it was impossible to request any more improvement effect from AR-1.

And there is another problem.

When a rectifier in the duct is moved up and down, the mutual location with the support shaft and the engine changes subtly.

From this, when a stress in the absorber which was inserted between the unit and the cabinet changes, the isolation level changes consequently.

After remodeling, the center of Gravity moves backward, and quality control of the sound becomes very severe in particular.

A cabinet tilts backward inevitably because sorbothane is inserted between a pillar and a tri-holder. A gap between the unit and the cabinet seems to become uniform.

Like "Steelyard" (scale), I experimented to set a metallic ball hanging from the front-end of the cabinet.

Thereby balance is correct, but the Sorbothane between the tri-holder and the pillar was squashed by the weight of the cabinet and the metallic ball.

It would be difficult for itself to control the height of the engine block consequently.

In addition, Sorbothane doesn't work functionally.

I judged this is the limit of the structure of AR-1, and I want to improve in AR-2.

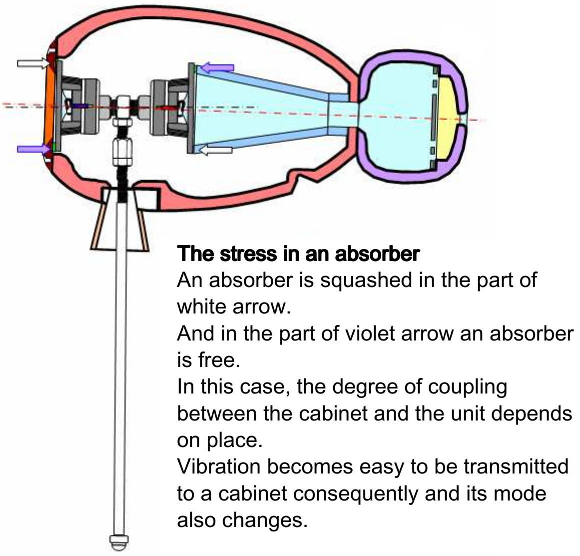


Fig.10 stress in an absorber

