LETTER

Design of Time-Varying Reverberators for Low Memory Applications

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SUMMARY Development of an artificial reverberator for low-memory requirements is an issue of importance in applications such as mobile multimedia devices. One possibility is to use an All-Pass Filter (APF), which is embedded in the feedback loop of the comb filter network. In this paper, we propose a reverberator employing time-varying APFs to increase the reverberation performance. By changing the gain of the APF, we can increase the number of frequency peaks perceptually. Thus, the resulting reverberation sounds much more natural, even with less memory, than the conventional approach. In this paper, we perform theoretical and perceptual analyses of artificial reverberators employing time-varying APF. Through the analyses, we derive the degree of phase variation of the APF that is perceptually acceptable. Based on the analyses, we propose a method of designing artificial reverberators associated with the time-varying APFs. Through subjective tests, it is shown that the proposed method is capable of providing perceptually comparable sound quality to the conventional methods even though it uses less memory.

key words: *all-pass filter, reverberator, time-varying*

1. Introduction

Artificial reverberators are key elements in three-dimensional spatial audio reproduction. In recent multimedia contexts, an artificial reverberation algorithm for low-memory applications is an important issue in applications such as hand-held multimedia devices. Time-varying algorithms [2] have often been considered to solve the memory issue. There are several reasons why one might want to incorporate time variation into a reverberation algorithm. One motivation is to reduce coloration and fluttering in the reverberant response by varying the resonant frequencies. Another reason is that the use of a time-varying algorithm provides an opportunity to reduce the memory size of the reverberator. There are several ways to add time variation to an existing algorithm: modulating the lengths of the delays [2] or varying the coefficients of the feedback matrix in the reference filter while maintaining the energy conserving property [3].

In this paper, the method using APFs whose responses are continuously varied in time is represented to reduce memory loads of the reverberator. Time variation of the APF [1] is often implemented by continuously changing the APF gain, which results in a change in the location of the frequency peaks in the frequency response. This process makes it possible to avoid a build-up of resonances at specific frequencies. Thus, the resulting reverberation sounds much more natural with less memory than the conventional approach. Also, in this paper, we investigate theoretical and perceptual aspects of reverberators employing time-varying APFs. Based on the results, we calculate the shift of frequency peaks caused by the change in the group delay of the APF. The analysis is to find a perceptually acceptable degree of phase modulation caused by varying the APF gain. Based on the analyses results, we propose a design algorithm for an artificial reverberator based on time-varying APFs. Subjective listening tests were conducted, and the results, together with memory usage, are compared with the conventional algorithm.

2. Reverberator Using Time-Varying APF

Reverberator with embedded APFs can be used to increase time density in a condition of reasonable frequency density [6]–[8]. The APFs inside the feedback loop create a build-up of echo density over time. But this method has a problem of building up resonances at specific frequencies. To alleviate this problem, we propose that the gain of the APF be continuously changed. This leads to variation of the phase response of APF and changes the frequency response of the reverberation system. A block diagram of a Feedback Delay Network (FDN) composed of time-varying APFs in the feedback loop is depicted in Fig. 1. Each delay channel

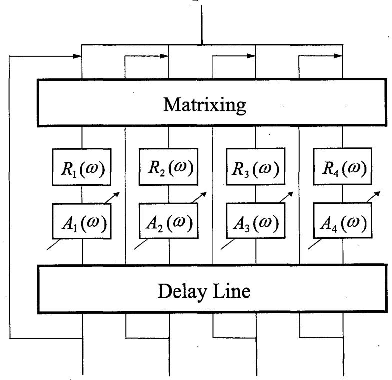

Fig. 1 Feedback delay network with time-varying all-pass filters in the feedback loop.

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contains a delay line, a low-pass filter $R_i(\omega)$, and an all-pass filter $A_i(\omega)$. The channel outputs are mixed by matrixing and fed back to the channel inputs. The APFs in the feedback loops are described by a transfer function:

$$A_i(z) = \frac{-g_i + z^{-M_i}}{1 - g_i z^{-M_i}} \tag{1}$$

where M_i denotes the internal delay of the filter. The phase response of the APF can be conveniently changed by varying the gain value g_i. Variation of APF gain results in changes of the echo density in the time response as well as the location of frequency peaks in the frequency response.

3. Analysis of Time-Varying APF Based Artificial Reverberator

According to psychoacoustics theory in chapter 7.2.2 [9], the frequency variation must be set to below the Just Noticeable Frequency Variation (JNFV) to prevent the click sound. Therefore, the control of frequency shift by the APF should meet this requirement, especially at low frequencies. This section summarizes the mathematical aspects on how the system response is affected by the phase modulation of the APF. Specifically, the shift of frequency peaks caused by the change in the group delay profile of the APF is calculated. Knowing that the maximum variation of frequency peaks for a given APF gain g, we can control the time variation of the APF below the JNFV to make clicks imperceptible. For the easy explanation, we consider a single-channel FDN composed of a time-varying all-pass filter, as shown in Fig. 2. The frequency response of the FDN in Fig. 2 is described using the transfer function of APF $A(z, n)$ as

$$H(z, n) = \frac{\alpha z^{-D} A(z, n)}{1 - \alpha z^{-D} A(z, n)} \tag{2}$$

where α is the feedback gain of the FDN, and D denotes the length of the delay channel. If we denote the phase response of the time-varying APF as $\theta_A(\omega, n)$, the frequency response of the APF can be abbreviated using its phase response as $\hat{A}(\omega, n) = e^{j\theta_A(\omega,n)}$. Now, the frequency response of the FDN can be obtained from Eq. (2) as

$$\hat{H}(\omega, n) = \frac{\alpha e^{-j\omega D} e^{j\theta_A(\omega,n)}}{1 - \alpha e^{-j\omega D} e^{j\theta_A(\omega,n)}}. \tag{3}$$

The location of the frequency resonance (peak) can be easily found from Eq. (3) as

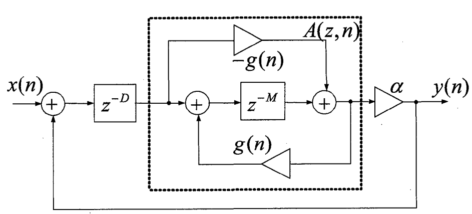

Fig. 2 Feedback delay network with time-varying all-pass filters in the feedback loop.

$$\omega D - \theta_A(\omega, n) = 2k\pi, \quad k = 0, 1, \cdots, D + M - 1 \tag{4}$$

where M denotes the internal delay of the APF. On the other hand, the phase response of the APF is given by [10]

$$\theta_A(\omega, n) = -M\omega + 2\tan^{-1}\left(\frac{-g(n)\sin(M\omega)}{1 - g(n)\cos(M\omega)}\right) \tag{5}$$

where $g(n)$ represents the APF gain at a time index n. Using Eq. (4), Eq. (5) can be rewritten as

$$\omega \times (D + M) - 2\tan^{-1}\left(\frac{-g(n)\sin(M\omega)}{1 - g(n)\cos(M\omega)}\right) = 2k\pi \tag{6}$$

We can estimate the shift in frequency peaks caused by the APF by solving Eq. (6) with respect to ω. Unfortunately, Eq. (6) doesn't have a closed form expression for ω. Thus, we define a new parameter:

$$\chi(\omega, n) = \tan^{-1}\left(\frac{-g(n)\sin(\omega M)}{1 - g(n)\cos(\omega M)}\right) \tag{7}$$

From Eq. (7), it can be noted that degree of frequency variation changes corresponding to $g(n)$, the maximum frequency variation is obtained at maximum negative value $g(n) = -g_p$ or maximum positive value $g(n) = g_p$. If the delay length is sufficiently large, the range of the loop gain doesn't need to be considered. Assuming that $\chi(\omega, n)$ at frequencies $\omega_k = 2\pi k/(D + M), k = 0, 1, 2, \cdots, D + M - 1$ are known, we approximate the locations of the frequency peaks as

$$\hat{\omega}_k(n) \approx \omega_k + \frac{2\chi(\omega_k, n)}{D + M}, \quad k = 0, 1, \cdots, D + M - 1 \tag{8}$$

where ω_k represents the k-th real resonant frequency of FDN composed of a delay line with a length of $D + M$, and $\hat{\omega}_k(n)$ represents the corresponding estimated resonant frequency at the time index n due to the presence of a time-varying APF. If $g(n)$ is zero, we have $\hat{\omega}_k(n) = \omega_k$. The approximation in Eq. (8) implies that the degree of the frequency peak shift is determined solely by the parameter $\chi(\omega_k, n)$. For a nonzero gain $g(n)$, the k-th frequency peak is shifted by $2\chi(\omega_k, n)/(D + M)$. Since $\chi(\omega_k, n)$ is a function of the APF gain $g(n)$, the degree of the frequency peak change is determined by $g(n)$. If $g(n)$ is continuously changed, say, from $-g_p$ to g_p, the location of frequency peaks is also continuously moved, which will instantaneously increase the frequency density. It should be noted that $\chi(\omega_k, n)$ shows maximum when $g(n) = g_p$, and minimum when $g(n) = -g_p$. To predict the maximum frequency density attainable using the time-varying APF, we need to estimate the maximum possible distance between the frequency peaks. The theoretical maximum variation of a frequency peak can be set by

$$\Delta_{\max} = \frac{2}{D + M}(\chi_{\max} - \chi_{\min}) \tag{9}$$

where $\chi_{\max}$ and $\chi_{\min}$, respectively, denote the positive and negative extremes of $\chi(\omega_k)$ when $g(n) = \pm g_p$. Due to the symmetry of $\chi_{\max}$ and $\chi_{\min}$, we have

$$\Delta_{\max} = \frac{4\chi_{\max}}{D + M} \tag{10}$$

Now, using Eqs. (7) and (10), the theoretical maximum variation of the frequency peaks for a given APF gain can be predicted.

4. Design of Time-Varying Reverberation Algorithm

In designing an artificial reverberator, we need to determine RT_{60}. The modal density of parallel comb filters is expressed as the number of modes per Hz. The modal density for the given delay channels is computed as a sum of delay-channel lengths [5]. Equating the computed modal density with the density of the frequency maxima of a real room [4], we obtain the following relationship between the total length of the delays and the maximum reverberation time we wish to simulate [11]:

$$\sum_i \tau_i = R_m > R_f \approx \frac{RT_{60}}{4} \tag{11}$$

where τ_i denotes the length of i-th delay channel in seconds, R_m denotes the number of modes per Hz, RT_{60} denotes the desired reverberation time and R_f denotes the minimum density of the frequency maxima (modal density) according to the statistical model for late reverberation. Eq. (11) specifies the minimum amount of total delay required according to the reverberation time. Based on analysis and perceptual requirements, we propose a design algorithm for the artificial reverberator implemented in a FDN with embedded time-varying APFs in the feedback paths. First of all, the minimum modal density R_f for natural sound is calculated using Eq. (11). Using the same equation, we can set the minimum length of the delay channel R_m. Using R_m, the length of delay channel D is initialized as $R_m > D_0 + M$ where M is fixed. We set D_0 to $R_f/2$. Next, $\Delta_{\max}$ is calculated using Eq. (10). If $\Delta_{\max}$ is lager than JNFV, the length of the delay channel D is increased by 1 msec, and then returns to the $\Delta_{\max}$ calculation procedure. This process is repeated until the desired frequency density is obtained by frequency peaks shift, while the frequency shift does not exceed the JNFV. If the required memory size exceeds the memory capacity of the system, the JNFV is raised and returned to the above procedure. To prevent overlapping of frequency peaks and void flutter,the lengths of the delay channels are chosen incommensurately. The proposed algorithm is summarized in Table 1.

Table 1 Pseudo code of the proposed algorithm.

Initialization
Set the internal delay M of the time-varying APF
Determine reverberation time RT_{60}
Compute minimum modal density R_f using Eq. (11)
Initialize D_0 (as small as possible)
Repeat:
a.Calculate $\Omega_{\max}$ using Eqs. (7) and (10)
b.If $\Omega_{\max} > JNFV$
$D_{i+1} = D_i + \Delta D (\Delta D = 1\, m\,\text{sec})$
Go to a
else
Break
End of if
c.If $D >$ Memory usage of the system
$JNFV = JNFV + \Delta JNFV\ (\Delta JNFV = 0.5\, Hz)$
Go to a
End of if
End of repeat

5. Experiments

Performance of the proposed algorithm was evaluated via subjective listening tests using various input samples. In the tests, we considered the following 4 reverberation cases: large memory without the time-varying algorithm (A), large memory with the time-varying algorithm (B), small memory without the time-varying algorithm (C) and small memory with the time-varying algorithm (D). All the reverberation algorithms were based on the FDN structure. Thus, there was no structural dissimilarity between the algorithms. Our goal is to prove that algorithm D can achieve a reverberation quality that is equivalent to algorithm A employing more memory. To this end, we conducted preference tests. For demonstration of proposed method about various cases, the tests were done for three different RT_{60} cases (0.5, 1, 2 sec). The memory size used in each of the 4 algorithms are listed in Table 2. In the tests, 10 audio clips were presented, and twenty listeners majoring in audio signal processing participated. The sources consisted of 3 groups: Group 1 (single instruments 1-3), Group 2 (multiple instruments 4-6), and Group 3 (complex music 7-10). All these clips were recorded digitally at 44.1 kHz without reverberation. The audio samples were 3-to-9 seconds long. The length of the audio sample might affect the perception of reverberation. However, since the tested audio samples didn't contain noticeable pauses, 3 to 9 seconds lengths were long enough to provide consistent impressions of reverberation to the listeners. Results also indicated that the listeners were not affected by the length of the audio sample. For each sample, listeners were asked to note which one they preferred. The results of the preference tests are shown in Table 3. Those are the average values for three different RT_{60} cases. The results were consistently similar for the three RT_{60} cases. Thus, the results were averaged over the cases. From Table 3, we can infer that listeners preferred the reverberation algorithms, which contain more memory and applies

Table 2 Memory usage (K words) of the reverberators for different RT_{60}.

$RT_{60}(sec)$/**Algorithm**	**A**	**B**	**C**	**D**
0.5	5.5	5.5	3.7	3.7
1	11	11	8.8	8.8
2	22	22	14.9	14.9

Table 3 Percentage of listeners who preferred any one algorithm.

	Group 1	**Group 2**	**Group 3**
A preferred over C	100	100	98
B preferred over D	90	89	85
B preferred over A	70	69	68
D preferred over C	90	87	88

Table 4 Percentage of listeners that were not able to find a difference between algorithm A and algorithm D.

Group 1	Group 2	Group 3
85	95	98

the time-variance. This procedure was needed to judge if listeners have the ability to differentiate the natural reverberation algorithm or not. Finally, we conducted blind ABX tests [12] with algorithms A and D. Table 4 shows the results. The values in Table 4 are also averaged ones over the three RT_{60} cases. The results show that most of the subjects could not differentiate the reverberation algorithms. We could note that even those who were able to perceive the difference between the algorithms said that the two reverberation types were really hard to differentiate. From the experimental analysis, it was proved that algorithm D could achieve a reverberation quality that was equivalent to algorithm A, which contained more delays. In addition, most of the listeners agreed that algorithm D had a smooth decay. This is because the time-varying algorithm reduced coloration and fluttering in the reverberant response by varying the resonant frequencies. Since the memory size of algorithm A was determined using Eq. (11), which is the requirements of natural sounding, it was assumed that algorithm A provides natural sounding reverberations. Naturalness of the reverberation generated by algorithm D was also warranted because the results of subjective tests showed that most of the listeners could not perceive differences between A and D.

6. Conclusions

In this paper, we presented a new design algorithm for a time-varying reverberator suitable for low-memory applications. By calculating the shift of frequency peaks caused by the change in the group delay profile of the APF, the perceptually acceptable degree of frequency variation was derived. Based on the derivation, we proposed a new design method for the reverberator based on an FDN structure with embedded time-varying all-pass filters in the feedback loop. To compare performance, we conducted subjective tests, and memory usages were compared. Experimental results indicated that the use of a time-varying algorithm provided improvement in the perceived characteristics of reverberation and a reduction in memory requirements.

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