Recent Progress in AFC Technology

—— New ideas for averaging the loop gain ——

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Technology of controlling sound field in an auditorium by electro-acoustic means, AFC (Active Field Control), has recently made a remarkable progress due to the development of time-variant control (TVC) technique. It improves the stability of AFC system to such a degree that the automation of the system adjustment is possible, which has made the AFC quite popular in acoustic design of auditoria. The TVC comprises various ideas such as EMR (Electronic Microphone Rotator) or fluc-FIR (FIR filter with time-variant control), where parameters shift momentarily on time axis (fluc-FIR), or microphones travel virtually in the room space, which both yield a notable averaging effect in the frequency domain smoothing out the loop gain of the system quite elegantly. In the EMR, for instance, the number of averaging samples increases up to N^2 (N:number of independent channels) reducing the irregularity (standard deviation) of the system by $1/\sqrt{N}$. Based on theoretical analysis of the validity of TVC, various examples of applying AFC are introduced including its history of development. Finally it has been shown AFC can be applied even for musical instruments, or for diffusion improvement in a reverberation room.

1. Introduction

As has been known already, technologies of controlling the sound field in auditoria by electro-acoustic means are often called AFC, standing for *active field control* of sound field.[1] They are divided into two categories; (a)frequency-independent system[2] and (b)broad-band system[1][3]. The former is represented by AR[2], and the latter by MCR[3] or AAS[1]. The key of each technology is how stably it can raise the loop gain of the system without bursting into acoustic feedback. That is, the higher is the loop gain of the system increased, the longer the reverberation time (T_{60}) of the room can be extended. Most recently, various ideas of TVC (Time-variant Control), developed for further improving the system stability, have markedly promoted the development of AFC in the construction or renewal projects of auditoria such as concert halls, theaters, churches, etc. Due to extremely high stability obtained, for instance, automation of loop gain adjustment is being realized for practical use.

Also, the environment surrounding the AFC, such as an increase in the construction of auditoria for exclusive use (concert halls, operahouses, or organ domes), progress in signal processing technology, popularization of multimedia, etc., has undergone a great change contributing to "the possibility of electro-acoustic control of room acoustics with the same naturalness as physical (architectural) means". We will here introduce background technologies for controlling the system stably, focusing on EMR (Electronic Microphone Rotator) and fluc-FIR (time-variant FIR),[4)5)] which both form part of TVC, as well as laying out the problems associated with the application of these technologies to auditoria.

Table 1 Definition of three categories developed in the field of AFC (Active Field Control)

Categories		Definition
S-SF	Synthesis of Sound Field	Synthesis of requested condition in a given, or highly absorbent, room
A-SF	Assistance of Sound Field	Control of room acoustic condition based on the given room condition
P-SF	Production of Sound Field	Production of spatial sound effect, mainly for a theatre, movie, etc.

Table 2 Application of A-SF shown with its target spaces

Area	Purpose of Use	Target Spaces
A	Extension of multi-purpose use	multi-purpose/event hall
	Conservation of acoustics of historical facilities	churches/chapels, old theaters
	Avoiding acoustic alomalies of huge spaces, promotion of loudness	domes, multi-event gymnasium
	Improvement of spatial impression	fan-shaped hall, theater
	Extension of reverberance for pipeorgan	concert halls, churches/chapels
	Sound evaluation or music training under various conditions	control room, music education room
B	Diffusion improvement for reverberation room measurement	measurement of absorption/power
	Extension of possibility of electronic musical instruments	electronic organ, piano
	Improvement of sound localization or speech articulation	operahouses, theaters
	Unifying sound field of the room composed of many concave spaces	under-balcony seats, VIP seats

2. Problems for introducing and applying the AFC

The technical field of AFC is usually classified into three categories as in Table 1. According to this classification, the expression "AFC for auditorium" refers to "A-SF (Assistance of Sound Field)" in the table, which is originally derived from the term "Assisted Acoustics" first used by one of the pipeorgan builders in Japan, Mr. Tokugoro Ohbayashi, in his private letter for the author. The author has thus reorganized the information on AFC to avoid the confusion with the term ANC (Active Noise Control) which is exclusively for noise reduction, or misuse of the term A-SF, such as "A-SF using electro-acoustics". One misunderstanding which we often encounter in its implementation is the confusion with reverberation enhancement that refers to adding reverberation generated in a separate process.[1)]

To include the microphone, speaker and the room itself in the feedback loop gives the A-SF a different sense in a way that links with natural extension of reverberation or reasonable increase of reflections. It is now within common sense that the musicians, apart from the argument on their allergy against the use of electricity, take a positive attitudes toward the use of the technology as far as it yields the better result regardless whether it is based on electro-acoustics or architectural acoustics. It is also true that the collaboration with interior designers is necessary in the planning phase for AFC system as part of the architecture.

On the other hand, A-SF has two areas; one in which it has already been put to practical use (Table-2A) and one which is still in the research phase (Table-2B). For example, the last item in (B) refers to reradiating the sound, picked up in the space over the major audience seat, toward the under-balcony seats so that they are virtually integrated into a single space. Examples and details of actual implementation of A-SF are shown in the reference[1)3)6)9)].

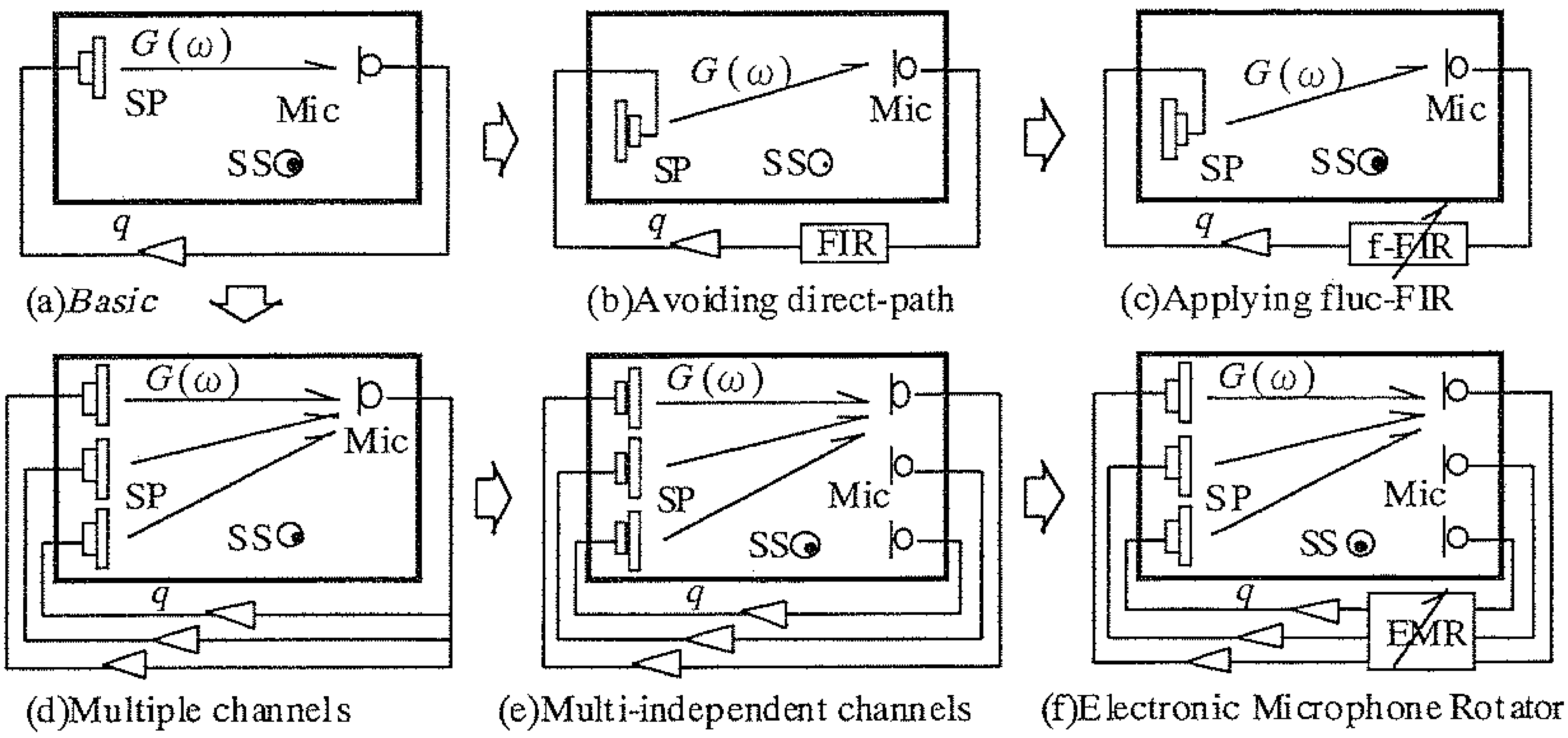
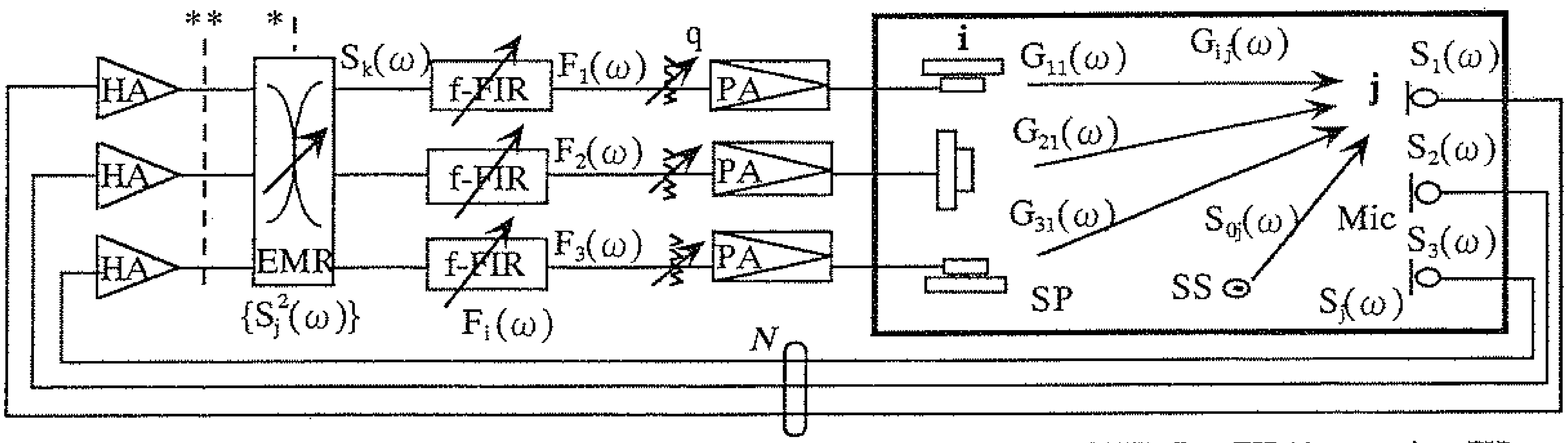


Fig.1 Control model for A-SF and history of its developement



*HA:Mic Amp. *PA:Power Amp. *EMR:Electronic Microphone Rotater *f-FIR:fluc-FIR/time-variant FIR

Fig.2 Model for the investigation of the possibility of smoothing and raising the loop-gain

3. Control models for A-SF

Common to all A-SF systems is that they take advantage of feedback loop containing both microphones and speakers (hereinafter, denoted by "Mic" and "SP"). Equation (1) below indicates how, by somehow increasing the diffuse sound energy density E_r, the reverberation time T_{60} can be extended:

$$E_r = (1-\overline{\alpha})W T_{60}/(13.8V) \qquad (1)$$

Table 3 Distribution pattern in EMR (N=3)

*Each input j is connected momentarily to one of N output channels. (N=3, here)

t	t_1	t_2	t_3	t_4	t_5	t_6
j=1	1	1	2	2	3	3
2	2	3	1	3	1	2
3	3	2	3	1	2	1

Table 4 Comparison of various averaging schemes

Method	Eq.	2'nd term in dominator	Virtual number of channels	Standard deviation	Averaging over
(conventional)	(12)	$[\Sigma_i G_i(\omega)]^2$	(N))	1	sound pressure
Multi-independent	(10)	$\Sigma_i \overline{G_i^2(\omega)}$	N	$1/\sqrt{N}$	squared pressure
EMR	(8)	$\Sigma_j \Sigma_i \overline{G_{ij}^2(\omega}$	N^2	$1/N$	squared pressure

,where W stands for the sound source power, V for the room volume, and $\overline{\alpha}$ for the average absorption coefficient. Thus, the essence of the technology is how to raise the "loop power gain β (shown below)" of each system stably without causing the recirculation, i.e., how to even out β on the frequency axis. This is equivalent to reducing the $\overline{\alpha}$ of the room, which enables not only the extension of reverberation decay but also the natural enhancement of perceived room acoustic conditions such as loudness or reverberance. In the past, the loop-gain control technologies for A-SF underwent innovations in the time domain as shown in (a) ~ (c) in Fig.1, as well as the averaging in the spatial domain as in (d) ~ (f). Finally, the structure of the current system for practical use has evolved into that shown in Fig.2. In the "TVC" technique included, such as Fluc-FIR in (c) or EMR in (f) of Fig.1, parameters change "continuously but very quasi-statically" to avoid both instability of the system and the aural anomalies:

(c) fluc-FIR

- Each element of FIR filter corresponding to each reflection moves gradually on the time axis (keeping the envelope constant)
- The velocity and the depth of the change is limited within the range of perceived naturalness (specifically for the field of music performance)

(f) EMR

- At any single point $t = t_i$ each input is connected with one of the output channels (each output is always independent)
- Shifting from t_i to t_{i+1} takes place smoothly, accompanied by amplitude mod (distribution ratio is constant with each input/output)

That is, the distribution of input signal occurs ${}_N P_N = N!$ times at random and with the same probability as in Table 3, which then leads to (at any $j=k$ channel):

$$S_k(\omega):\quad \overline{S_k^2(\omega)} = \{S_j^2(\omega)\} = (1/N)\,\Sigma_j\, \overline{S_j^2(\omega)} \qquad (2)$$

In order to examine the effects of EMR, above-mentioned β was defined and computed for the system in Fig.2. As in Eq.(3) it is equal to the ratio of E_r in Eq.(1), each of which pertains to the system on and off, and finally to that of reverberation times.

$$\beta \equiv \overline{S_j^2(\omega)} \diagup \overline{S_0^2(\omega)} = E_{r1} \diagup E_{r0} = T_1 \diagup T_0 \tag{3}$$

Now, with the expression of N, the number of channels, as well as $G_{ij}(\omega)$, transfer function between $(\mathrm{Mic})_i$ and $(\mathrm{SP})_j$, the response of j-th receiving point for sound source SS is written as follows:

$$S_j(\omega) = S_{0j}(\omega) + q\,\Sigma_i S_k(\omega) F_i(\omega) G_{ij}(\omega) \tag{4}$$

Next, by averaging the squared responses of both sides, the equivalent power $\overline{S_j(\omega)}$ is calculated as follows. Taking it into account that each component is nearly equal and mutually independent, that the integral of the product of diagonal functions is zero, and that the average of the products of mutually independent functions correspond with the product of their averages (i.e., $F_i \neq F_j$, $S_i \neq S_j$, $G_{ki} \neq G_{kj}$ and $\overline{F_i^{\,2}} = \overline{F_j^2}$, $\overline{S_i^2} = \overline{S_j^2}$, $\overline{G_{ki}^{\,2}} = \overline{G_{kj}^{\,2}}$, $\overline{S_{0j}^{\,2}} \equiv \overline{S_0^2}$ (constant), $\Sigma_j \overline{G_{ij}\,G_{i'j}} = 0$, etc., $\Sigma_i \overline{S_k^2 F_j^2 G_{ij}^{\,2}} = \overline{S_k^2}\,\overline{F_j^2}\,\Sigma_i \overline{G_{ij}^{\,2}}$, etc.), we obtain from Eq.(2):

$$\overline{S_j^2(\omega)} = \overline{S_0^2(\omega)} + (q^2/N)\,[\Sigma_j \overline{S_j^2(\omega)}]\,\overline{F_j^2(\omega)}\,\Sigma_i \overline{G_{ij}^2(\omega)} \tag{5}$$

If we set up N equations (for j =1,2,...N) similar to Eq.(5) and sum up both sides to obtain Eq.(6), then Eq.(7) is finally obtained by substituting it into Eq.(5).

$$\Sigma_j \overline{S_j^2(\omega)} = N\,\overline{S_0^2(\omega)} \diagup \{1 - (q^2/N)\,\Sigma_j [\overline{F_j^2(\omega)}\,\Sigma_i \overline{G_{ij}^2(\omega)}]\} \tag{6}$$

$$\overline{S_j^2(\omega)} = \overline{S_0^2(\omega)}\,[1 + q^2\,\overline{F_j^2(\omega)}\,\Sigma_i \overline{G_{ij}^{\,2}(\omega)}\,] \diagup [1 - (q^2/N)\,\overline{F_j^2(\omega)}\,\Sigma_j\Sigma_i \overline{G_{ij}^{\,2}(\omega)}] \tag{7}$$

Since the the system is usually operated around the region where the denominators of the following equation, i.e., the first two terms in the numerators are nearly zero, β is ultimately calculated as:

$$\begin{aligned}\beta &= \{1 - (q^2/N)\,\overline{F_j^2(\omega)}\,[\Sigma_j\Sigma_i \overline{G_{ij}^{\,2}(\omega)}] + q^2\,\overline{F_j^2(\omega)}\,\Sigma_i \overline{G_{ij}^2(\omega)}\,\} \\ &\qquad \diagup [1 - (q^2/N)\,\overline{F_j^2(\omega)}\,\Sigma_j\Sigma_i \overline{G_{ij}^2(\omega)}] \\ &\fallingdotseq q^2\,\overline{F_j^2(\omega)}\,\Sigma_i \overline{G_{ij}^2(\omega)} \diagup [1 - (q^2/N)\,\overline{F_j^2(\omega)}\,\Sigma_j\Sigma_i \overline{G_{ij}^2(\omega)}]\end{aligned} \tag{8}$$

Consequently, the stability of the system is determined by the flatness of the two terms of the denominator, $\overline{F_j^2(\omega)}$ and $\Sigma_j\Sigma_i \overline{G_{ij}^2(\omega)}$ (the number of samples increases up to N^2), on the frequency axis.

4. Methods for avoiding the feedback

Based on the above, let us examine the ideas illustrated in Fig.1, each developed subsequent to Assisted Resonance.[2] The key to A-SF technologies are summarized into these three points: (1) stability (against the feedback), (2) controllability (assuring the range of operation) and (3) naturalness (to the degree of imperceptibility of the system in operation). Especially, item

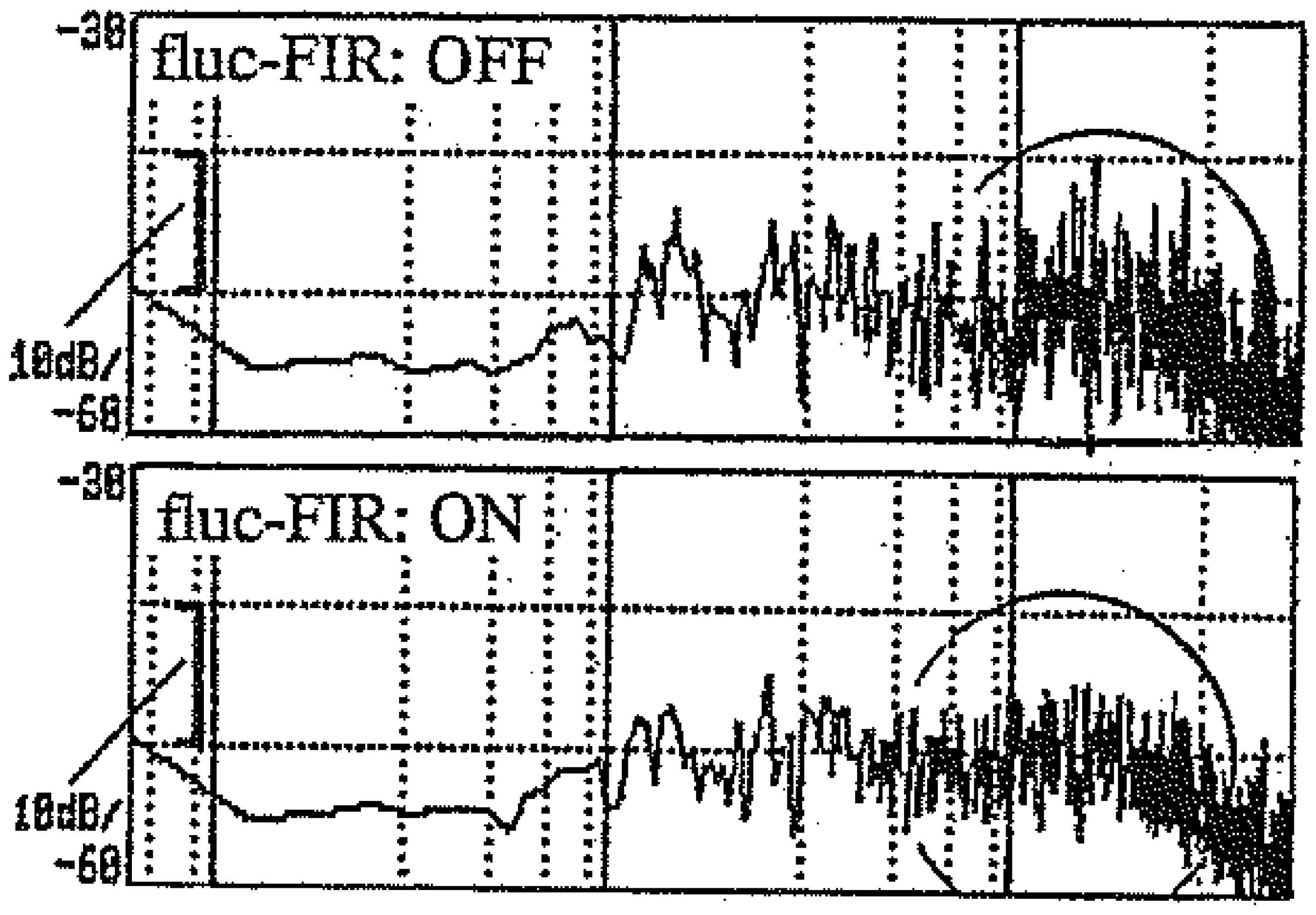


Fig.3 Averaging effect of fluc-FIR on β

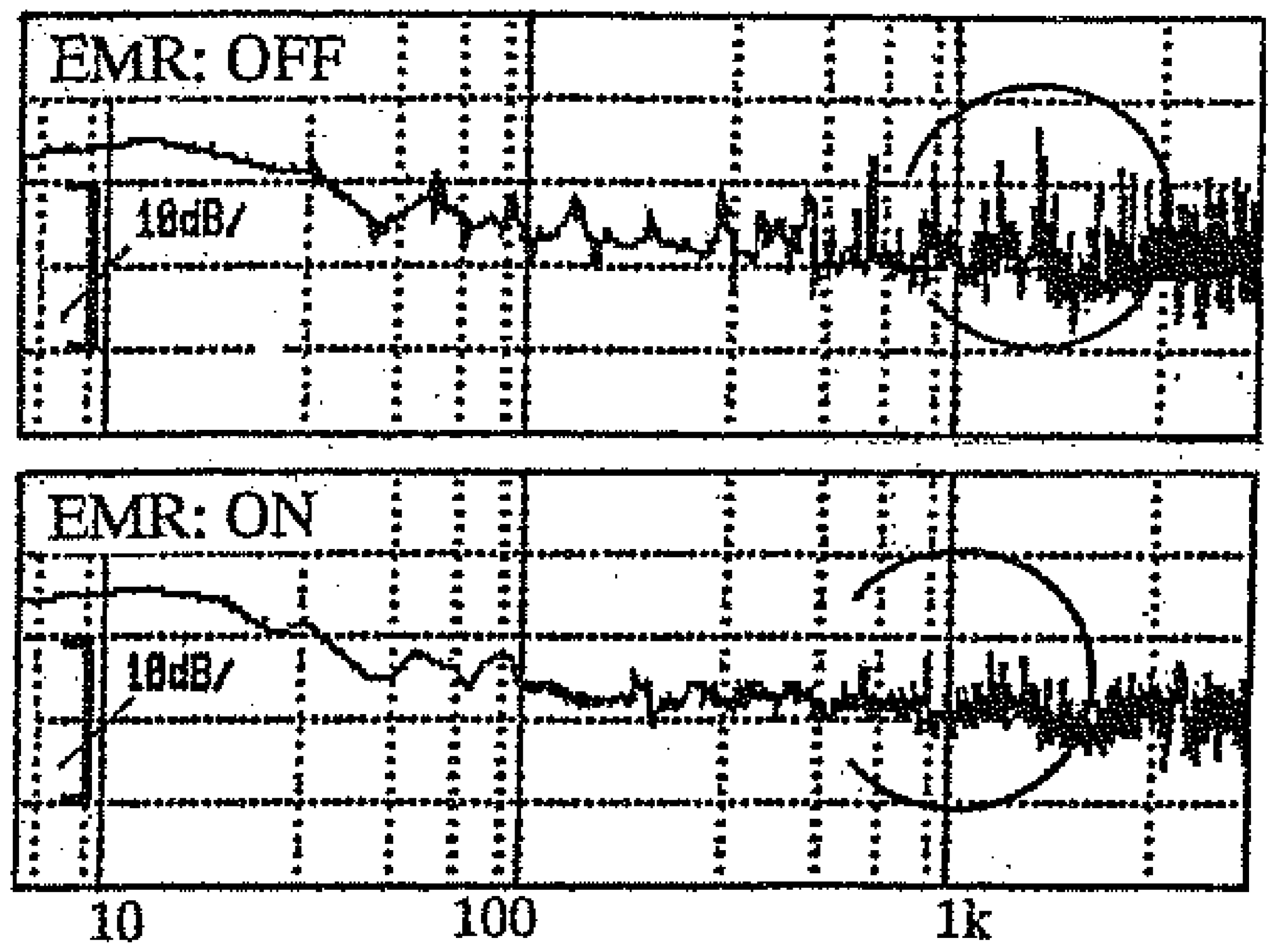


Fig.4 Averaging effect of EMR on β

(1), overcoming the acoustic feedback, dictates the other two.

4.1 Eliminating the direct path ---Fig.1(b)

As shown in Fig.1(b), the speaker axis can be slipped away from the direction of microphone, or speaker itself can be placed in a concave spot or cavity-like place constructed on the room boundary, each so as to avoid the direct path for easily causing the comb filter effect.[1] In general, this yields the improvement of smoothing out the transmission frequency characteristics of the room as much as the reduction of peak factor 10 -12 dB.

4.2 Introducing the FIR Filter[6] ---Fig.1(b)

Furthermore, by inserting an FIR filter (abbreviated hereinafter to "FIR") in the loop aiming at the following two points, considerable improvement in the frequency characteristics is obtained compared to a singly-delayed system.[6] However, due to the finite number of FIR elements and the initial reflection structure (aural unnaturalness), there is a limit to the control.

- avoiding comb filter effects inherent in a singly-delayed system, thus improving the frequency characteristics.
- controlling the reflection structure (such as "room size", etc.) and the early reflection index

4.3 Introducing time-variant FIR ("fluc-FIR")[4][5] ---Fig.1(c)

The above -mentioned limitation can be removed by shifting each of the FIR elements on the time axis (TVC technique) due to the averaging effects on the frequency axis. To control the system "faster than the growth of the peak of instability on the frequency axis but slower than the limit of hearing naturalness for musical sound", various ideas are required such as modulating each element of FIR with incoherent signals. Figures 3 and 4 show examples of loop gains observed in a sound evaluation room of 84 m^3 using the system in Fig.2, which demonstrate remarkable reduction of peak factor due to the use of fluc-FIR.

4.4 Employment of many independent channels[3] ---Fig.1(e)

Meanwhile, preparing many independent closed channels as in (e), which are mutually disconnected, new system is obtained that is similar to the one in Fig.2 but without EMR located at "*" in the figure. The response at j and loop power gain, $S_j(t)$ and β, respectively, will be obtained as follows by the same procedure as Eqs.(5) and (8).[1)3)]

$$S_j(\omega)=S_0(\omega)+q\ \Sigma_i S_i(\omega)F_i(\omega)G_i(\omega) \tag{9}$$

$$\beta=1\ /[1-q^2\overline{F_i^2(\omega)}\ \Sigma_i\overline{G_i^2(\omega)}] \tag{10}$$

On the other hand, a conventional system (d), which does not meet "many independent channels" condition, is equivalent to the system produced by unifying all the channels at "**" in Fig.2. Although each channel following FIR appears to be independent, β is calculated as that for single channel:

$$S(\omega)=S_0(\omega)+q\ S(\omega)\Sigma_i F_i(\omega)G_i(\omega) \tag{11}$$

$$\beta=1\ /\{1-q^2\ \overline{[\Sigma_i F_i(\omega)G_i(\omega)]^2}\} \tag{12}$$

In effect, while Eq.(12) comprises sum of sound pressures, Eq.(10) exhibits sum of <u>squared</u> sound pressures, and thus the irregularity (standard deviation) relating to the portion of $G_i(\omega)$ is reduced by $1/\sqrt{N}$, all due to the employment of many independent channels.

4.5 Introduction of Electronic Microphone Rotator (EMR) [4)5)7)] ---Fig.1(f)

If $\overline{F_j^2(\omega)}$ in the denominator of Eq.(8) is smoothed out or evened, then the $\Sigma_j\Sigma_i\ \overline{G_{ij}^2(\omega)}$ portion becomes significant. To even the transfer function of the entire system requires an alternative means. The EMR differs from simple spatial averaging or continuous averaging, in the way it switches the input j (discrete averaging) moment by moment as in Table 3, creating N (i.e., N^2 in all) independent input signals of $G_{ij}^2(\omega)$.[7)] Consequently, due to the process of $\Sigma_j\Sigma_i$ in Eq.(8), the number of averaging samples in the denominator increases up to N^2, and then the irregularity is reduced to $1/N$, yielding further averaging effect as shown in Fig.4. These ideas are summarized as follows and in Table 4:

- addition of squared sound pressures over "many independent channels" produces a reduction of irregularity by $1/\sqrt{N}$:

  $\overline{[\Sigma_i\ G_i(\omega)]^2}$ in Eq.(12) $\rightarrow\Sigma_i\ \overline{G_i^2(\omega)}$) in Eq.(10)

- the EMR operates to increase the number of averaging samples to N^2, and to reduce the irregularity by 1/N:

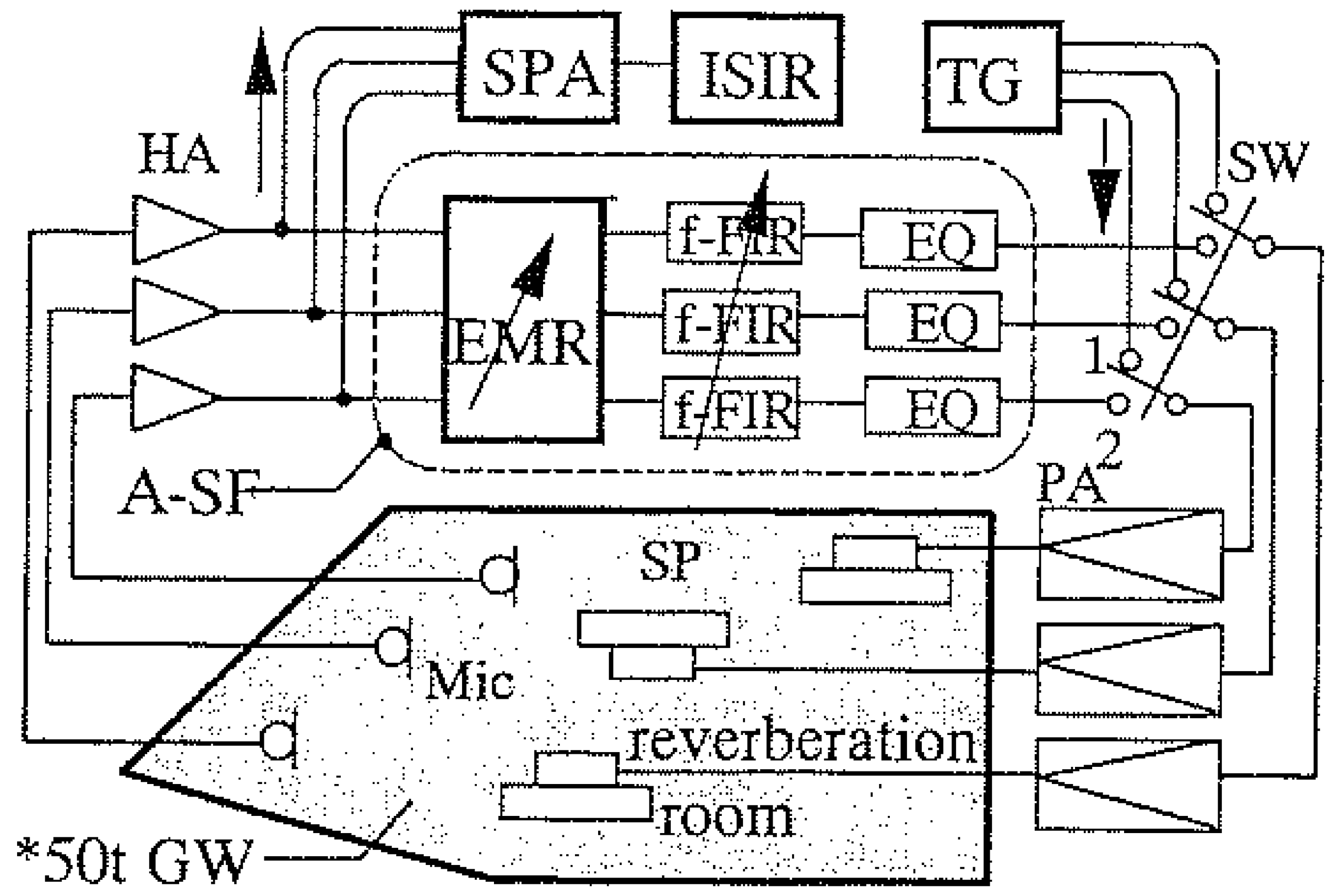
  $\Sigma_i\ \overline{G_i^2(\omega)}$ in Eq.(10) $\rightarrow\Sigma_j\Sigma_i\ \overline{G_{ij}^2(\omega)}$ in Eq.(8)

- the EMR function of "momentarily traveling over the space" improves the diffusion condition of the room

  : a specific effect similar to that given by the momentary change in the room boundary conditions such as a rotating vane [8)], as will be stated in the next chapter.

5. New possibilities of A-SF

One example of time-variant AS-F system developed by the author, AAS-II, comprises such functions as multiple independent channels (N=4), elimination of direct path, fluc-FIR, EMR, etc., yielding ultimate stability so as to enable the automation of loop gain adjustment. In addition to the application to auditoria, it may be considered for various other applications such as marketed for the use in small spaces[9].



SPA:Squared Pressure Averager TG:Tone Generator
ISIR:Integrated Squared Impulse Response Processor

5.1 Application for improving the diffusion

First of all, functions such as EMR are effective in improving diffusion. EMR does not perform simple squared pressure averaging, but like a rotating vane[9] which momentarily shifts the input signals to each channel. The system shown in Fig.5 was brought into a reverberation room (273m^3 with entire floor covered with 50 mm thick fiber-glass) and operated with the same number of channels as in AAS-II, i.e., with N=4 (the figure illustrates the case of N=3, though), to observe the space-ensemble average of reverberation decay curve $\{<S^2(t)>\}_{NO}$.[9] The notable curvature on decay record "a_0" caused by the excessive concentration of absorption to the the floor does not change merely by inserting the time-variant A-SF portion as in "a". If the switches ("SW" in the figure) are all closed at once after the sound source stops, however, the feedback loop is activated during the decaying process and the linearity of the decay record, along with the extension of reverberation itself, is significantly improved as shown in "b" with apparent reduction of Q_{30} (CI:Curvature Index, defined as the ratio in dB of decay rate at -30 dB attenuation of the decay to the initial decay rate)[9] from 2.2 to 0.5 at 250 Hz, or 2.1 to 1.1 at 2 kHz.

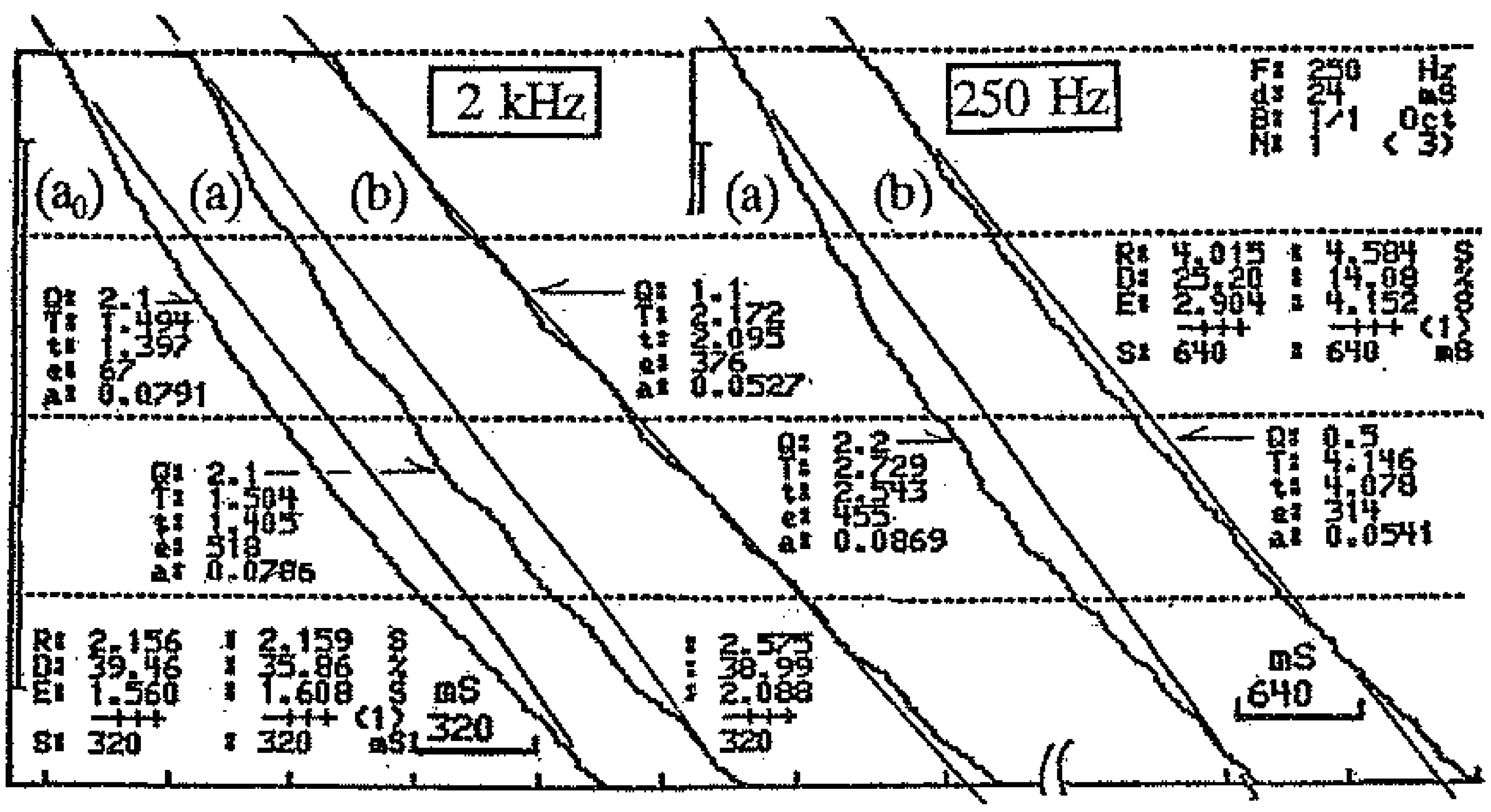


Fig.5 Improvement of diffusion by time-variant A-SF in (b)

5.2 Sound field control of small room spaces

The overall effect obtained by the combination of the individual methods is illustrated in Fig.6. It is an actual example implemented in an audio evaluation studio (28 m^2 × 3 mH), where the time-variant system with EMR and fluc-FIR has extended reverberation time stably as much as by ten times as in (b), which would be most conventionally limited, as in (a), at most to twice as much in those without TVC. Also, the decay curve itself also exhibits, compared to a conventional decay curve in (A), a smooth and straight line as in (B) over the entire decay process. Thus, A-SF system provided with these TVC techniques can be applied even to small room spaces where it is usually impractical to control them stably due to excessively high reflection density, or to extremely wide range of operation requested because of the size of the room,.

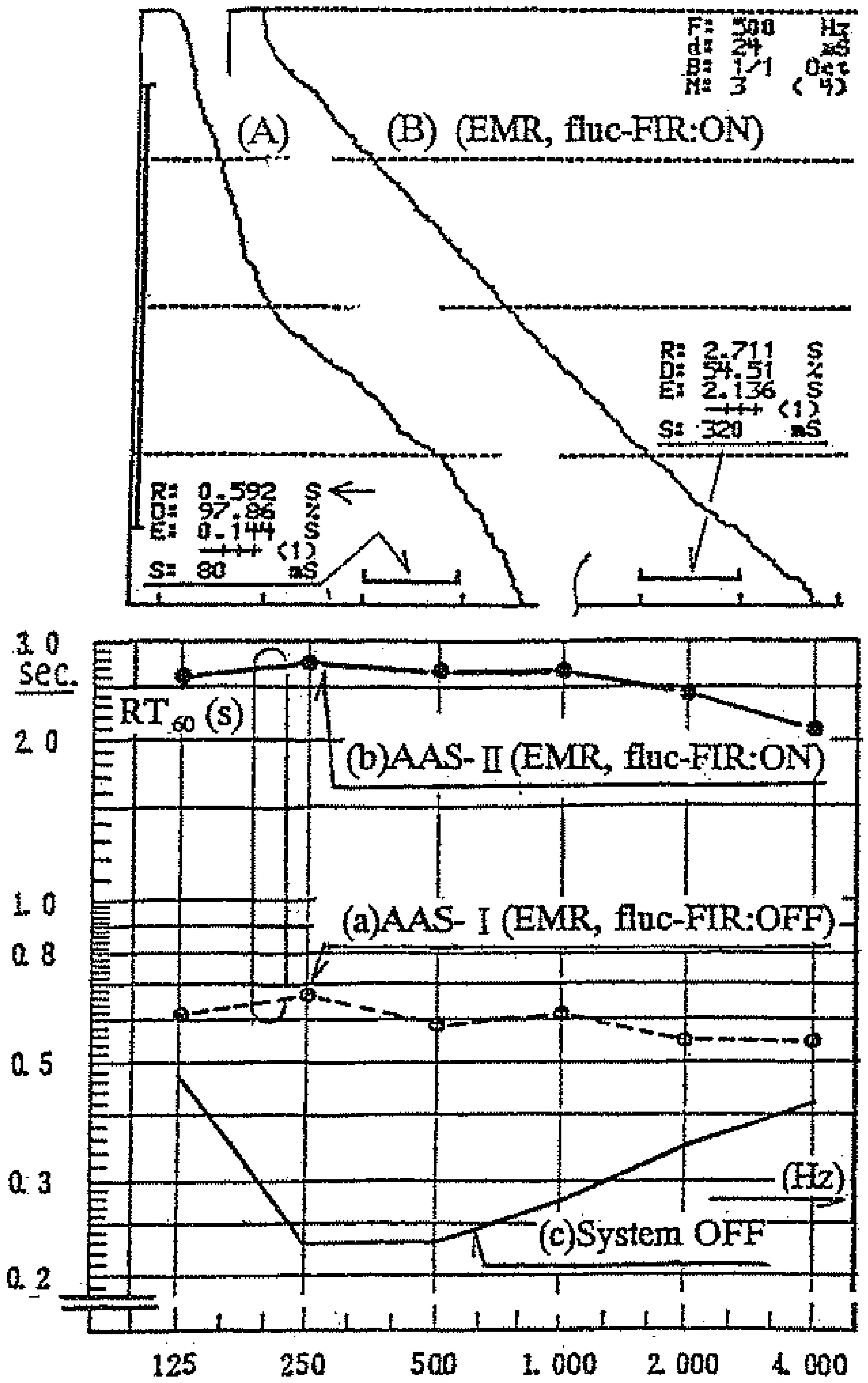


Fig.6 Possibility of extending RT_{60} by A-SF

5.3 Possibility of applying AFC to musical instruments

Applying the same idea to musical instruments, there will be an possibility of "sound field control by sound source itself". Figure 6 shows an example where the same system as in Fig.4 is installed in an electronic musical instrument with small alteration of "SW" into summing circuits. In this example, however, some specific devices for compensating the influence of "direct path" between speaker and microphone may be needed as illustrated with "EC" in the figure. The accuracy of the compensation and the time-variance of sound field have to be studied separately.

6. Discussions

Planning A-SF system for auditoria requires a broad knowledge of sound field control, signal processing, and operation methods for actual concerts. Incidentally, time-variant control (TVC) technologies

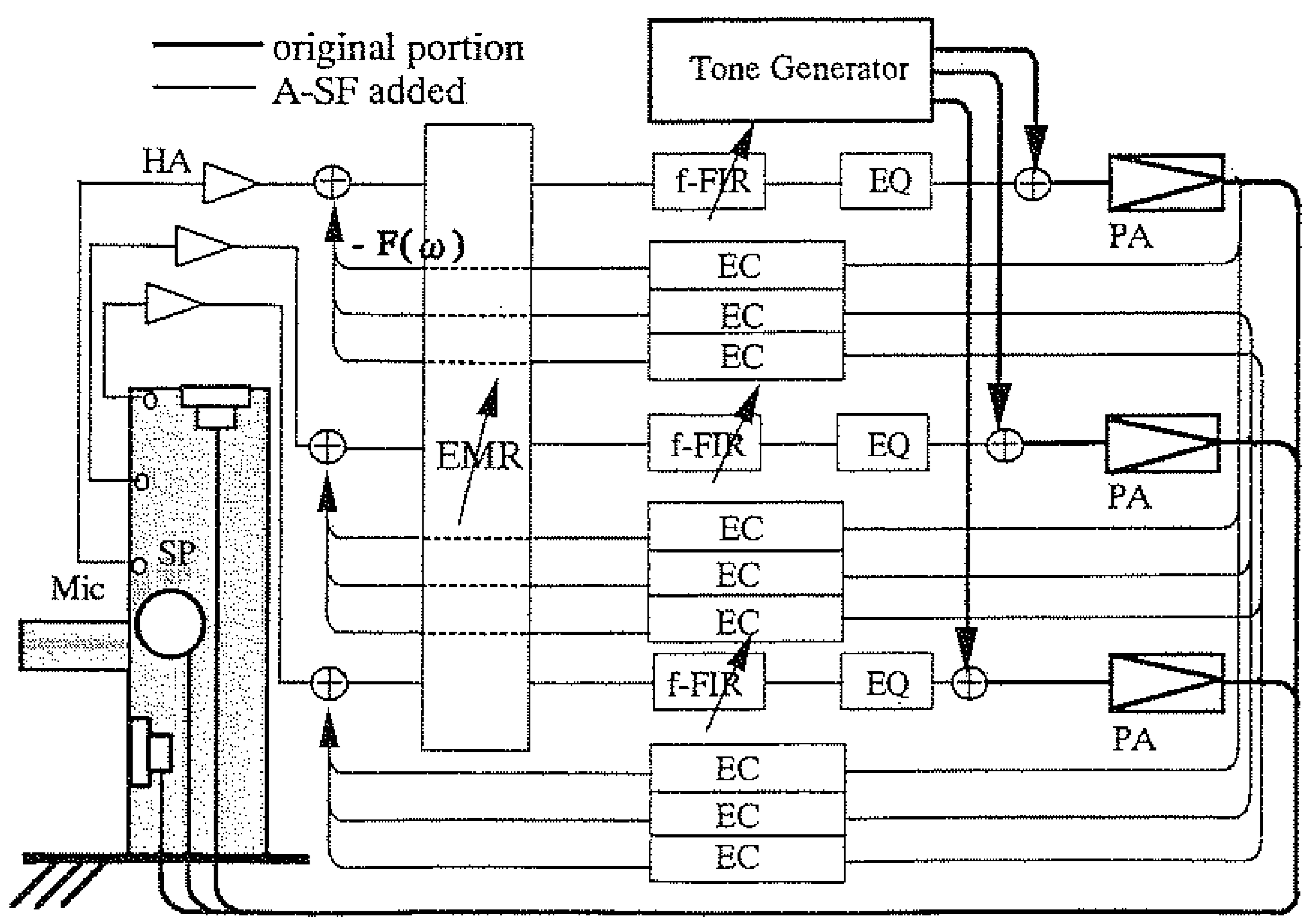


Fig.7 Application of A-SF to musical instruents

such as EMR or fluc-FIR are patented both in the USA and Japan. From here, the following items are to be studied as projects:

1) developing optimum design method for AFC system based on TVC scheme
2) introducing adaptive signal processing for time-variance of sound fields
3) investigating possibility of radically improving sound field under-balcony seats
4) application of A-SF technology for opera (without using orchestra shell)
5) developing a concrete method of introducing AFC to a reverberation room
6) investigating possibility of combined use with sound reinforcement system
7) studying how to announce the operation of AFC to audience and performer

As for the last one, it is rather important to always keep the level of announcement constant to anyone, not very important to announce or not itself. Incidentally, there has been only one example out of 30 auditorium project with AFC, where they keep their clear position of "not announcing on AFC operation".

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