Applied Acoustics **31** (1990) 47–75

Active Field Control in Auditoria

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ABSTRACT

In this paper, technology for the control of the sound field in an auditorium by electro-acoustic means, A-SF (Assistance of Sound Field), is described. The A-SF is one of the AFC (Active Field Control) technologies which have made rapid progress due to the latest advances in digital signal processing, and is characterized by an acoustic feedback loop in the system. Methods for handling loop gain, $G_c(\omega)$, to increase loudness, reverberance, spatial impression, etc., as well as technological problems, are presented on the basis of actual examples and reference to the various existing systems. In particular, the AAS (Assisted Acoustics System), developed by the Yamaha group, is used in a detailed discussion on the validity of A-SF system with an FIR filter in the feedback loop. Finally, the possibility of using an adaptive filter in A-SF is touched on.

1 INTRODUCTION

Control of the room acoustics of an auditorium using electro-acoustic means is sometimes called 'Electro-acoustics in Architecture', or simply 'Electronic Architecture'. Here, we use means which extend over both fields of architectural acoustics and electro-acoustics in a technology to positively control the sound field in an auditorium, which we call AFC (Active Field Control). The AFC differs from ordinary public address (PA) or sound reinforcement (SR) in that the ultimate purpose of using electro-acoustic means in AFC is to obtain room acoustic effects which are the same as if they were obtained by purely architectural means. In the field of acoustic design

Applied Acoustics 0003-682X/90/$03·50

for architecture, AFC exists somewhere in the middle between architectural acoustics and electro-acoustics, and a knowledge of both fields is required in AFC design, particularly room acoustics and signal processing.

AFC is used not only in auditorium acoustics, but also in the broadcasting, audio and recording fields. It has developed to a certain extent independently in each field, causing the definition of AFC to differ slightly from field to field. In particular confusion arises as to the precise meaning of certain terms, for example the difference between 'reverberation control' and 'sound field simulation'. The author has classified AFC technology, in accordance with the objective of control, into the three categories as shown in Table 1. When the various systems and applications make their appearance in the future as expected, these categories will help to keep things from getting confused.

TABLE 1
Three Categories in AFC Technology

Categories of AFC	*Definition*
S-SF[a]	Synthesis of requested sound field in a *given*, or *highly absorbent*, room
A-SF[b]	Control of loudness, reverberance, spatial impression, etc., *on the basis of the existing condition of a room*
P-SF[c]	Production of *spatial sound effect*, mainly in a theatre or an opera house

[a] *Synthesis* of sound field.
[b] *Assistance* of sound field.
[c] *Production* of sound field.

The S-SF (Synthesis of Sound Field) category includes SFS (Sound Field Synthesis: Multi-channel Sound Field Synthesis),[1] SFR (Sound Field Reproduction: Multi-channel Sound Field Reproduction)[2] and various other sound field simulation techniques. The A-SF (Assistance of Sound Field) category includes AR (Assisted Resonance)[3] used in the Royal Festival Hall, MCR (Multiple-Channel Amplification of Reverberation)[4] from the Philips company, the ERES of C. Jaffée (Electronic Reflection Energy System)[5] and the Yamaha group's AAS (Assisted Acoustics System).[6] In addition, the Delta-Stereophony[7] and VISION-Stage,[8] developed in the area of theatre stage technology, may be included in A-SF because of their function of achieving natural and correct localization of the sound image at the position of the sound source on the stage. On the other hand, P-SF (Production of Sound Field) surpasses the framework of the conventional sound effects (SE), producing drastic and natural changes over the entire room space. Examples of this system include the SICS (Sound Image Control System)[8] developed for Science Expo '85 and the SES (Spatial Effect System)[8] for the Ginza Saison Theater, Tokyo, Japan.

In this paper, the technical aspects of sound field control, AFC, are presented with particular concentration on A-SF.

2 APPLICATION AND TECHNICAL PROBLEMS WITH A-SF

Auditorium acoustic effects, that is, the subjective impressions, are said to be composed of the three factors in Table 2. These correspond respectively to physical parameters and the architectural conditions. Just which factor is to be controlled is decided on a case by case basis. The control of reverberance is never the only purpose. In that sense, AFC is a comprehensive technology and the hardware as well must have the optimum design to meet the conditions at the place itself and the purpose it is meant to fulfill. In addition, in A-SF, on the basis of these three factors, improvement of sound distribution, sound quality or the articulation of speech, etc., is also aimed for whilst trying to ensure that the operation of the A-SF system is not perceived.

TABLE 2
Three Major Factors Relating to the Entire Subjective Impression in Auditoria

Subjective parameters	*Objective parameters*	*Architectural conditions*
1 Loudness	Level of direct sound	Distance between source and receiving point
	Level and density of reverberation sound	Room volume (V)
	Sound energy density	Material and area of each room surface
2 Reverberance	Reverberation time (T_{60})	Room volume (V)
	Average absorption coefficient ($\bar{\alpha}$)	Room surface area (S)
		Material and area of each room surface
3 Spatial impression	Relative level of lateral reflections (LE)	Room shape
	Interaural cross correlation (IACC)	Layout of absorption

At present, it seems that the application of this technology is quite widely spread in various fields of acoustic design. Examples are shown in Table 3. For example, in the case of No. 2 in the table, a building of historical importance, any changes in the room shape are not permitted and there is no other alternative to improving or altering the acoustic conditions except to employ A-SF. In addition, with the recent construction of huge multi-event domes or gymnasiums and large convention halls, it is often the case that the design flow proceeds as follows: large room volume → echo anomalies →

TABLE 3
Application of A-SF Technology

Names of A-SF	*Target rooms and spaces*	*Application (items to be controlled)*	*No.*
AR	Multi-purpose hall	Extension of multi-purpose use of auditoria	1
AAS	Historical building	Improvement of existing room acoustics without changing the room interior	2
MCR ERES	Multi-event gymnasium	Raising loudness and reverberance in a room with large volume/absorption	3
RODS	Fan-shaped auditorium	Improving room acoustics on the basis of spatial impression	4
Delay/rev.	Church/chapel	Reverberation time control for playing pipe-organ	5
etc.,	Music training/ practice room	Variable acoustic conditions for music education	6
Delta-stereophony	Opera house	Enhancing direct sound from the stage	7
	Theatre	Improving sound localization	8
VISION-stage	Outdoor hall	Creation of natural reverberation reflections in the free field	9
Recording studio		Facilitation of performance and recording conditions	10
Control room		Monitoring sound under various field conditions	11
Auditorium stage		Replacing orchestra shell further improving the field in the space	12
Under-balcony seat		Improvement of under-balcony condition, even 'removing the balcony'	13

increasing total room absorption → lack of loudness and reverberance → employment of A-SF. It is felt that new fields for application of A-SF technology will increase rapidly as a result of attempts to fulfill future needs. Numbers 10–13 in Table 3 are examples of applications which arise from suggestions of current users of this technology.

As a technology which enables controllability of both electro-acoustics and the naturalness of architectural acoustics, AFC can be said to exist in between the two. Together with a correct understanding of the mechanisms in which the sound field is created, it is necessary to have sufficient knowledge of the hearing sensation and instrumental music. The three technical problems which must be faced in getting A-SF to become commonly accepted are the following items: (1) *stability*; (2) *naturalness*; (3) *controllability*.

The first is safety against instability or tone coloration which is common to all acoustic feedback systems. The second involves whether or not the 'assisted' sound field is natural to our hearing sensation, and the third involves whether the various sound parameters can be controlled as desired. If we include the ease of operation in the third, it is required that the job of 'operation' must not be assigned at the user end. That is, it is necessary for the system to be *designed on an 'operation-free' basis*. Only the minimum number of limited operations are permitted to reproduce the preset conditions. This is extremely important in the design process of actual A-SF hardware.

The factor which governs the above three points is the stable control of the acoustic feedback loop. That is, how the loop gain of each feedback loop can be increased without instability (howling), or in other words, how the margin against instability can be assured. If this can be skillfully handled, item 1 above is assured. Then the assurance of naturalness is given by avoiding tone coloration. Also, the control for achieving the required sound field conditions is promoted simultaneously. Hereafter, some technical ideas for achieving these ends will be introduced by use of the AAS as an example of A-SF.

Figure 1 shows a schematic diagram of the AAS system. The system is divided into a Rev (reverberation control) and ER (early reflection control) section, each consisting of four independent sub-systems. In spite of the increase in complexity caused by an extension system which is routed via a delay or supplementary FIR filter for large room volumes, the number of independent channels is maintained at eight (4×2). It should be noted that the system does not include within the hardware itself, a device or process to create a reverberation signal. Reverberation is created in the acoustic feedback loops in the system which include the same type of FIR filters as the reflection cluster units which produce a bunch of required reflections. For sound pickups, H sets of a quadra-microphone system each consisting of

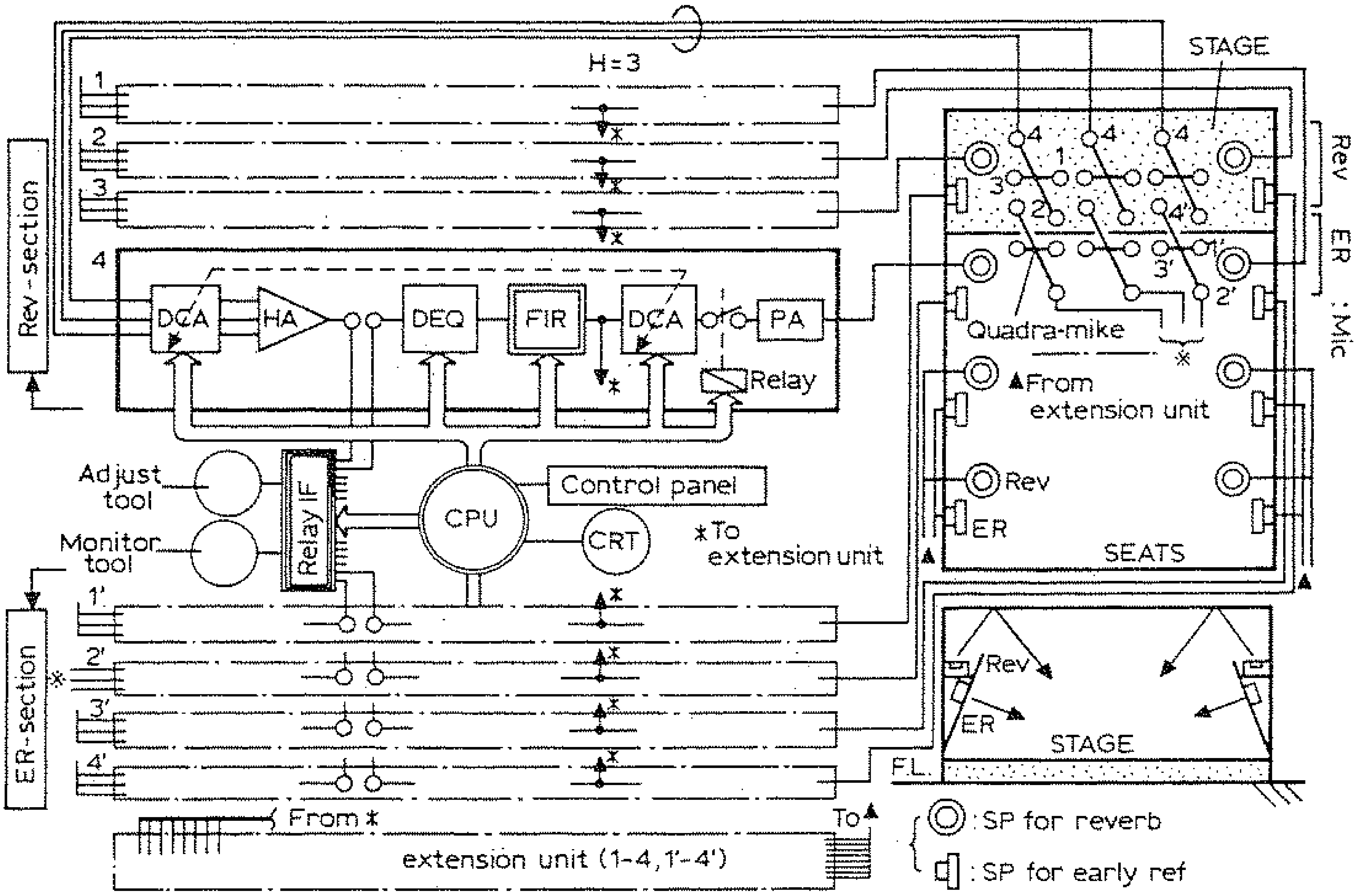


Fig. 1. Schematic diagram of AAS.

four microphones are provided for each respective channel in the Rev and ER section. The value of H is determined within a range of 1–4 according to the scope of the stage area. Of these microphone sets, the system for the ER section is placed so that it is nearer than the reverberation distance with respect to the sound source (instruments on the stage), and that for the Rev section is set around, or slightly further, than the reverberation distance. Both types are suspended from the ceiling so that they are hardly recognized by the audience. In addition, the mutual distance between the four microphones is selected within a range of 0·5–2 m, where the output signals of the units are considered to be sufficiently independent and the signal contents virtually the same.

The loudspeakers are installed within the architecture itself so that they are basically invisible as shown in Fig. 2. The ER speaker is aimed toward the audience while the Rev section is aimed at the direction of the room boundaries, both embedded in the wall or the ceiling. The reasons for this will be explained in the following section. As shown in Fig. 1, all parameters of each unit are controlled digitally by a CPU (Central Processing Unit), and each signal itself from the DEQ (Digital Equalizer) to the DCA (Digitally Controlled Attenuator) is also digitally processed. Of these, the DCA in the output stage functions coordinately with the DCA in the input stage, which is thus used both to control the amplitude of the input signal picked up by the quadra-microphone system, and to adjust the loop gain.

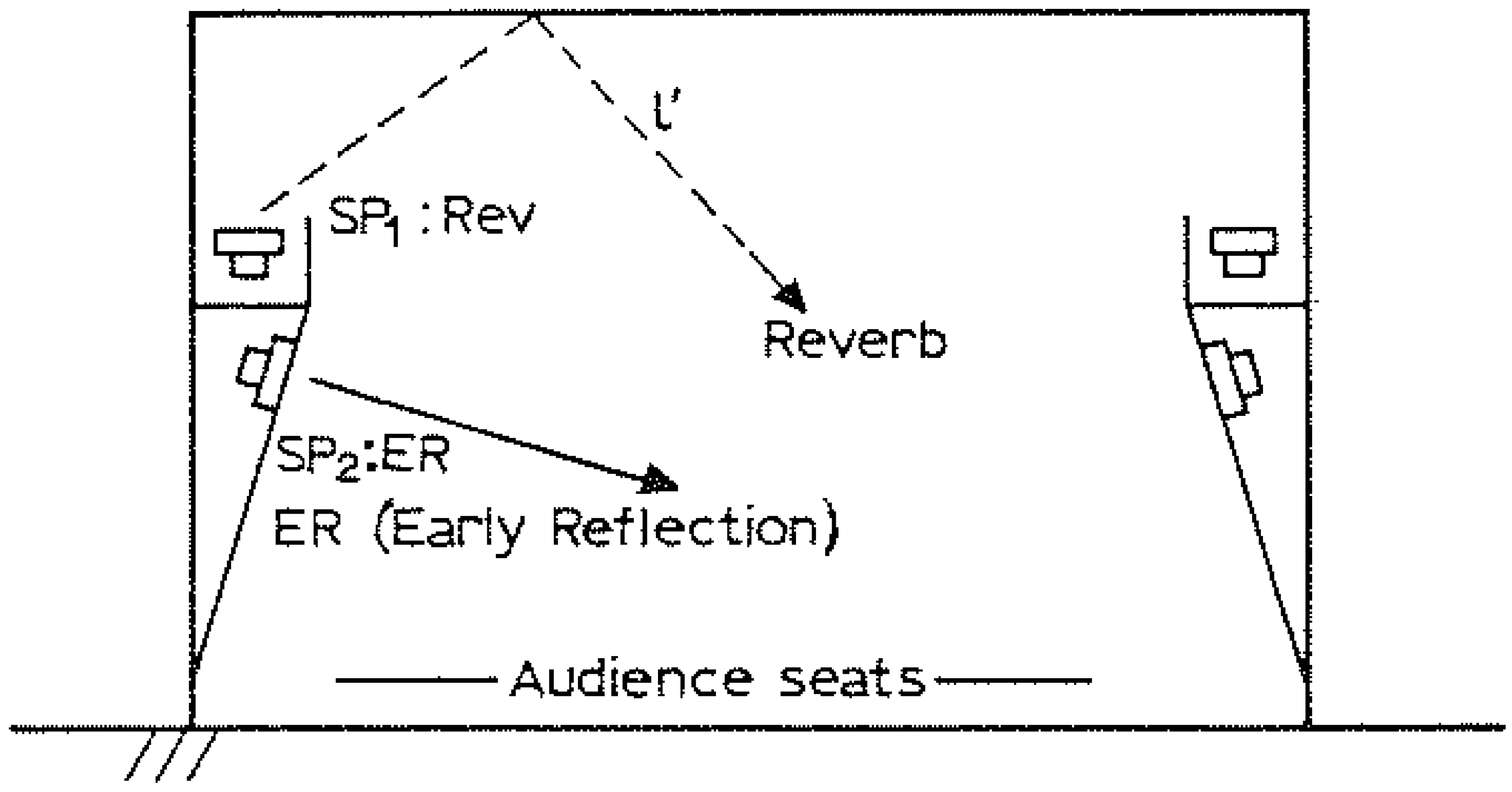


Fig. 2. Basic idea for installation of loudspeakers for A-SF.

In addition to these, a remote control panel for selecting the operation mode, monitoring devices for measuring reverberation time (T_{60}) and adjusting devices for precisely setting loop gain, are all connected to relay units or to the CPU directly. The CPU is specially designed to greatly facilitate the adjustment of the total system. The AAS incorporates all the methods conceivable for achieving stable control of the system. The basic ideas and the technical points of these methods are briefly explained below.

3 EFFECTIVE ARRANGEMENT OF SPEAKERS: RAISING THE REVERBERATION ENERGY

First, in the system shown in Fig. 3, let us consider the case of extending the room reverberation. If a sound source with output W emits sound in a room of volume $V\,\mathrm{m}^3$ and reverberation time T_{60}, the energy density E_0 is obtained by eqn (1). And assuming the first and subsequent reflections form the

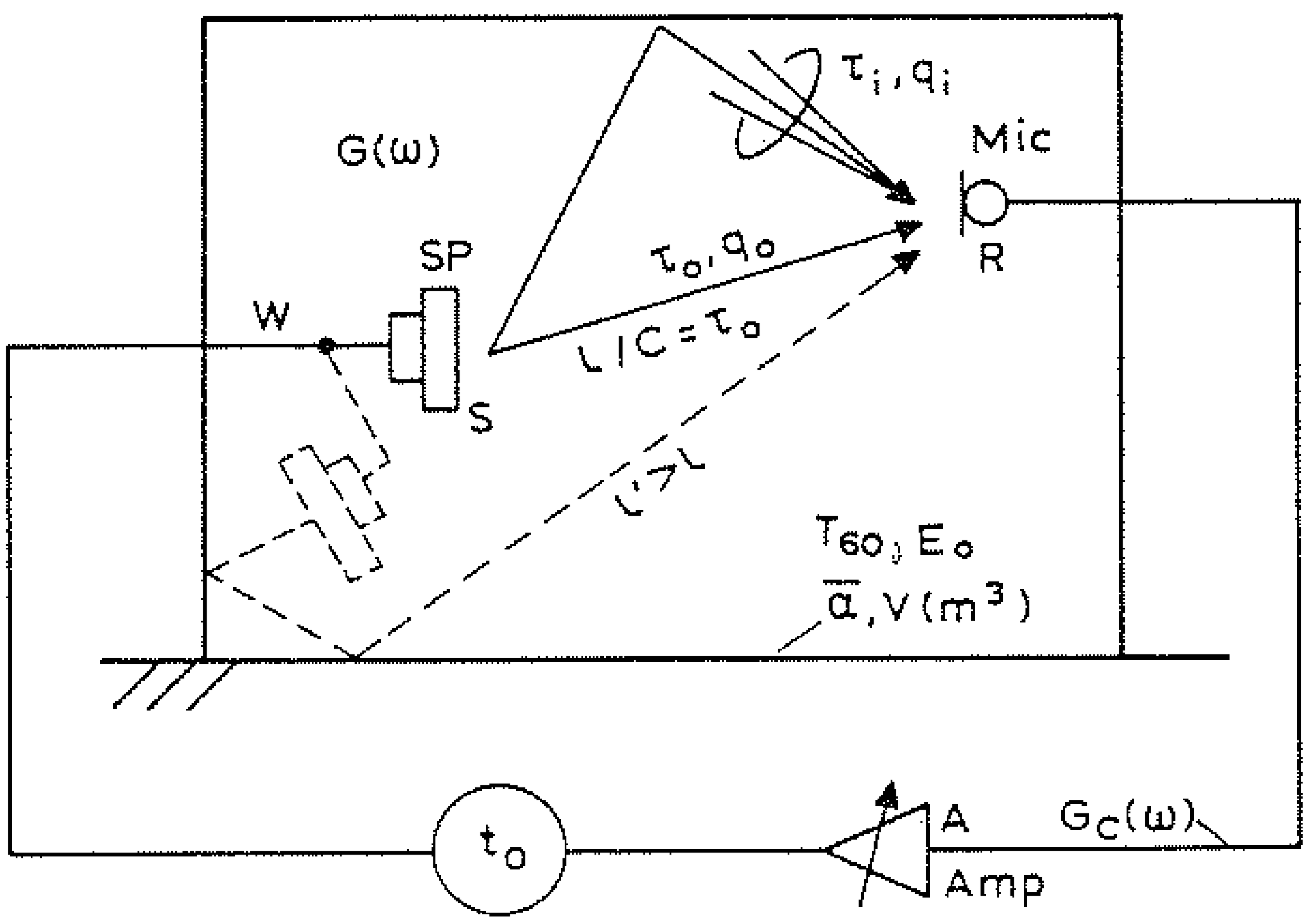


Fig. 3. Basic structure of A-SF system with single delay feedback loop.

reverberation, the diffused sound (reverberation) energy density E_r can be written as eqn (2).

$$E_o = 4W/(cA) = WT_{60}/(13{\cdot}8\,V) \tag{1}$$

$$E_r = (1-\bar{\alpha})E_o = (1-\bar{\alpha})WT_{60}/(13{\cdot}8\,V) \tag{2}$$

where A is the total absorption of the hall, and $\bar{\alpha}$ is the average absorption coefficient. Equation (2) yields extremely interesting results: under the condition W is constant, T_{60} becomes larger if the diffuse energy E_r is increased in some way, say, by electro-acoustic means. If it is possible to increase the original E_r level by 3 dB, this is enough to double T_{60}. The upper limit of the T_{60} extension is determined by the critical point for instability.

What would happen if the speakers are aimed at the ceiling or the wall as shown by the dotted lines in Figs 2 and 3, so as to avoid the direction of the microphones? Since the only purpose is to increase the diffused energy, whether or not the sound is delivered efficiently to the audience is now not a serious problem. All that is needed is to determine the orientations and positions of the speakers so as to prevent instability, and to place them as determined. The validity of this arrangement can be confirmed by conducting an experiment, as shown in Fig. 4, where speakers are aimed at (a) the microphone, and (b) the room boundary. Note that the condition is the same in both cases except for the speaker setting. To enhance the

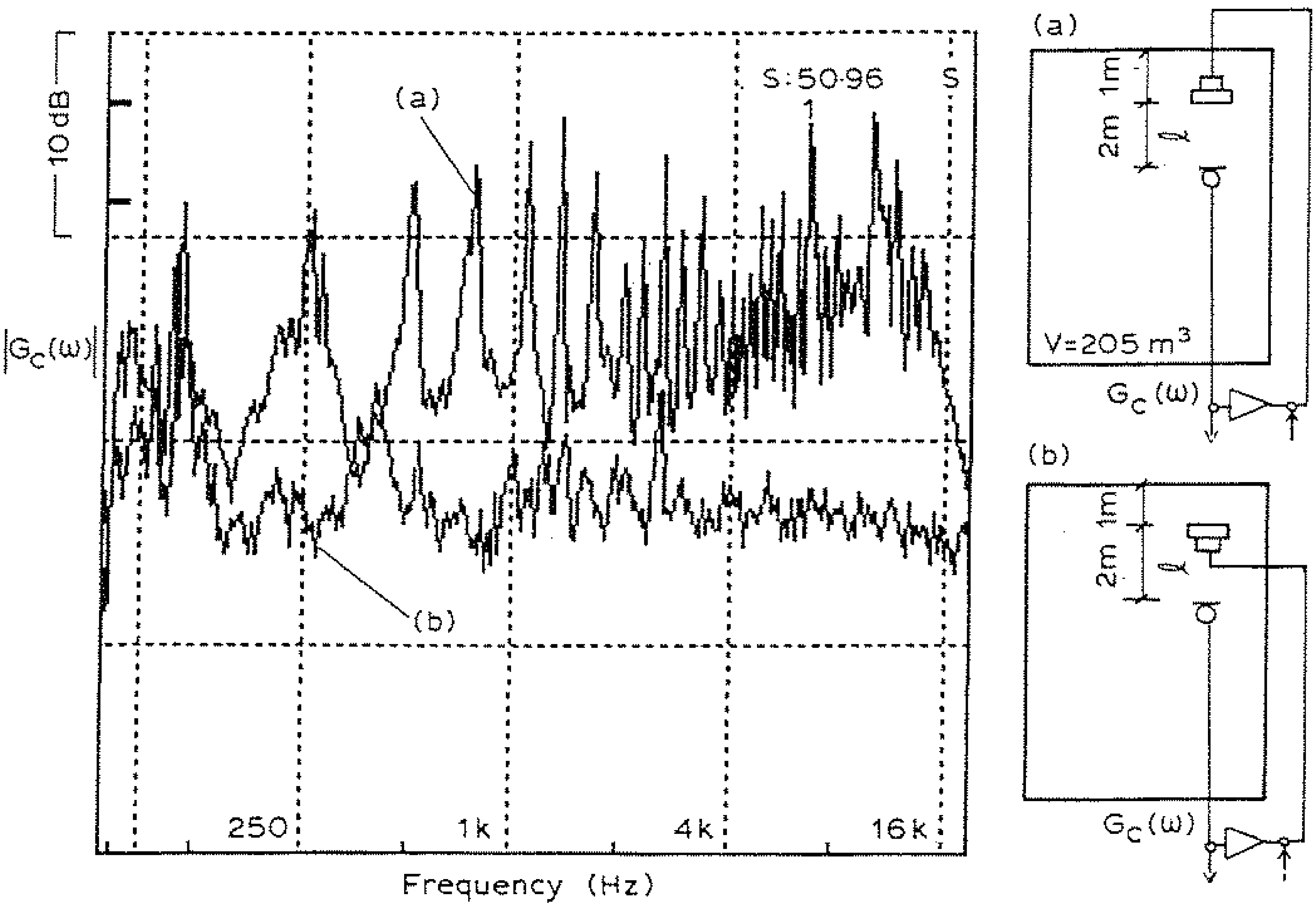


Fig. 4. Comparison of $|G_c(\omega)|$ with SP (speaker) aimed at; (a) microphone and (b) room boundary.

difference, the microphone is much closer to the speaker than in an actual A-SF installation. Due to the direct sound component, a strong comb filter effect is recognized in (a). By aiming at the room boundary, it has almost disappeared in (b), which thus ensures enough margin against the instability. Even if a large volume of power is fed to the speakers, instability will still not occur. Together with the arrangement of placing microphones within the diffuse sound field, aiming the speakers toward the room boundaries is the first step in A-SF system design. The purposes of this are summarized below:

(1) Effective amplifications of diffused sound energy E_r.
(2) Assurance of naturalness of the 'assisted' sound.
(3) Facilitation of 'invisible' installation of speakers.

Each of these items should be given consideration at the actual design stage of A-SF hardware. The sound from each speaker travels a greater distance than the direct sound ($l' > l$ in Fig. 3) and reaches the audience as almost a plane wave, creating extra reflections each time the sound is reflected at the room boundaries. Human hearing sensation can distinguish the difference between a plane wave and a spherical wave, so the naturalness of the sound will then be improved.

In addition, in the acoustic design of an auditorium, it is necessary that the audience does not detect the existence or the performance of an A-SF system. The speaker installation method in Fig. 2 (SP_1) is then also advantageous in this regard. Thus, it is necessary that the A-SF system be designed in conjunction with the architectural design. The validity of the above-mentioned consideration will be verified theoretically in Section 5.

4 SMOOTHING OF LOOP GAIN: USE OF MANY INDEPENDENT CHANNELS

The most important problem in an A-SF system is the control of the acoustic feedback loop including the speakers and microphones. In order to increase E_r, that is, to lengthen T_{60}, one has to somehow bring the loop gain G_c very close to the critical point of instability. This problem is always present, making things even more complex and difficult. Once control is lost, coloration or self-excited oscillation takes place. The reason why multiple channels are necessary in AFC is not to achieve uniformity of the sound field or to increase the number of reflections, but to avoid self-excited oscillation (*guarantee of stability*) and to reduce ringing or coloration (*improvement of naturalness*).

The necessity for many independent channels is evident from the comparison of loop power gains (β's) in Fig. 5(a) and (b). If the transfer

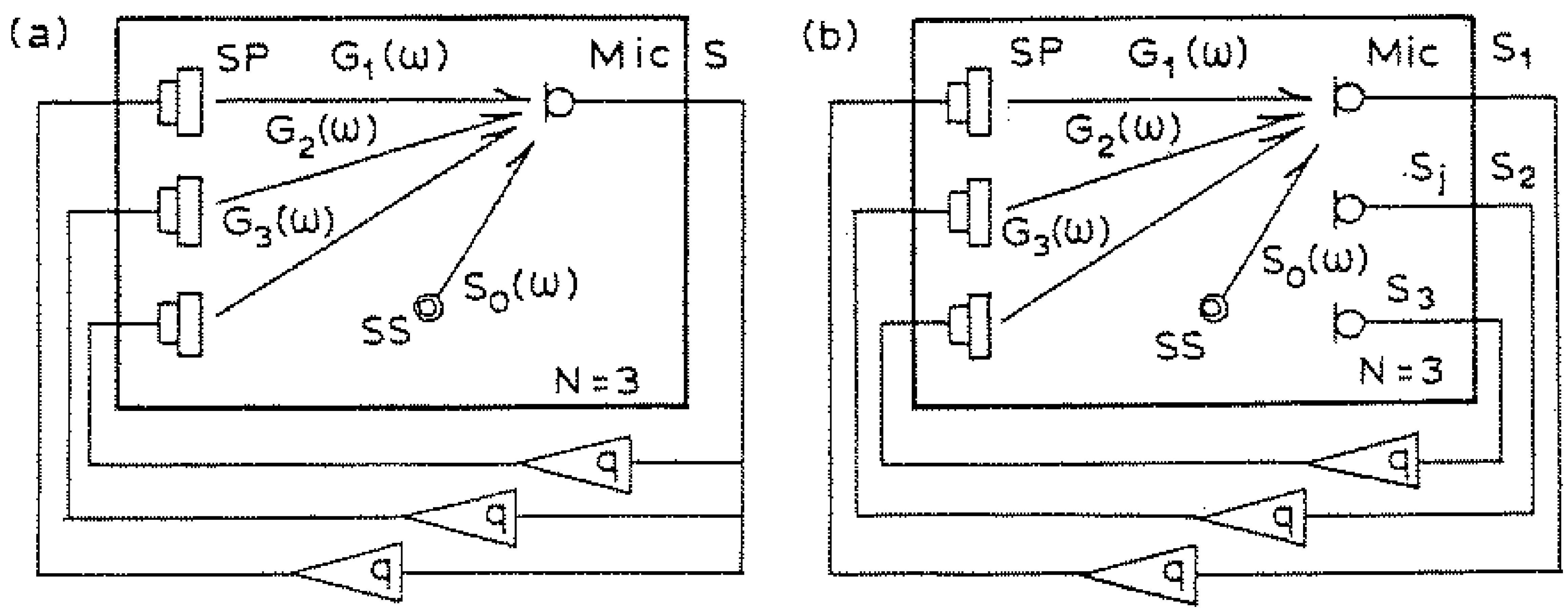


Fig. 5. Illustration for the necessity of independence of each loop, where $N = 3$.

function between speaker and microphone, $Gi(\omega)$, and microphone response $Si(\omega)$, are equal and statistically independent of each other, i.e.

$$|Gi^2| \approx |Gj^2|, \ |Si^2| \approx |Sj^2|, \quad \text{but} \quad Gi \neq Gj, \ Si \neq Sj \tag{3}$$

$\beta \equiv |S^2(\omega)|/|S_o^2(\omega)|$ and for (a) is obtained from eqn (5) by solving eqn (4) for $|S^2(\omega)|$.

$$S(\omega) = S_o(\omega) + qS(\omega)\sum^{N} Gi(\omega) \tag{4}$$

$$|S^2(\omega)| \approx |S_o^2(\omega)| \Bigg/ \left[1 - q^2 \left|\sum^{N} Gi(\omega)\right|^2\right] \tag{5}$$

Incidentally β is here identical to the T_{60} extension ratio, T_1/T_o, where T_1 and T_o are respective T_{60}s with the A-SF system on and off.

Equation (5) shows changes in β with respect to q and illustrates a well known and important fact: as q becomes gradually greater, the denominator $1 - q^2|\sum Gi(\omega)|^2$ approaches zero at a certain frequency ω_0, β will grow selectively as a marked peak at ω_0. In such a situation, only those sounds near to ω_0 will be emphasized and coloration or ringing will start. The important point here is that, even in an ordinary SR system, if the margin against instability is insufficient, the tone quality could be harmed for this reason alone.

Meanwhile, the absolute maximum ΔL_{max} with respect to the average value of $G(\omega)$ is expressed by the following formula. In an ordinary hall ($T_{60} = 2$ s, observation bandwidth: $B = 10$ kHz), we have: $\Delta L_{max} \approx 4{\cdot}3 \ln(\ln BT_{60}) \approx 10$–$12$ dB.[9]

Therefore, it is to that degree that the level of diffused sound energy can be raised without causing instability. However, we can understand from eqn (2) that the degree of T_{60} extension, β is not small. To achieve a large β, the average value of $|Gi(\omega)|$ should be raised but without instability. Averaging on the frequency axis intuitively comes to mind. In actuality, continuous movement of the microphone, frequency modulation,[10] phase modulation,[11] etc., have been proposed. Also, unique trials have been conducted in which a velocity microphone is rotated around the axis connecting the speaker and microphone. However, of these trials, most resulted in various types of distortion or noise, which impeded their development due to their auditory unnaturalness.

In contrast to this, the outline of the multi-channel system[12] proposed by Franssen is rational and directly illustrates the necessity of many independent channels as shown in Fig. 5(b). Here $\beta \equiv |S_j^2(\omega)|/|S_o^2(\omega)|$ and is calculated as follows:

$$Sj(\omega) = S_o(\omega) + q\sum^{N} Gi(\omega)Si(\omega) \tag{6}$$

$$|Sj^2(\omega)| \approx |S_o^2(\omega)| \Big/ \left[1 - q^2 \sum^{N} |Gi^2(\omega)|\right] \tag{7}$$

Equation (7) is obtained from the consideration that the integration of the product of orthogonal functions is zero. While $\sum$ in eqn (5) comprises a *vector sum*, that in eqn (7) comprises an *energy sum* of each $Gi(\omega)$. That is, 5(a) is represented by a single loop of $\bar{G}(\omega) \equiv \sum Gi(\omega)$ with the same statistical property as each $Gi(\omega)$, but in 5(b) the standard deviation of the $\sum$ term is reduced by $1/\sqrt{N}$, which enables β to be increased to that degree.[13]

$$\sigma(\langle Gi^2(\omega)\rangle) = \sigma\left(\sum^{N} |Gi^2(\omega)|/N\right) = \sigma(|Gi^2(\omega)|)/\sqrt{N} \tag{8}$$

where $\sigma(x)$'s are standard deviations of x. In this connection, the supposition for the derivation of eqn (8) that each channel be independent, is important for the design of the system so that all channels are *energy-additive*, not on a *vector-sum* basis. So, even if a number of the speakers or microphones in a given channel are connected in series or in parallel, the open loop gain $q\bar{G}(\omega)$ is given as a vector sum of these devices, which would never promote averaging on the frequency axis. After all, the number of channels is still unity. Also, even if q in 5(a) is replaced by something like an FIR-filter with transfer function $Fi(\omega)$, the situation is not changed except that $|\sum Gi(\omega)|^2 \rightarrow |\sum Gi(\omega)Fi(\omega)|^2$.

In an actual multi-channel system, the number of channels N is ordinarily determined within the range 50–100 according to room volume and the desired value of β. On the other hand, in the AAS of the Yamaha group shown in Fig. 1, the number of channels is kept extremely low, i.e. $N = 4$. Increasing the actual number of microphones or speakers to more than four could be done through adders or extension interfaces only for the purpose of covering the physical size of the performance area on the stage, or covering a large room volume and room shape. The extremely small value of N in AAS is realized by the introduction of the FIR filter, discussed in the next section.

5 ACHIEVING BOTH STABILITY AND CONTROL: INTRODUCTION OF AN FIR FILTER

The advantages of inserting an FIR filter in the acoustic feedback system include the stable control of the feedback loop as well as the flexible modification of the early reflection structure. That is, in the AAS system, it is also possible to control such parameters as Definition (D), Lateral Efficiency (LE), etc., together with the extension of T_{60}. Here, the possibility of stable control is to be verified first. One structural component of AAS in Fig. 1 is shown in extracted form in Fig. 6(a). The FIR portion can be represented by Fig. 6(b).

5.1 Single delay system

First consider the system in Fig. 3, which includes only a single delay t_0 in the feedback loop. In this type of system, if the total gain of the system A is increased, coloration or instability will easily occur at the frequency which corresponds to $t_0 + \tau_0 = t_0 + l/c$, as mentioned above. Thus, it is difficult to raise the diffused sound energy E_r in the room and maintain stability. In typical reverberant spaces, the transmission function $G(\omega)$ between the sound source and the receiving point can be written as follows:

$$\begin{aligned} G(\omega) &= q_o \exp(-j\omega\tau_0) + \sum q_i \exp(-j\omega\tau_i) \\ &= q_o \exp(-j\omega\tau_0) + q(\omega) \exp(-j\omega\bar{\tau}) \end{aligned} \tag{9}$$

where the first term is the direct sound component and the second term corresponds to all the successive reflections including the first reflection. Also, $q(\omega)$ is the vector sum of those components. Ordinarily, since q_0 is the

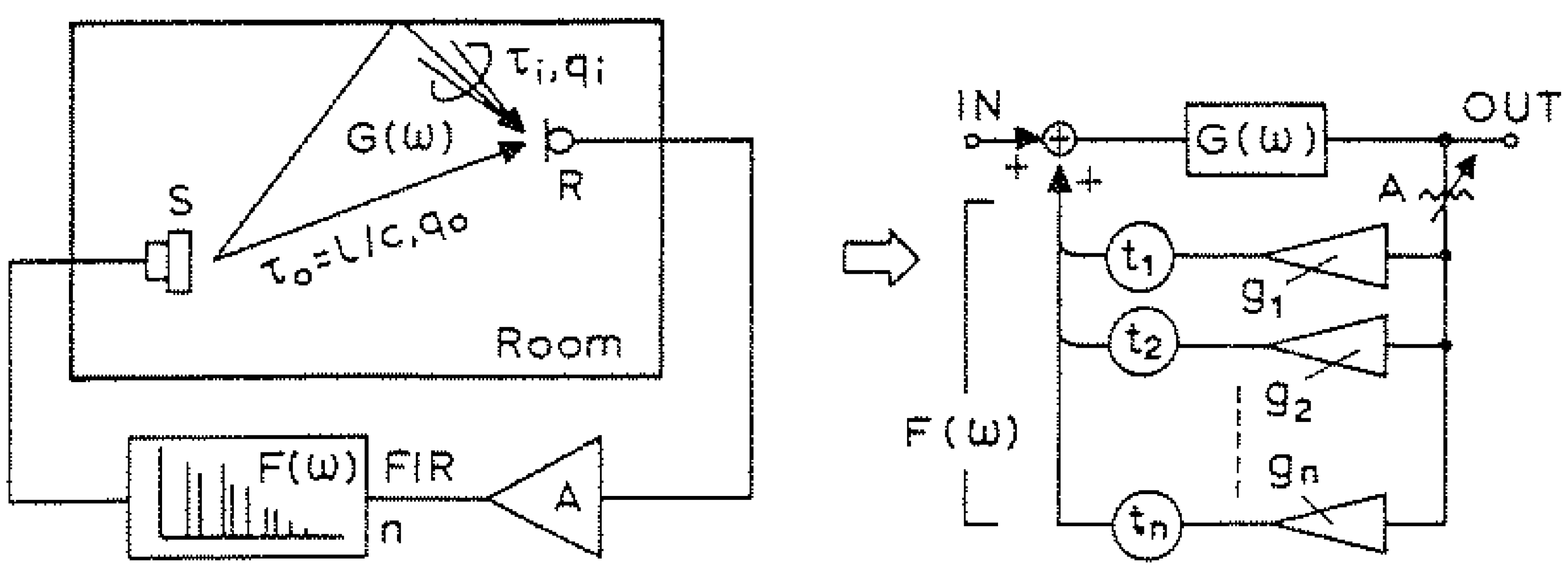


Fig. 6. FIR filter as reflection cluster unit in AAS.

largest and τ_0 is the smallest, the first term has a strong influence on the tone quality of the system:

$$q_o > q_i, \qquad \tau_0 < \tau_i \quad (i = 1, 2, \ldots) \tag{10}$$

Then, with the delay t_0 and gain A of the feedback system, the open loop and loop gain of this system are written as follows:

$$\begin{aligned} G_o(\omega) &= G(\omega)A \exp(-j\omega t_0) \\ &= A[q_o \exp(-j\omega(\tau_0 + t_0)) + q(\omega)\exp(-j\omega(\bar{\tau} + t_0))] \end{aligned} \tag{11}$$

$$\begin{aligned} G_c(\omega) &= \sum_{n=1}^{\infty} A^n[q_o \exp(-j\omega(\tau_0 + t_0)) + q(\omega)\exp(-j\omega(\bar{\tau} + t_0))]^n \\ &= \sum_{n=1}^{\infty}\sum_{r=0}^{n} {}_nC_r A^n q_o^{n-r} q^r(\omega) \exp(-j\omega(n\tau_0 + nt_0 - r\bar{\tau} - rt_0)) \end{aligned} \tag{12}$$

where ${}_nC_r$ is the binomial coefficient. Since eqn (12) is to be solved for $|G_c^2(\omega)|$, while taking into consideration that the direct sound q_o and the reflected sound group $q(\omega)$ are mutually incoherent and that the integration of the orthogonal product is zero, $G_c^2(\omega)$ can be written as:

$$\begin{aligned} G_c^2(\omega) &\approx \sum_{n=1}^{\infty} [Aq_o \exp(-j\omega(\tau_0 + t_0))]^{2n} + \sum_{n=1}^{\infty} [Aq(\omega) \exp(-j\omega(\bar{\tau} + t_0))]^{2n} \\ &= \frac{A^2 q_o^2 \exp(-2j\omega(\tau_0 + t_0))}{1 - A^2 q_o^2 \exp(-2j\omega(\tau_0 + t_0))} + \frac{A^2 q^2(\omega) \exp(-2j\omega(\bar{\tau} + t_0))}{1 - A^2 q^2(\omega) \exp(-2j\omega(\bar{\tau} + t_0))} \end{aligned} \tag{13}$$

Equation (13) shows the sum of two separate feedback loops. Also, the direct sound q_o on the time axis is outstanding and the amplitude is at the maximum, that is, generally, $q_o^2 \gg q_i^2 (i = 1, 2, \ldots)$. Therefore, if A is increased, the denominator in the first term preferentially approaches zero, resulting in the generation of coloration or instability at the frequency $\omega_0 = 2\pi k/(\tau_0 + t_0)$ (k is an integer), as in Fig. 7(a). This is the reason why a single delay system does not work very well. In cases where delay t_0 is missing in the loop, the condition is the same except for $\tau_0 + t_0 \rightarrow \tau_0$. Rather, the spacings between peaks on the frequency axis become large, which serves only to aggravate the situation.

5.2 Feedback system with an FIR filter

The first term of eqn (13) vanishes by avoiding any influence from the direct sound. An example was shown in Section 3, the method of aiming the

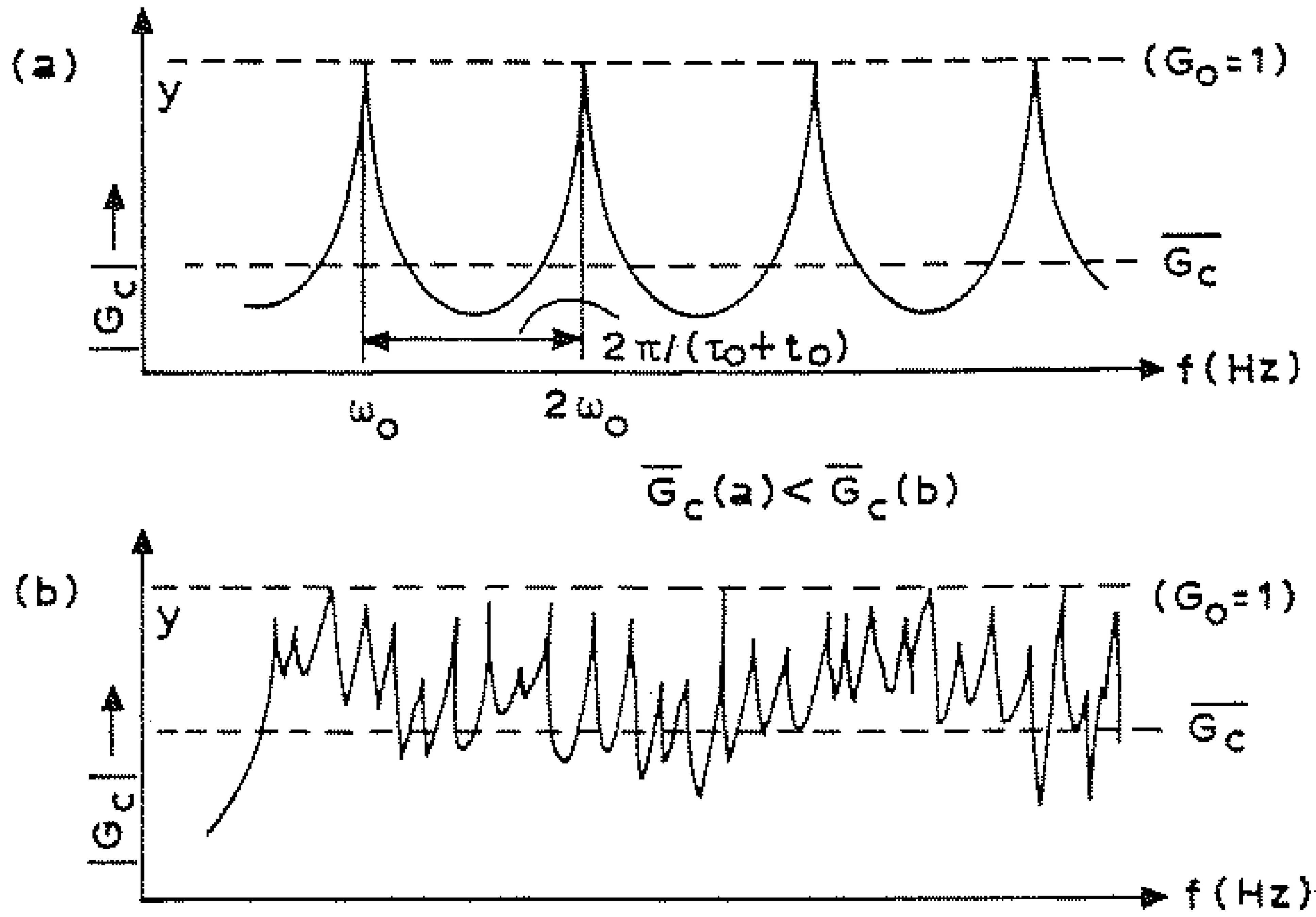


Fig. 7. Spectrum of $|G_c(\omega)|$ around $G_o(\omega)=1$: (a) loop with direct sound; (b) loop with reverberant sound.

speakers at the room boundaries. Through this consideration, only the second term remains, that is, the spectrum appears in accordance with the group of reflections as in Fig. 7(b). The circumstances are significantly improved. However, this is not enough because the influence of the first reflection replaces that of the direct sound. To accomplish a large improvement, FIR filters are used as reflection cluster units. If FIR filters $F(\omega)$, made up of n elements, as shown in Fig. 6, are placed in the loop in place of the single delay t_0, the new $G_o(\omega)$ and $G_c^2(\omega)$ can be derived as follows in accordance with eqns (11) and (13):

$$G_o(\omega) = AF(\omega)q_o \exp(-j\omega\tau_0) + AF(\omega)q(\omega)\exp(-j\omega\bar{\tau}) \qquad (14)$$

$$G_c^2(\omega) \approx \frac{A^2F^2(\omega)q_o^2 \exp(-2j\omega\tau_0)}{1 - A^2F^2(\omega)q_o^2 \exp(-2j\omega\tau_0)} + \frac{A^2F^2(\omega)q(\omega)\exp(-2j\omega\bar{\tau})}{1 - A^2F^2(\omega)q(\omega)\exp(-2j\omega\bar{\tau})} \qquad (15)$$

Meanwhile the irregularity of the transmission function in an ordinary room is expressed by the relative standard deviation from the mean value, say, about 0·523. If the number of elements n in the FIR filter is sufficiently large and its parameters are designed or adjusted, variation of $F(\omega)$ can be lowered to this degree. In other words, if the new peaks produced by $F(\omega)$ are inserted between the peak frequencies (ω_0s) in a well balanced manner, the peaks due to the direct sounds $q_o \exp(-j\omega\tau_0)$ are 'masked' by $F(\omega)$ and then the first and second terms in eqn (15) both take on the characteristics in Fig. 7(b). Thus, they reach the range of instability virtually simultaneously as A is increased. As explained so far, in the FIR system, it is possible to keep the diffused sound energy E_r large compared to a single delay system, which makes it possible to lengthen T_{60} stably. In the actual AAS, 4–8 independent

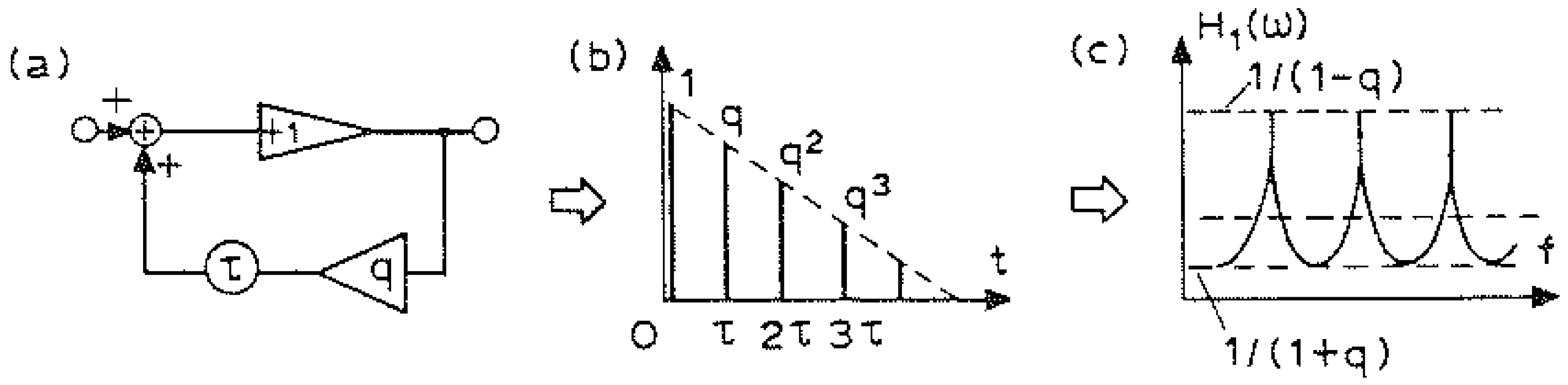


Fig. 8. Feedback loop with single delay ($M = 1$).

loops are overlapped promoting the averaging. The transmission function is then further smoothed out.

5.3 Comparison of transfer functions of feedback systems

In order to confirm the effects of an FIR filter in the acoustic feedback system, a simple comparison was attempted, as shown in Fig. 8. In the discussion, respective room responses are not included. The absolute value of the transfer function, $|H_1(\omega)|$, of the single delay system (the number of elements $n = 1$) in Fig. 8 is determined by the following formulas.

$$H_1(\omega) = \sum_{n=0}^{\infty} [q \exp(-j\omega\tau)]^n = (1 - q \exp(-j\omega\tau))^{-n} \tag{16}$$

$$|H_1(\omega)| = 1\Big/\sqrt{1 + q^2 - 2q\cos\omega\tau} \xrightarrow{\omega = 2\pi k/\tau} 1/(1-q)$$

$$\xrightarrow{\omega = 2\pi(k+1)/\tau} 1/(1+q) \tag{17}$$

$$|H_1(\omega)|_{\max}/|H_1(\omega)|_{\min} = (1+q)/(1-q) \tag{18}$$

$$T_{60} = -60\tau/(20\log q) = -3\tau/\log q \tag{19}$$

That is, for $H_1(\omega)$, the maxima and minima on the frequency axis are taken in turns, and the discrepancy increases as q increases as shown by eqn (18). Thus, as q approaches 1, the peak level with respect to the average value of $|H_1(\omega)|$ becomes large quickly. Stable control of the system is thus difficult. However, from the auditory point of view, if there are 15 or more independent peaks for each 100 Hz, the system cannot be distinguished at all from a flat system. If τ is made large, the peak density increases, but in this case, a flutter echo would be developed on the time axis and, as shown by eqn (19), a smooth reverberation decay curve may not be obtained.

On the other hand, let us look at the system with an FIR filter of $n = 2$, shown in Fig. 9. As in (b) of the figure, reflections are generated at the times $t = k\tau_1 + j\tau_2 (k, j = 0, 1, 2, \ldots)$ on the time axis and their density increases in proportion to t. The peaks of the transfer function in eqn (20) appear on the

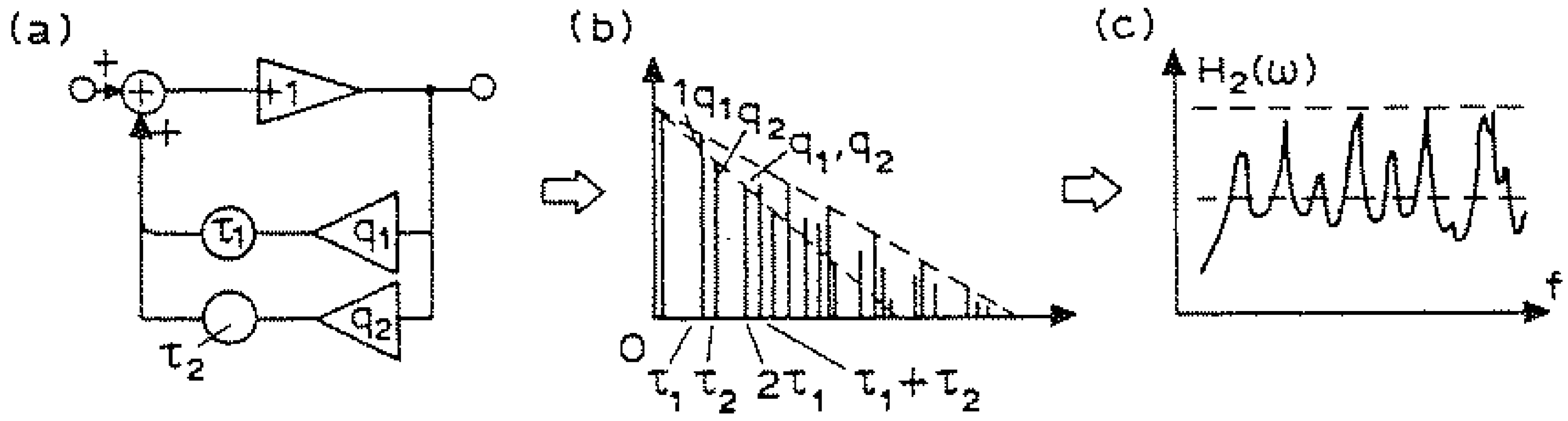


Fig. 9. Feedback loop with FIR filter ($n=2$).

frequency axis at the frequencies where the vector of the denominator takes a minimum value on a complex plane. Incidentally, in this connection, $k=j=0$ corresponds to the direct sound.

$$H_2(\omega) = \sum_{n=0}^{\infty} A^n[q_1 \exp(-j\omega\tau_1) + q_2 \exp(-j\omega\tau_2)]^n$$

$$= 1/[1 - A(q_1 \exp(-j\omega\tau_1) + q_2 \exp(-j\omega\tau_2))] \tag{20}$$

where A is the coefficient of the attenuator, and for the stability of the system, $A < 1/(q_1+q_2)$. In Fig. 10, each component of the denominator of $H_2(\omega)$ in eqn (20) is expressed as $\overrightarrow{OA}$, $\overrightarrow{OB}$, etc., and then the minimum of $\overrightarrow{OE}$ in the denominator is verified.

$$\overrightarrow{OE} \equiv 1 - A(q_1 \exp(-j\omega\tau_1) + q_2 \exp(-j\omega\tau_2)) \tag{21}$$

(1) In the case of $\angle DOA = 0$: First, at $\omega_1 \equiv 2k\pi/\tau_1$, $\overrightarrow{OA}$ overlaps the X-axis, i.e. $\angle DOA = 0$. For this ω_1, if an integer k exists which satisfies eqn (22),

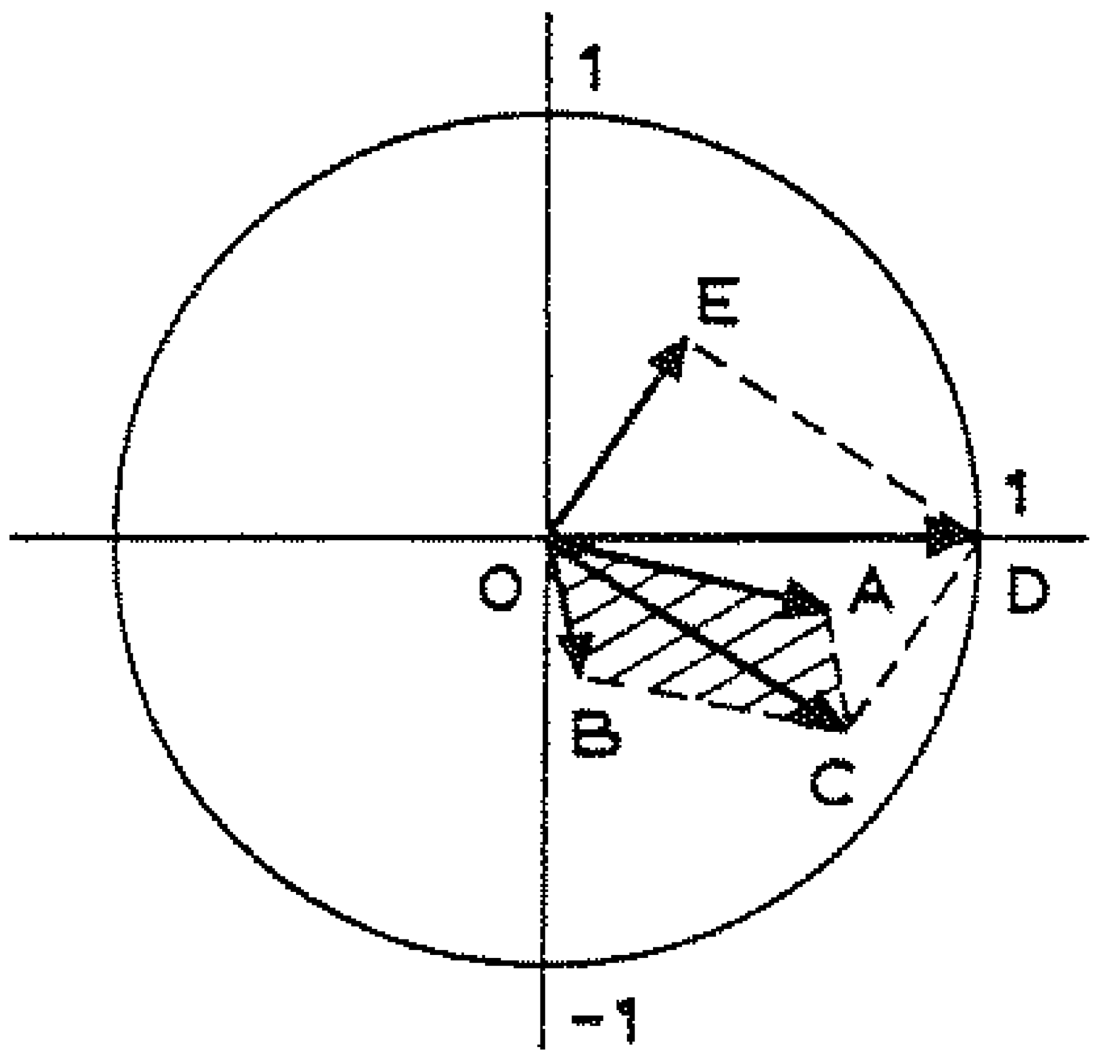


Fig. 10. Vector space for illustrating $H_2(\omega)$ components:

$\overrightarrow{OA} = Aq_1 e^{-j\omega\tau_1}$

$\overrightarrow{OB} = Aq_2 e^{-j\omega\tau_2}$

$\overrightarrow{OC} = A(q_1 e^{-j\omega\tau_1} + q_2 e^{-j\omega\tau_2})$

$\overrightarrow{OD} = 1$

$\overrightarrow{OE} = 1 - A(q_1 e^{-j\omega\tau_1} + q_2 e^{-j\omega\tau_2})$

$\overrightarrow{OB}$ overlaps $\overrightarrow{OD}$ in the same way as $\overrightarrow{OA}$, and as a result, the minimum value is taken for $\overrightarrow{OE}$.

$$\omega_1\tau_2 = 2n\pi, \quad \text{i.e. } 2k\pi\tau_2/\tau_1 = 2n\pi \ (n = 0, 1, 2 \ldots) \tag{22}$$

A number of ω_1s satisfy this condition and $|H_2(\omega)|$ has peaks at those frequencies.

(2) In the case of $\angle DOB = 0$: In exactly the same way, $|H_2(\omega)|$ has a peak at a frequency $\omega_2 = 2k\pi/\tau_2$ which satisfies eqn (23):

$$\omega_2\tau_1 = 2n\pi, \quad \text{i.e. } 2k\pi\tau_1/\tau_2 = 2n\pi \ (n = 0, 1, 2, \ldots) \tag{23}$$

(3) In the case where $\overrightarrow{OC}$ overlaps $\overrightarrow{OA}$: The vector sum $\overrightarrow{OC}$ out of $\overrightarrow{OA}$ and $\overrightarrow{OB}$ could also overlap $\overrightarrow{OA}$.

$$|\overrightarrow{OC}| = A\sqrt{q_1^2 + q_2^2 + 2q_1q_2\cos\omega(\tau_2 - \tau_1)} \tag{24}$$

If $\omega(\tau_2 - \tau_1) = 2k\pi$, that is, if $\omega = \omega_3 \equiv 2k\pi/(\tau_2 - \tau_1)$, $\overrightarrow{OA}$ and $\overrightarrow{OB}$ become the same amplitude and both overlap. Further, if $\omega\tau_1$ (or $\omega\tau_2$) is $2n\pi$ with respect to ω_3, $\overrightarrow{OC}$ overlaps $\overrightarrow{OA}$ and $|\overrightarrow{OE}|$ becomes the minimum. Thus $|H_2(\omega)|$ has a peak.

$$|\overrightarrow{OC}| = A(q_1 + q_2) \xrightarrow{\omega\tau_1 = 2n\pi} /\!/ \overrightarrow{OA} \tag{25}$$

As mentioned above, $|H_2(\omega)|$ has peaks at many points ω_1, ω_2, ω_3 on the frequency axis, and the shape of the response curve becomes complex, as seen in Fig. 9(c). As a result, the average level of $|H_2(\omega)|$ can be increased without fear of instability compared with simple delay systems. With an actual FIR filter, the number n is much larger, promoting the averaging. Thus, the total gain can be easily raised to reinforce the reverberation energy E_r.

5.4 Other possibilities of the FIR system

As is already known, the reverberation time $T_{60} = 0{\cdot}16V/A$ can be extended by both reducing the sound absorption A and increasing the room volume V. However, as shown in Fig. 11, both of these phenomena differ physically and then their perceived impressions are also completely different. Those ideas mentioned above and adopted for AAS and MCR, i.e. the attempts to reduce the decay rate of each reflection by strengthening the diffuse sound energy E_r, should pertain to the former, i.e. to Fig. 11(a).

With respect to this, if the volume V is increased, the reflection density is lowered, as shown in Fig. 11(b), thus extending T_{60}. At this time, loop gain is kept constant, and therefore, the reverberation energy E_r does not increase either. Thus, by keeping the intervals of n reflections created by the FIR filter sufficiently large, and by matching the overall decay shape with the desired decay rate, T_{60} can be extended without fear of instability. This method is effective for pipe-organ concerts, where control of, not the level of reverberation, but the length of the reverberation itself, is desired. This is also quite valid when it is aimed to increase the perceived room size.

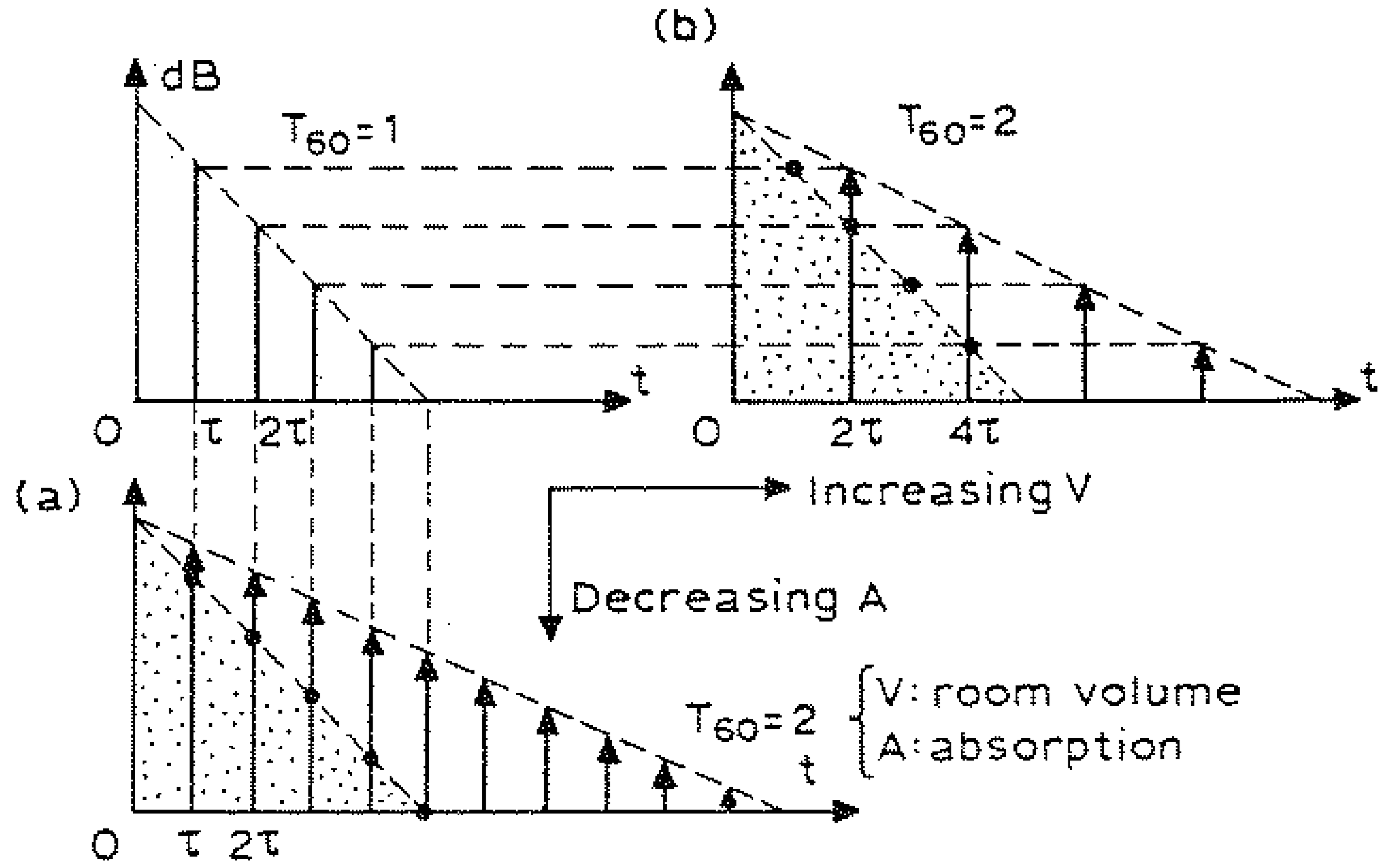


Fig. 11. Two different approaches for extending T_{60}: (a) by controlling absorption A; (b) by changing room size V.

The AAS combines both of the types above in an attempt to stabilize the acoustic feedback loop and to expand the range of control of T_{60}. However, it should be carefully noted that the use of type (1) is limited by instability or coloration, and type (2) by the perceived feeling of unnaturalness of sound such as flutter echoes due to the excessively large reflection intervals.

The final possibility of the FIR system is the control of early reflections. Through the appropriate layout of microphones and speakers, it is possible to vary LE (Lateral Efficiency) and D_{50} (Deutlichkeit) as well as to improve the sound pressure distribution. Actually, by this means, the ease of performance on the stage and the spatial impression at the audience seats are heightened. In addition, the articulation of speech and sound quality can be improved at the seats underneath a balcony. In this way, the degree of freedom in *altering*, *correcting* or *improving* the sound field is quite high in the FIR system. Nevertheless, to control the initial reflection effectively, the microphones should be placed rather close to the sound source on the stage. Further, in the AAS system in Fig. 1, for example, the Rev section and the ER section are completely separated.

6 COMPARISON OF ACTUAL EXAMPLES OF T_{60} EXTENSION

6.1 Other methods for extending reverberation

Other than those discussed above, there have been different types of ideas for reverberation control as follows:

(a) individual control of room modes,
(b) removing the direct path,
(c) cutting the feedback loop.

As an example of (a), Assisted Resonance installed in the Royal Festival Hall in 1964[3] is famous, where the peaks in $G_c(\omega)$ due to the room modes are precisely and individually controlled by very high-Q filters composed of 172 pairs of microphones and resonators embedded in the ceiling. Considering this, one may have two major approaches for increasing $G_c(\omega)$. One is called a *frequency independent* system which aims at the fine tuning of each peak of $G_c(\omega)$ as represented by Assisted Resonance above. In contrast to this, the other is called a *broad-band* system which aims at averaging and smoothing $G_c(\omega)$ on the frequency axis as in MCR and AAS. It may be noted that, in the frequency independent system, even a small change in the sound field condition, which shifts the peak position of the loop gain $|G_c(\omega)|$ on the frequency axis, could cause instability which seems then somehow sensitive requiring ceaseless tuning before concerts.

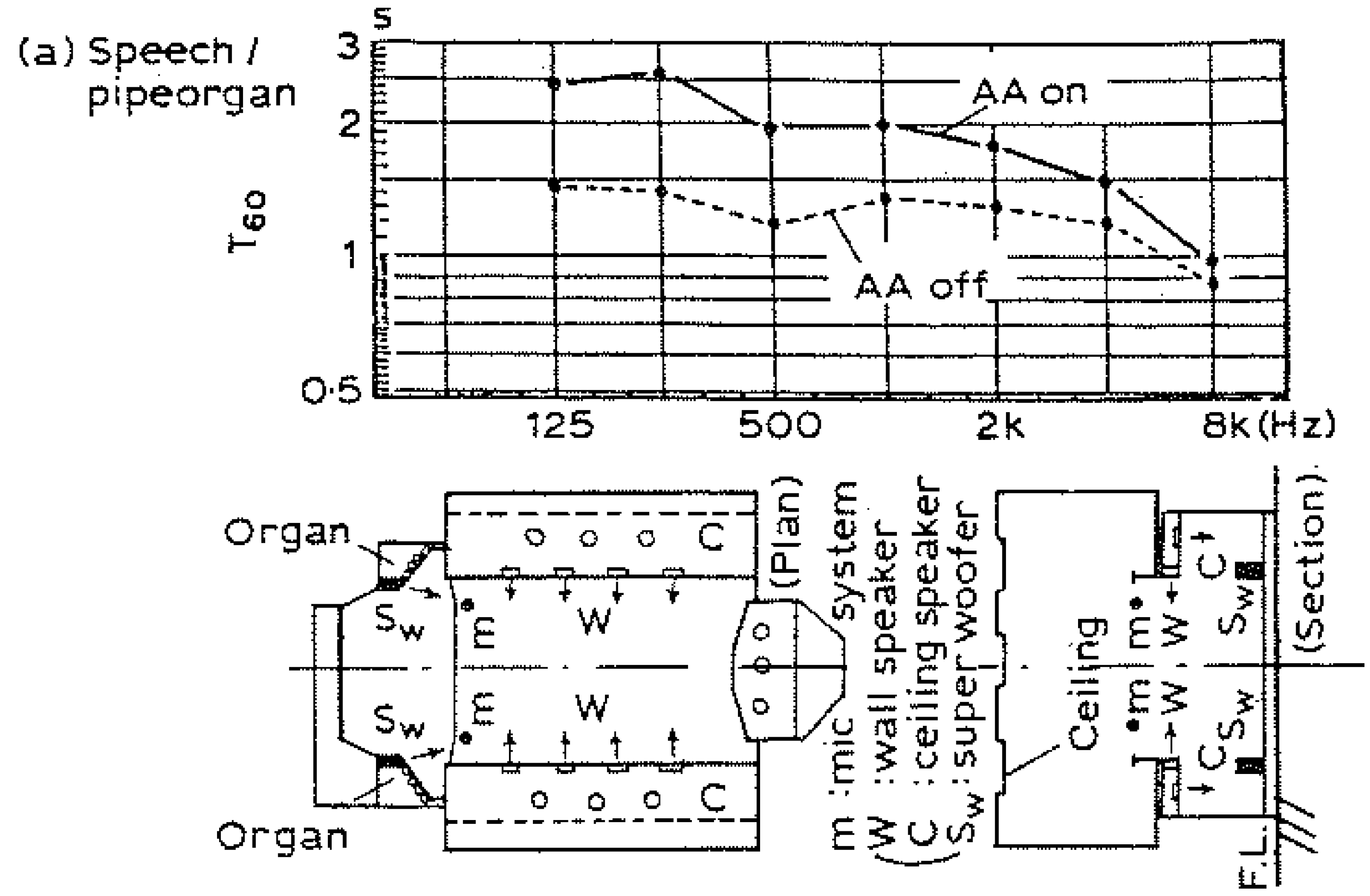
The idea of the Direct Feedback system, based on the idea (b), was developed for the Beatie Theater in Capetown, South Africa,[14] in which the sharp directivity of horn speakers and parabolic microphones were utilized to remove the direct path between them, i.e. for elimination of the first term in eqn (13). With only two channels, T_{60} was extended with the E_r raised by 1 dB.

The so-called RODS (Reverberation On Demand System)[15] is based on the idea (c), cutting the feedback loop by placing switches S1 and S2 at the input and output stages of reverberation generation in the loop. They close conversely according to the rise and decay of the sound field keeping the loop always 'open', which removes the risk of instability or of coloration.

6.2 Example of AAS implementation

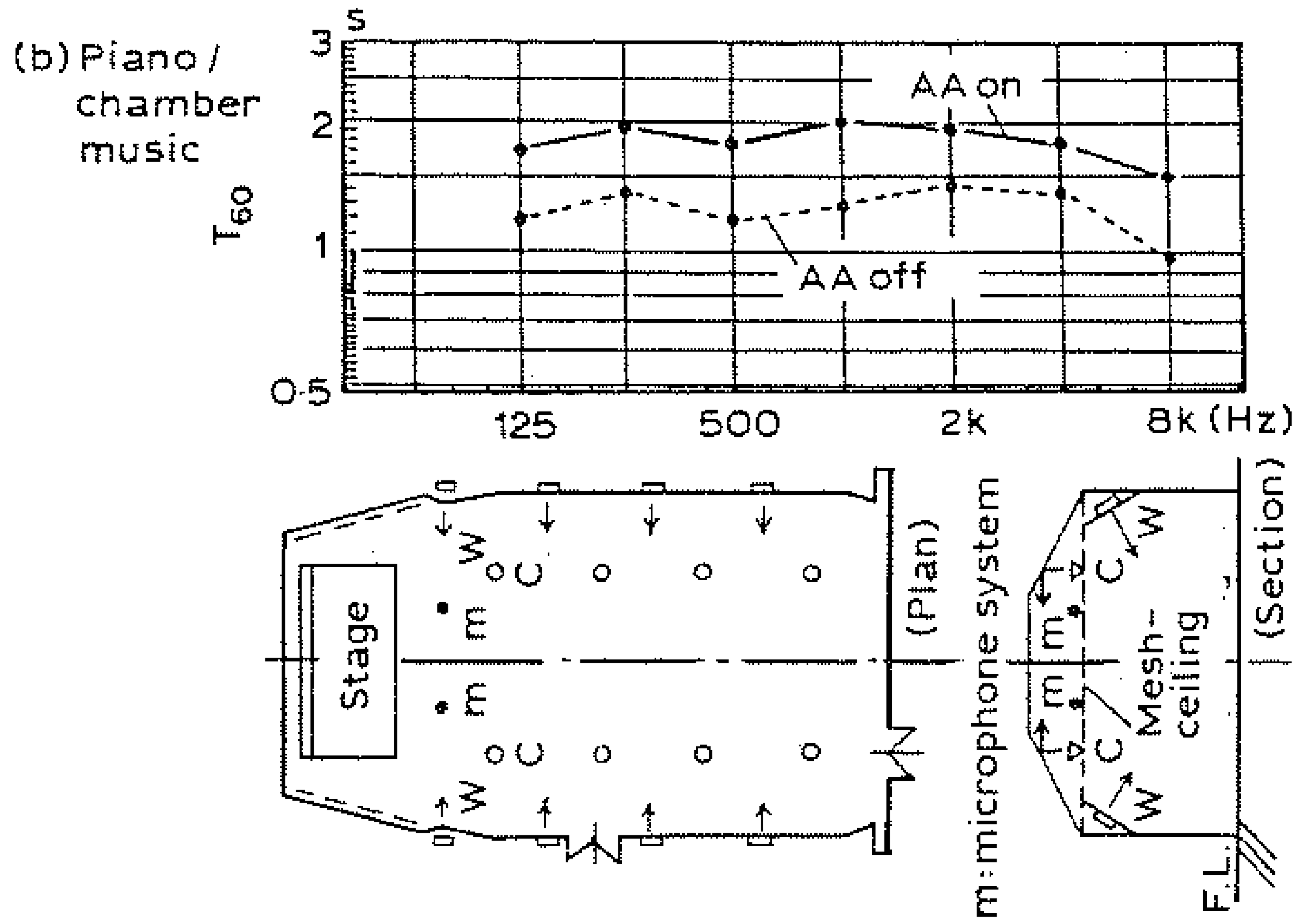
The technical framework of the AAS system which includes an FIR filter in the feedback loop is as follows:

(1) Assurance of sufficient loop gain by the combination of (a): speakers aiming at room boundaries, (b): many independent channels and (c): FIR feedback loops.



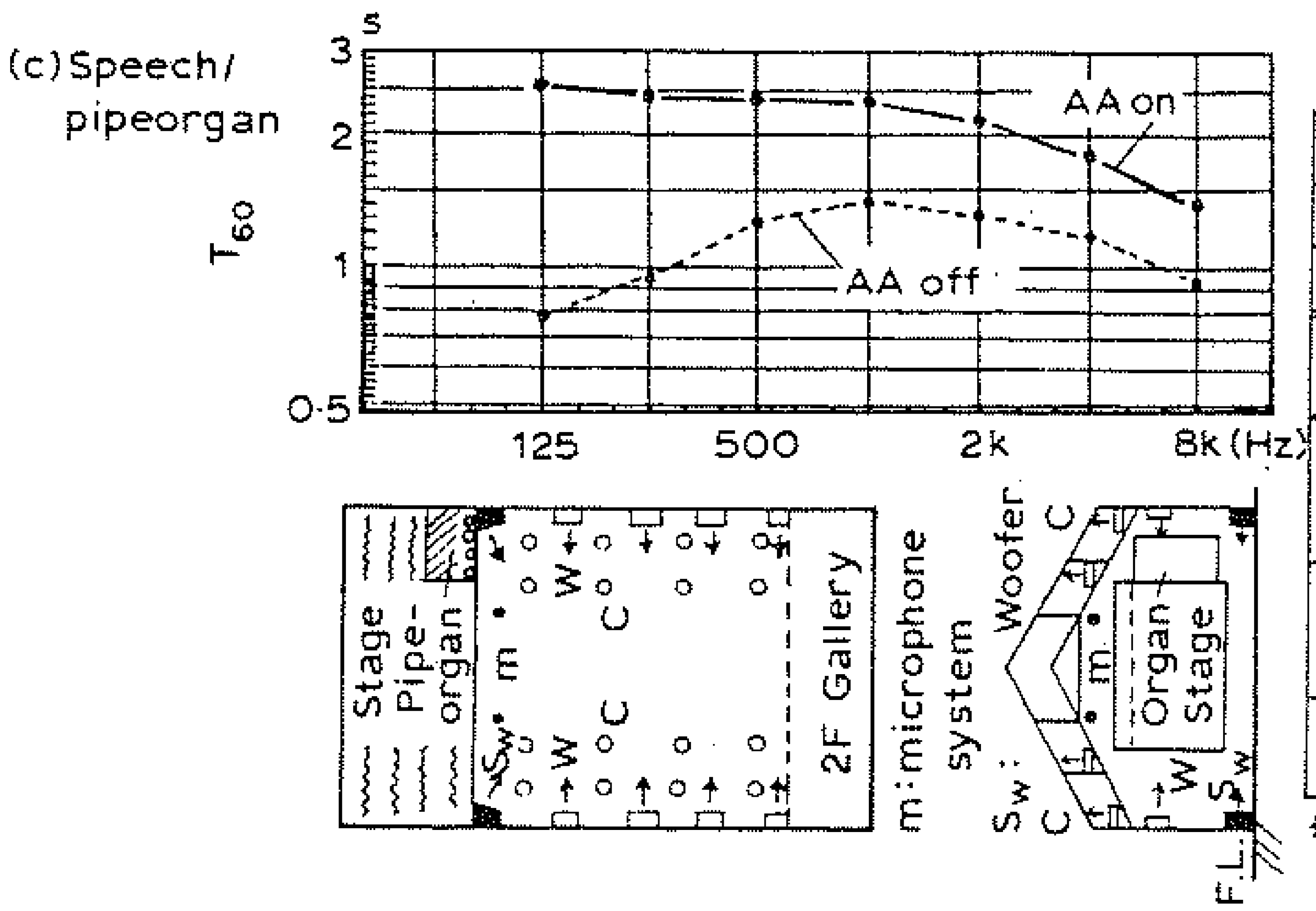
Auditorium	
V	4900 (m^3)
S	3800 (m^2)
N	1200 seats
	AA off / AA on
T_{60}*	off: 1·3 s
	on: 2·0 s
LE*	off: 15 (%)
	on: 25 (%)
ΔL*	1·5 (dB)

*average (250–2kHz)



Multi-use hall	
V	1845 (m^3)
S	1056 (m^2)
N	200 seats
	AA off / AA on
T_{60}*	off: 1·4 s
	on: 2·0 s
D_{50}*	off: 47 (%)
	on: 42 (%)
ΔL*	1·8 (dB)

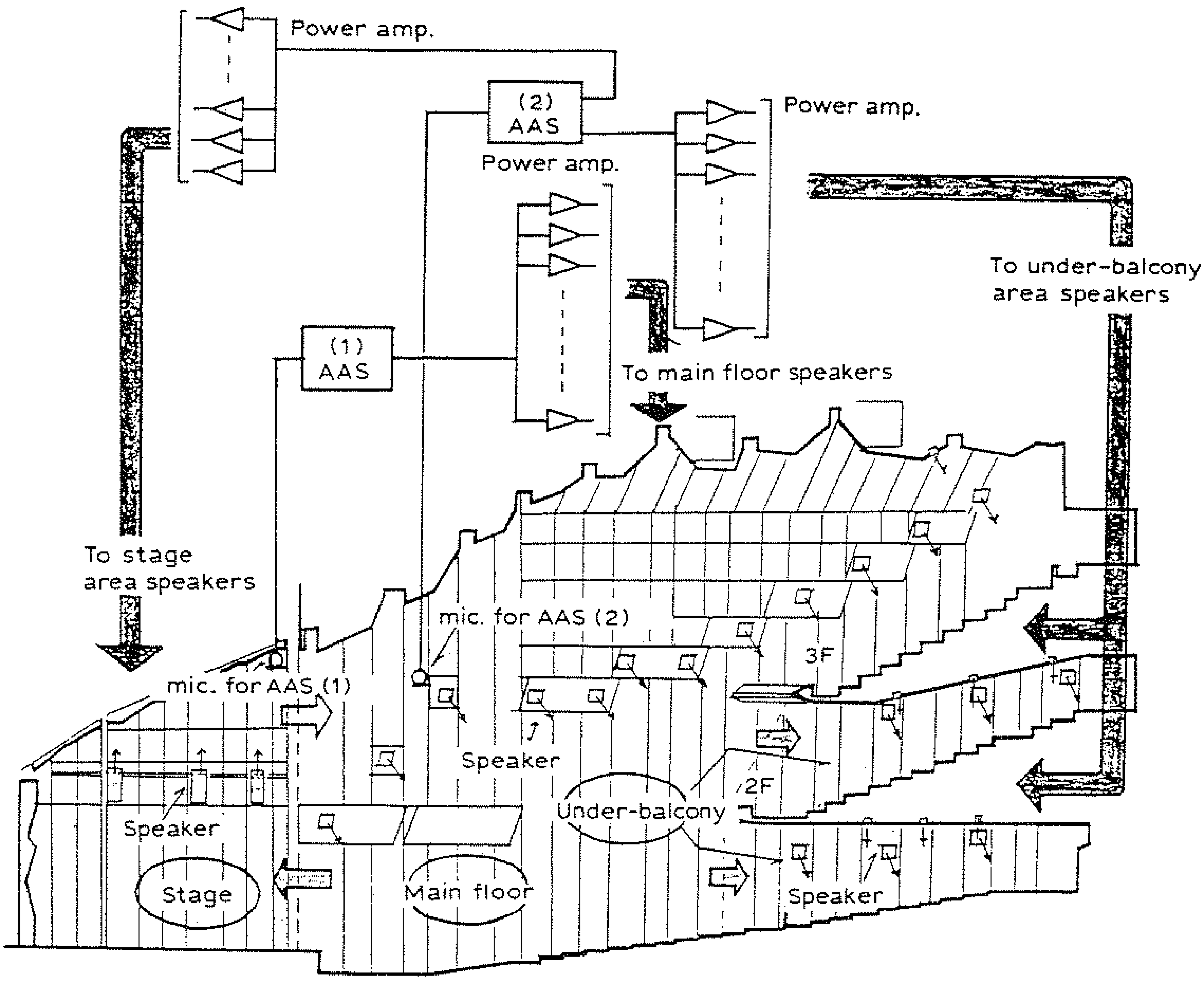
*average (250–2k)



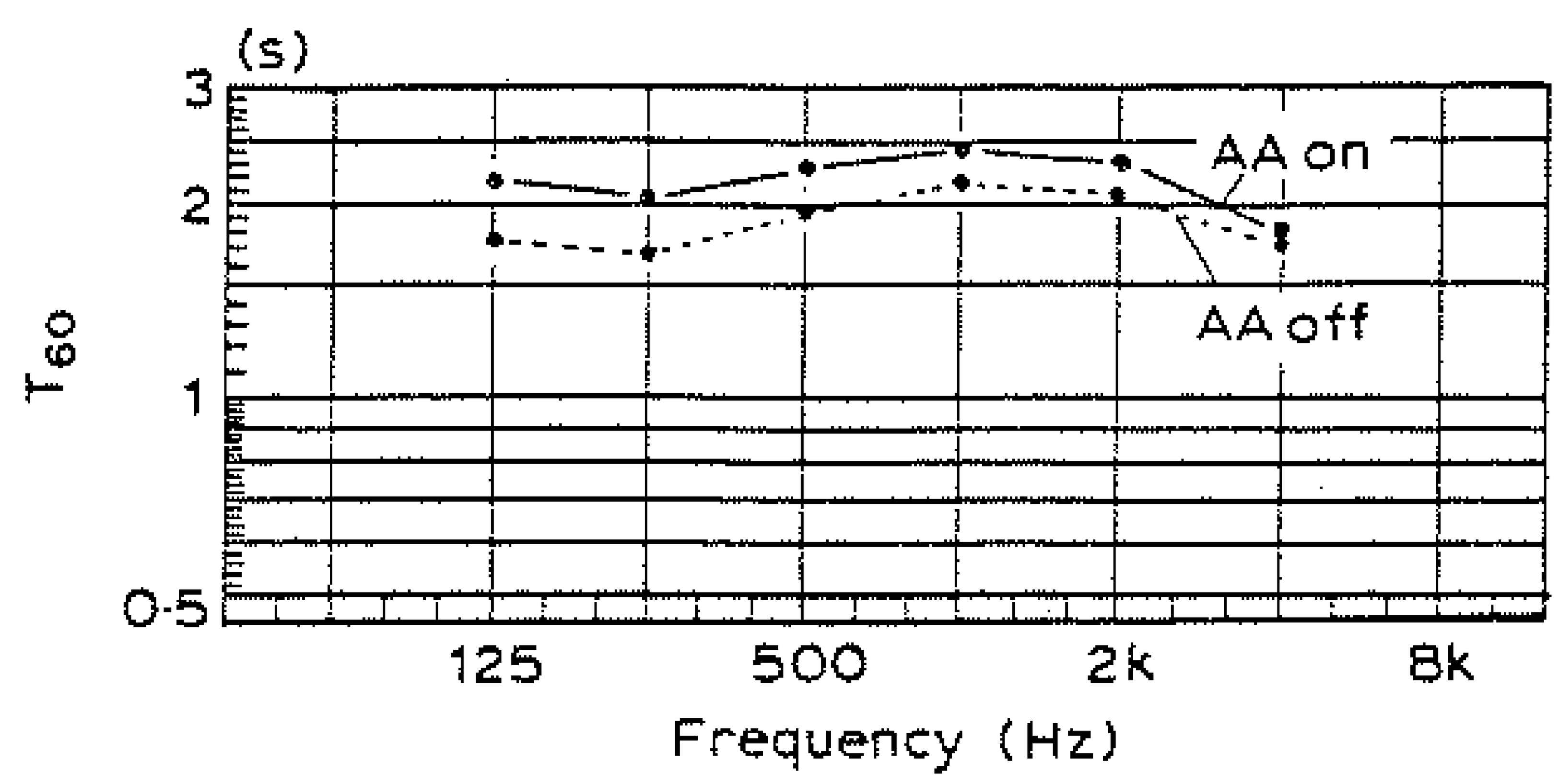
Chapel	
V	~3000 (m^3)
S	~1500 (m^2)
N	~1000 seats
	AA off / AA on
T_{60}*	off: 1·3 s
	on: 2·9 s
D_{50}*	off: 47 (%)
	on: 40 (%)
ΔL*	1·7 (dB)

*average (250–2k)

Fig. 12. Actual examples of AAS implementation, where LE, D_{50}, T_{60} and ΔL are Lateral Efficiency, Deutlichkeit, reverberation time and level difference with AAS on and off, respectively.



AAS (1); Sound energy exchange system for stage and main floor area.
AAS (2); Sound energy exchange system for main floor, under-balcony and stage area.



V	23601 (m^3)
S	7998 (m^2)
N	3012 seats
	AA off / AA on
T_{60}*	off : 2·0 s
	on : 2·3 s
D_{50}*	off : 41 (%)
	on : 33 (%)
ΔL*	1·0 (dB)

*average (250–2k Hz.)

Fig. 13. Actual example of AAS for sound energy exchange between spaces for stage, main floor seats and under-balcony seats in the multi-purpose hall, mainly used for speech and classical music.

(2) Control of early reflections and perceived room size by effective use of FIR filters as a reflection cluster unit.
(3) Digital control of the entire system through the use of a CPU, as well as memorization and reproduction of a number of reflection patterns and operation modes with respective $G_c(\omega)$.

Through the consideration of these items, conditions of *stability*, *naturalness* and *controllability* are each satisfied. Examples of AAS installations are shown in Fig. 12. Details of the system differ from hall to hall depending on their uses and dimensions, although it is basically configured from the hardware as shown in Fig. 1. The example in Fig. 12(a) was designed to meet the requirements for both lectures and pipe-organ concerts, which are poles apart in terms of the T_{60} recommended. Also, LE can be varied no less than 10% with this system. In the system shown in Fig. 12(b), it is possible to select nine patterns in all which differ in 'liveness' and 'virtual room size' depending upon the types of performance. This extends over methods (a) and (b) in Fig. 11 for varying T_{60}.

Thus, through these two possibilities, T_{60} can be extended a substantial amount even at low frequencies as shown in Fig. 12(c).

The last example shown in Fig. 13 is the Main-hall in Shirotori Century Plaza completed in July 1989 at Design EXPO '89, Nagoya, Japan, as one of the permanent buildings. Because of its large capacity (3012 seats), it was determined at the beginning of the design process to employ AAS to recover the expected lack of sound volume and reverberation as well as to improve the under-balcony seat conditions at classical music concerts. The AAS was designed on the basis of the idea of an 'energy exchanging system' composed of two separate portions of four independent channels as in Fig. 13. At the series of opening concerts, sufficient loudness and reverberation were confirmed under the conditions of a full audience.

So far, six projects with AAS have been completed and another six major projects are now underway. On average, when the system is operated, the reverberation time T_{60} can be extended by approximately two times and the loop gain increased by 2 dB. Of course, this is all within the range of naturalness of sound. Since T_{60} itself can be increased rather simply through the method of increasing the virtual room volume as in Fig. 11(2), strict expression concerning this point has no meaning.

7 OTHER IMPORTANT ITEMS CONSIDERED

Discussions have been presented on AFC technologies with special emphasis on A-SF, with its operation theory. However, in the implementation of the design of these systems, a number of matters remain to be considered. These are discussed briefly in the following sections.

7.1 Tools and methods for adjustment

As discussed earlier, there are two basic ideas regarding adjustment of loop gain $G_c(\omega)$ in an A-SF system. One is to regard a precise structure of $G_c(\omega)$ on the frequency axis aimed at fine tuning of this structure. The other attempt is to reduce the peak level of $|G_c(\omega)|$ with respect to an average value for smoothing out the response curve and for a stable increase in $|G_c(\omega)|$. On the point of control at low frequencies where room modes are clear and isolated, the former (frequency independent system) seems to have an advantage over the latter (broad-band system), whilst on the point of stability against any kind of changes in the sound field condition, or of efficiency of hardware utility, the latter has an advantage. At any rate, one has to adjust the system for optimization.

It is rather easy to measure $|G_o(\omega)|$ or $|G_c(\omega)|$ during adjustment of the system. A unity gain buffer amplifier is inserted in a part of the feedback loop, and a sweep generator or pink noise generator is connected to the input terminal while a level recorder or FFT analyser is connected to the other end to record/observe the response curve of $G_o(\omega)$ or $G_c(\omega)$ as shown in Fig. 14. In the same way, T_{60} can be measured with the loop open and closed. However, what is needed most is a specific technology for the optimization of each parameter of the FIR filter. In the case of an n-element filter, the degree of freedom $2n$ is given. In order to determine these parameters reasonably from various viewpoints such as reflection structure on the time axis, smoothing of loop gain $|G_c(\omega)|$, T_{60} and its frequency dependence,

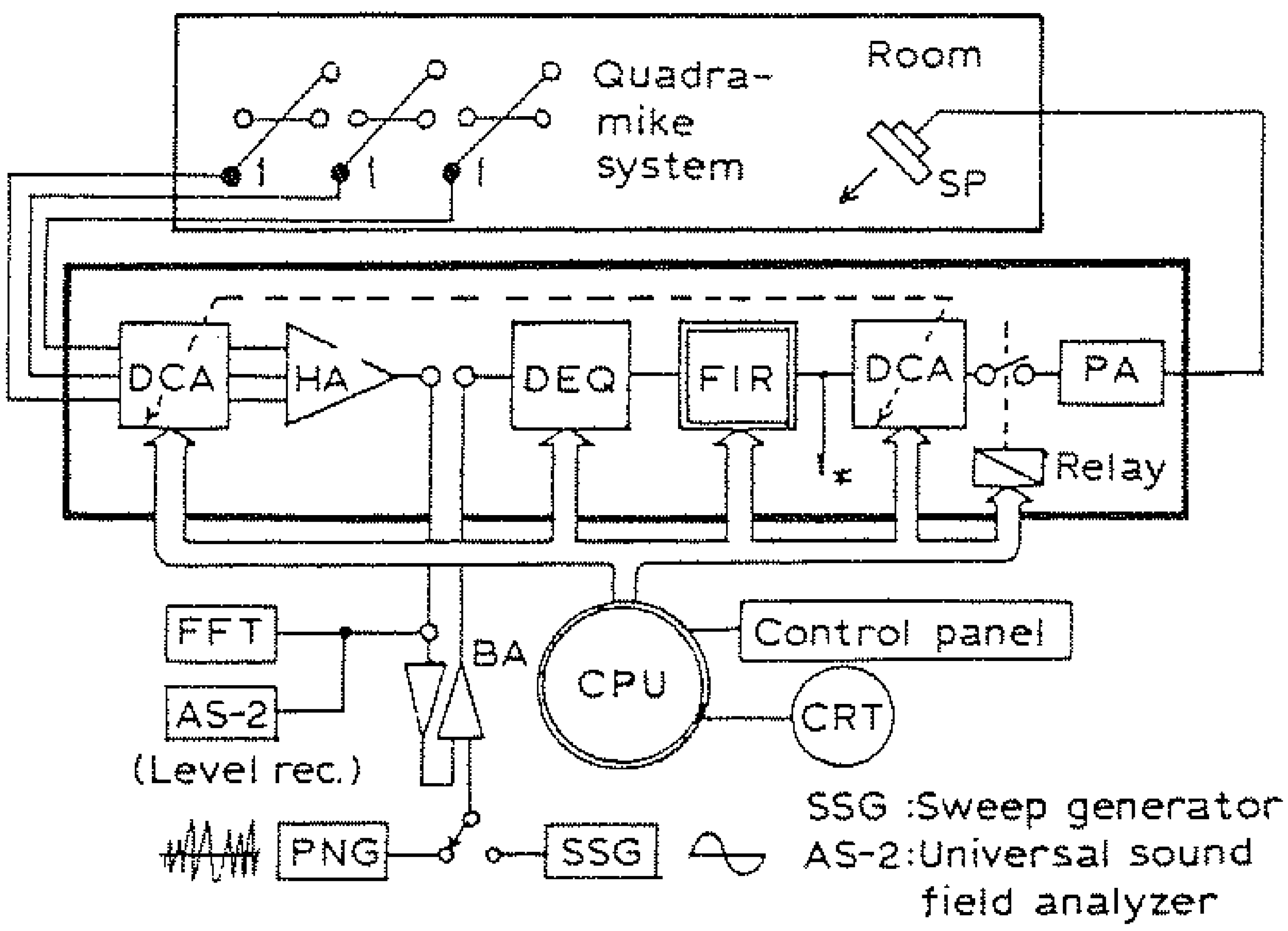


Fig. 14. Block diagram for measurement of T_{60} and $|G_c(\omega)|$ for adjusting AAS hardware.

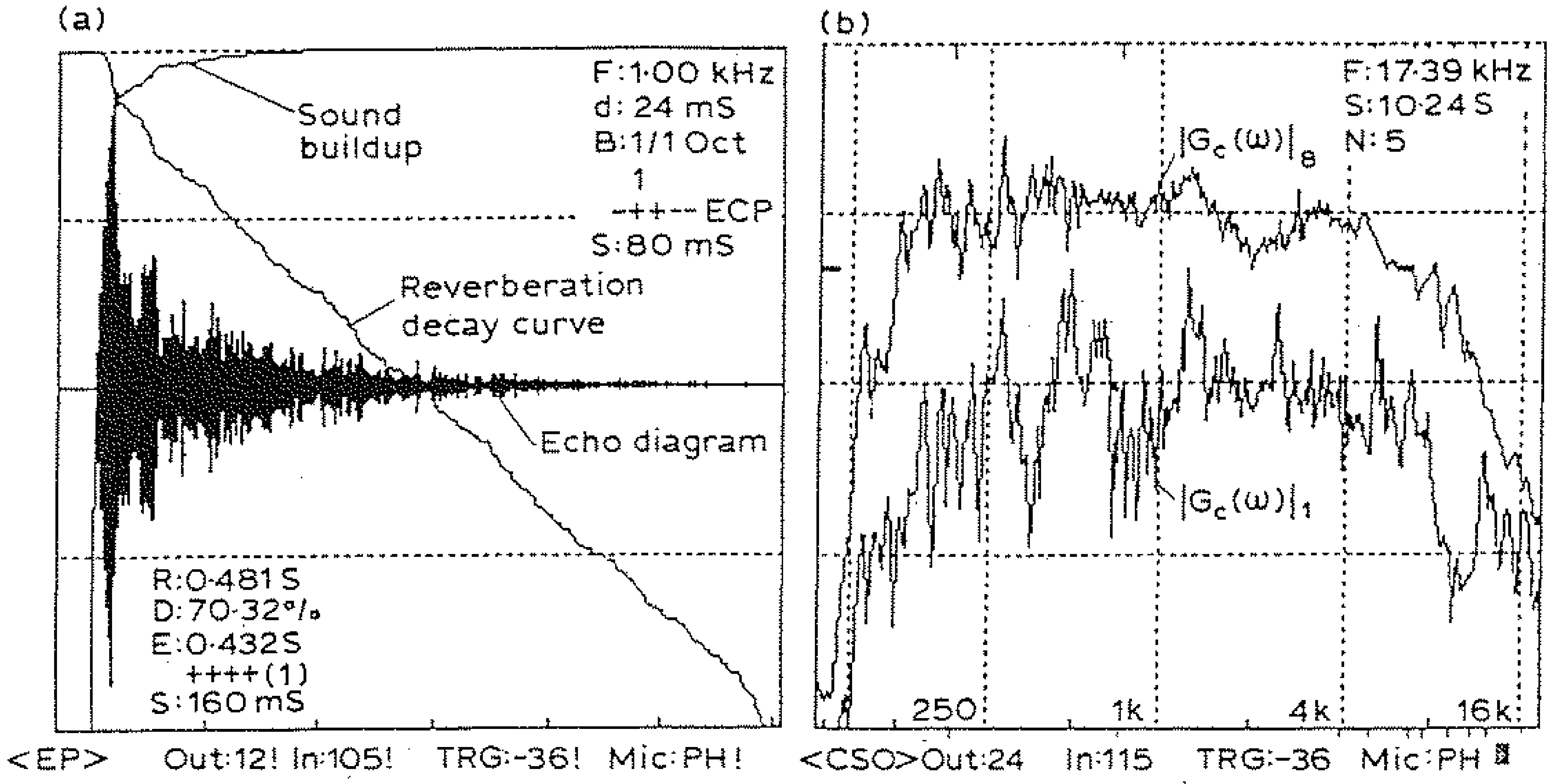


Fig. 15. Examples of output data of monitoring/adjusting devices for AAS. (a) space-ensemble average decay curve with its echo diagram. (b) Loop gain $|G_c(\omega)|$ of a single channel and total system.

perceived naturalness of sound margin against instability, etc., a vast accumulation of technological knowhow and experience are required, as well as a full range of adjustment tools suited for each type of method. One example of a measurement tool is mentioned. This is the sound field measuring instrument built into the system rack for adjustment and monitoring of the system of AAS. This is based on the squared impulse response method developed by Schroeder[16] and is capable of measuring T_{60} with a reverberation decay curve and a response curve of $G_o(\omega)$ or $G_c(\omega)$. Examples of its output are shown in Fig. 15.

7.2 Monitoring the sound field condition

Strictly speaking, the conditions of the sound field of the auditorium change from time to time, due to the entry of the audience, changes in temperature and humidity, etc. It is desirable to monitor the loop gain $|G_c(\omega)|$ continuously without affecting the source signal during the concert. As an example, the method proposed by Behler *et al.*, is shown in the literature.[17] In this system, the main and supplementary microphones are used to separate the direct sound from the background components coming into the microphones, thus enabling continuous measurement of the loop gain.

However, this is not effective for use in A-SF, since instability could occur even when no input signal exists. A microphone placed very close to the sound source could be another non-acoustic problem. Thus, a very large number of other problems remain unsolvable. No satisfactory methods have yet been developed. In order to respond accurately and rapidly to changes in the sound field, it is necessary to realize the following items:

(1) *Continuous measurement* of $|G_c(\omega)|$ without affecting the source signal.
(2) Measurement of $|G_c(\omega)|$ *under the condition that no source signal exists.*
(3) *Rapid control of* $G_c(\omega)$, fast enough to capture the growth of instability.
(4) *Response to a sudden input* with excessively large level.

As for item (4), the idea of an automatic limiter could be helpful as employed in MCR and AAS. Through this consideration, the signal does not have to be lowered unnecessarily at each stage of the system, which expands the dynamic range of the system.

One possibility to meet all the requirements above is the introduction of adaptive signal processing which uses an adaptive filter. Research has only just begun on applications to AFC, but if a suitable design is worked out, this technology offers the following advantages:

(1) *Minimizing the differences or changes in sound field conditions.*
(2) *Release from strict adjustment*, or even from adjustment itself, of the system.
(3) *Deliverance from the worry of instability* or coloration.
(4) *Monitoring the operating conditions* including the loop gain.

Item 1 suggests the possibility of a rehearsal under the same reverberation condition as during the actual orchestra concert with full audience. Also, since the risk of instability is eliminated, $G_c(\omega)$ can be set higher and thus the control range of T_{60} is expanded. Although the detection of processing errors remains itself a problem to be studied, the first hurdle would be cleared fairly easily by composing the FIR filter, each parameter of which can be processed at a relatively low rate, say, 1 ms. That is, ordinarily, in an A-SF system in an auditorium, the locations of the speakers and microphones are fixed so that the speed of changes in the sound field is of the order of several seconds at most. The loop gain $G_c(\omega)$ would be corrected before severe instability occurs. As soon as the growth of instability is detected, appropriate signal processing will be executed within about 1 ms, which will not bring any major trouble. One example of various approaches of thinking for an A-SF system provided with adaptive filters is shown in Fig. 16. In general, the point of this technology is the development of a powerful method for continuous measurement of $|G_c(\omega)|$ and for optimization of FIR parameters.

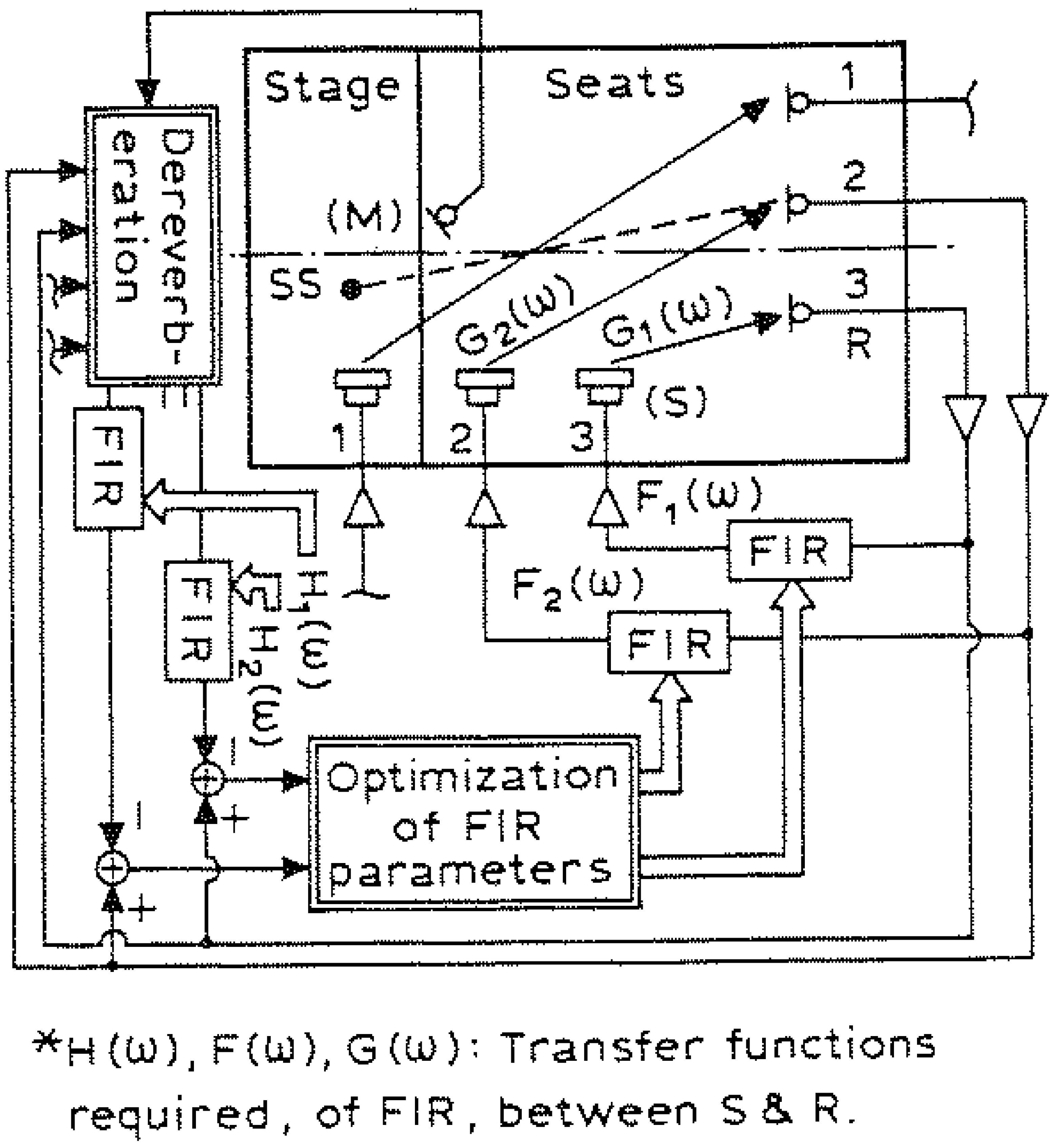


Fig. 16. An example of an approach for the development of an A-SF system provided with adaptive FIR filters.

7.3 Digital signal processing units for AFC

As technical background supporting the advances of AFC technologies, digital signal processing and accompanying hardware technologies have played a leading role. The real time FIR filter, the basic device used in A-SF for performing convolution of the reflection data, has the maximum number of reflections for convolution, at present, within the range 30–100, but it appears that, in a few years, convolution units with thousands of elements will be available on the market. Furthermore, though accompanied by limitations on the physical size and writing speed for reflection data, it has been reported that a prototype of an 'all-sample-type real time convolver' has been completed, which is capable of convoluting the impulse response $r(x)$ at each sample point exactly. In this prototype, approximately 2 s of $r(x)$ data for real time convolution is possible.

Through the writing of reflection data or $r(x)$ data in the ROM/RAM areas of these units as in Fig. 17(a), acousticians or designers of A-SF systems can administer various controls within the domains of *time*, *space* and *frequency*.

In Fig. 17(b) is a digital version of the so-called 'L/D-EQ' Matrix. Both level and delay time of all the cross-points ($m \times n$) for m input and n output signals can be set or altered in 0·357 dB and 1 ms steps, respectively, and these can be accessed in virtually real time (each 1 ms). In addition, the output stage is provided with n parametric equalizer units. This device can be

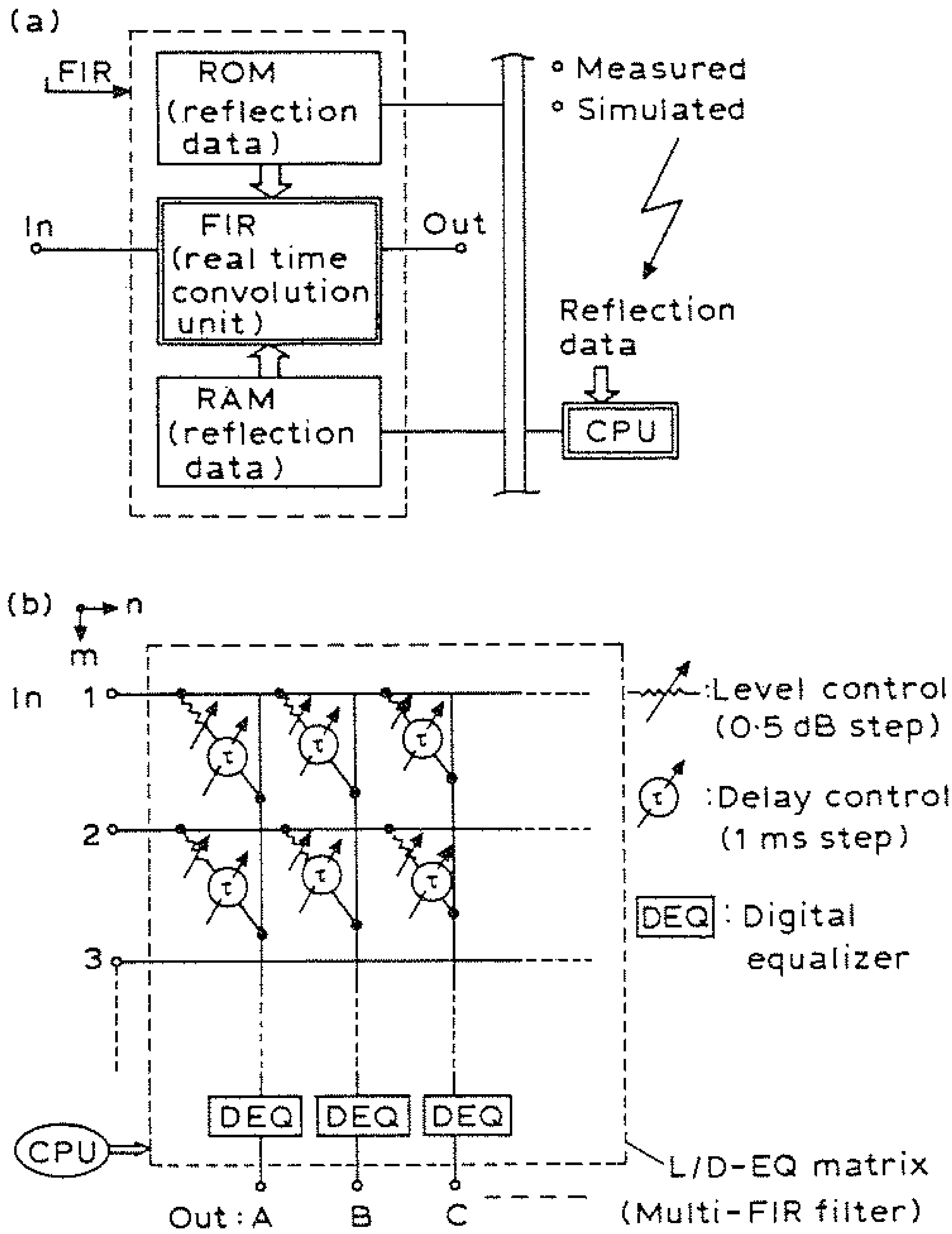


Fig. 17. Examples of digital signal processing unit for A-SF. (a) Real time convolution unit with ROM/RAM. (b) Level/delay-EQ matrix with $m \times n$ cross-points.

used for many purposes, such as for a multi-channel and multi-element FIR filter in the AFC field, or for a single channel FIR filter with $m \times n$ elements as well as for adaptive filters described above.

They are used, however, mainly for sound localization control as in VISION-stage or Delta-stereophony,[7,8] etc., in theatres, as well as for output stage controllers in the SR system of multi-purpose halls and for channel dividers in speaker systems. The function of this device is simple and primitive, and therefore, it is safe to say that it is a device with unexpected potential.

8 CONCLUDING SUMMARY

The author attempted to limit the discussion to the basic ideas used in A-SF, one of the fields of AFC. One topic we hear a lot about is the problem of a performer's allergy to electricity. It is of course necessary that vigilance be exercised in the discussion of production methods for the announcement of the operation of AFC. However, the public trend to such concerns is gradually tending to weaken. A few initial successes would drive away such fears. On the other hand, engineers who are involved in these fields must not be lax in their efforts in developing new ideas. The present subject in developing an A-SF system is the achievement of both *naturalness* and a *suitable control range*. The essence of AFC is the aim of achieving, by electrical means, effects similar to those experienced in ordinary architecture.

On the other hand, a way of thinking also exists that the acoustic feedback loop should be eliminated because of its fear of instability by, for example, synthesizing all required signals by an electrical process in advance. But this is a misjudgement. By doing that, A-SF system designers would lose many possibilities such as controlling the entire area in the enclosure, the naturalness of sound, the uniformity of the 'assisted' sound field, etc., as well as the possibility of controlling both reverbance and loudness.

ACKNOWLEDGEMENT

The authors thank the members of the Yamaha Acoustic Research Laboratories who extended their fullest cooperation in the preparation of this paper.

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