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A physically motivated room reverberation enhancement system that is stable in any (dissipative) room: An application of sound portholes

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Prior active room reverberation enhancement systems have typically employed microphones, artificial reverberator filters, and loudspeakers to change the reverberant properties of a room. However, acoustic feedback from the loudspeakers to the microphones has had the potential to drive such systems unstable. To avoid feedback instabilities, we apply passivity techniques from control theory to design a stable room reverberation enhancement system from the ground up. In particular, we employ *sound portholes*, which are special transducers that operate concurrently as microphones and loudspeakers. With these sound portholes, the feedback controller implements a passive connection to virtual acoustic spaces, which are realized using digital waveguide networks. Because the feedback controller models a passive system, it is theoretically stable in any (dissipative) acoustic environment and for arbitrarily large loop gains. As a consequence, the system does not suffer from "ringing tones" at high loop gains in the same way as prior systems have suffered. Furthermore, the system does not need to be re-calibrated if the properties of the room change or even if moved to a whole new room. This method for designing room reverberation enhancement systems may generally result in more realistic reverberant sound because it implements the acoustical features of a system that could exist naturally in the physical world.

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1 INTRODUCTION

As early as 1961, Manfred Schroeder formulated a well-defined technique for processing a prerecorded signal to make it sound more reverberant[9]. He coined the technique *artificial reverberation*, and since then many engineers have sought to devise more realistic sounding artificial reverberators that are employed extensively by the recording industry. Then as early as the 1990's, researchers began developing systems for enhancing real rooms using microphones, artificial reverberators, and loudspeakers to improve their reverberant properties. These properties could be changed on the fly for various applications such as concerts, plays, speeches, and more[8]. Figure 1 shows an example configuration of such a "regenerative" system for a single room. Microphones are employed to sample the sound pressure at a variety of locations (see Figure 1, right). These signals are fed to a feedback delay network (FDN)[11], consisting of delay lines and a mixing matrix, which produces output signals that are fed to a series of loudspeakers (see Figure 1, left).

The system can increase the reverberation decay time of the room by feeding acoustic energy back into the room. However, the acoustic feedback from the loudspeakers to the microphones has the potential of driving the room reverberation enhancement system unstable. Even when stable, certain room modes may exhibit marginal stability and result in "ringing tones." In this case, the room with enhanced reverberation may sound artificial to listeners.

2 FEEDBACK CONTROL

2.1 Revised Bode Stability Criterion

To help provide the reader with some intuition into how feedback can affect the stability of a system, we review some concepts from feedback control. Figure 2 shows a simple single-input single-output configuration. A person speaks into a microphone, which sends an electrical signal to a controller with Laplace-domain transfer function $K(s)$, which drives a loudspeaker accordingly (see Figure 2). For convenience, we also lump the microphone and loudspeaker transducer dynamic responses into $K(s)$. $G(s)$ represents the acoustic feedback path from the loudspeaker back to the microphone. The Revised Bode Stability Criterion states the following:

> A linear and time-invariant single-input single-output closed-loop system is stable if the open-loop transfer function $G(s)K(s)$ is stable and the frequency response of the open-loop transfer function has an amplitude ratio of less than unity at all frequencies corresponding to an angle of $-\pi-n(2\pi)$, where $n=0,1,2,...,\infty$.[4]

We now contemplate two different ways of satisfying the criterion in the context of room reverberation enhancement.

2.2 Considering Only The Magnitude

Given any loudspeaker and any microphone in the system shown in Figure 1, the acoustic delay between the loudspeaker and microphone is relatively large. As a consequence, the frequencies where

$$\angle G(s)K(s)\Big|_{s=j\omega} = -\pi - n(2\pi) \tag{1}$$

are very sensitive to small perturbations in the placement of the loudspeaker and the microphone. In practical contexts where persons, furniture, or other objects may move about the room, it can be infeasible to robustly predict these frequencies. As a consequence, active room reverberation enhancement system designers have typically chosen to ensure that $|G(j\omega)K(j\omega)|<1$ for all ω. This can be realized by reducing the gain of the controller $K(j\omega)$ or by reducing the magnitude of $G(j\omega)$, which can be achieved by adjusting the orientation of the loudspeakers and microphones. For example, the loudspeakers and microphones can be placed on opposite sides of the room as shown in Figure 1. However, these measures also lessen the effect of the room reverberation enhancement.

In response, some researchers have introduced some time-varying approaches to try to skirt by the criterion's requirements[6,8,10]. However, these techniques cause distortion of the amplified signal.

2.3 Considering The Magnitude And Phase

In contrast, the premise of this work is that we should design room reverberation enhancement systems so that they are always guaranteed to be stable, regardless of transducer placement and loop gain. This approach requires us to take the *magnitude and phase* of the transfer functions into account. The criterion implies that if the phase lag introduced by $K(s)$ and $G(s)$ is small enough, then the system shown in Figure 2 will be stable no matter how large the loop gain is. Expressed mathematically, the system is stable if $K(s)$ is *positive real* and $G(s)$ is *strictly positive real*. However, $G(s)$ cannot be strictly positive real if it includes any delay. In other words, in order for $G(s)$ to be strictly positive real, and similarly to keep the phase lag bounded at high frequencies, **the microphone and loudspeaker need to be placed at the same position in space, i.e., they need to be *collocated***[1,5].

A well-collocated microphone and loudspeaker must share the same membrane. Hence, such a device can be constructed using a dual voice coil loudspeaker, in which two electrically isolated voice coils are wound over one another on the same bobbin[2-3]. Beneath a certain "impedance crossover frequency," the voltage V across one coil is proportional to the velocity $\dot{x}$ of the bobbin, and the current i through the other coil is proportional to the force F_C exerted on the bobbin. We like to term such a collocated microphone/loudspeaker transducer a *sound porthole* because it acts like a passive port connecting the acoustical domain to the electrical domain. Any passive elements or simulations of passive elements can be connected to one domain of the sound porthole, and the resulting system as seen from the other domain will also be passive. This property reveals the **power of using mechanical analog systems for deriving control laws**: if an electrical feedback controller operates analogously to any passive mechanical analog system and it is connected to a sound porthole, then the acoustic domain sees a passive acoustic system through the sound porthole.

As a consequence, **if we choose a feedback controller *K(s)* that emulates a passive mechanical analog system, and if it is connected to a dissipative acoustic environment by way of a sound porthole, forming *G(s)*, then the control system is guaranteed to be stable**. No element in the feedback loop is capable of generating energy. In particular, we note that in this ideal configuration, the stability of the control system does not depend on the magnitude of *K(s)*. In theory, the control loop gain may be made arbitrarily large, implying that we can exert significant influence over the acoustical domain by using sound portholes and feedback control laws corresponding to passive mechanical analog systems.

3 ROOM REVERBERATION ENHANCEMENT WITH MECHANICAL ANALOGS

3.1 Basic Implementation

We design a room reverberation enhancement system whose components correspond to passive mechanical analogs. As a consequence, the system is guaranteed to be stable, regardless of the placement of the transducers or how far the control loop gain is increased. Consider the configuration shown in Figure 3, where sound portholes are placed along the walls. We use a loudspeaker schematic symbol overlaid with a microphone schematic symbol to represent a sound porthole (see Figure 3). Each sound porthole is connected to a feedback control system that emulates a passive mechanical system consisting primarily of virtual, strongly coupled vibrating strings and connections to the other sound portholes (not shown).

Although it is probably more advantageous to implement a control system that passively interconnects the sound portholes, for simplicity, we assume in this paper that each sound porthole is controlled independently. One reason for this is that so far we have experimented with only a system involving a single sound porthole. Figure 4 depicts a sound porthole, whose diaphragm moves vertically with velocity $\dot{x}$ and is connected to N vibrating strings. We choose to use the mechanical analog of transversely vibrating strings instead of acoustic waveguides because it is easier to draw, but an equivalent analog consisting of longitudinal pressure waves in acoustic waveguides could be made. The first vibrating string has transverse wave impedance R_1, the velocity wave from the first string impinging on the sound porthole is W_1^+, while the velocity wave traveling away from the sound porthole on the first string is W_1^- (see Figure 4). Note that the system shown in Figure 4 is entirely passive—no components generate any energy.

The signal flow graph for a digital feedback controller emulating the same system is shown in Figure 5. The single-sample delay labeled z^{-1} for digital feedback control is employed to avoid a delay-free loop. The ith string is modeled by a digital waveguide with a time delay of N_i samples and a gain factor of l_i. Typically, $|l_i|\approx 1$ to encourage long reverberation decay times but $|l_i|\leq 1$ for passivity. $L_i(z)$ is a digital Schur filter that models the change in shape of a wave as it reflects off of the end of the string[11]. It is best to make $L_i(z)$ have a lowpass characteristic to prevent the single-sample delay z^{-1} from destabilizing the control system. The controller is an example of a *digital waveguide network* as known in the field of artificial reverberation[11]. Since the velocity of each string at the sound porthole diaphragm is equal to the velocity of the sound porthole diaphragm itself, we have that

$$W_i^- = \dot{x} - W_i^+ \tag{2}$$

for each i. In addition, the control force F_C to be exerted by the virtual model on the sound porthole is equal to the sum of the forces due to the virtual strings[11]:

$$F_C = \sum_{i=1}^{N} R_i (W_i^+ - W_i^-). \qquad (3)$$

The reader should note that the single-sample delay z^{-1} prevents the mechanical analog for the controller from being accurate at high frequencies, although operating at higher sampling rates can mitigate this effect. The most important consequence is that the maximum loop gain becomes limited in practice. In addition, it is necessary to have a one-pole lowpass filter somewhere in the main control loop to roll off the control energy (not shown). This filter may be part of the power amplifier, analog-to-digital converter, digital-to-analog converter, or it can be included explicitly in the digital controller program.

3.2 Alternative Implementation

In contrast, although we have not done so for this particular project, it would be possible to eliminate the single-sample delay z^{-1} by employing techniques from wave haptics[7]. We essentially factor the delay-free loop out of the digital controller into an analog controller. If we substitute (2) into (3), then we obtain the following:

$$F_C = 2\sum_{i=1}^{N} R_i W_i^+ - \dot{x}\sum_{i=1}^{N} R_i. \qquad (4)$$

The first term consists of delayed waves that could be output by the digital feedback controller, whereas the second term, while undelayed, could be computed directly in analog and summed in analog with the output from the digital feedback controller.

4 EXPERIMENT

4.1 Part One

To demonstrate the feasibility of our method for enhancing the reverberant properties of a room, we conducted an experiment using a single sound porthole. The sound porthole was constructed using an 8” (20.32cm) Quam 8C10DVPAXB dual voice coil loudspeaker driver with a mechanical resonance frequency of about 100Hz. For this driver, the impedance crossover frequency mentioned in Section 2.3 was approximately 325Hz. We placed a Shure Beta 57 microphone 2cm away from the sound porthole on axis and an ADAM A7 loudspeaker 70cm away on axis on the table in a regular office as shown in Figure 6. We measured the magnitude of the transfer function from the loudspeaker to the microphone. The magnitude response without control (i.e., without reverberation enhancement) is shown with the thick blue line in Figure 7. The magnitude response with reverberation enhancement is shown with the thin red line in Figure 7.

The digital waveguide network response caused the magnitude response with reverberation enhancement to become much more complex. The bandwidth of the control was effectively from about 70Hz to 500Hz (see Figure 7). The sound porthole was most efficient near and slightly above the driver's natural resonance frequency of about 100Hz, which was clearly related to the frequency region where the control was most effective.

4.2 Part Two

We further hypothesized that increasing the efficiency of the sound porthole would improve the performance. In particular, we wanted to demonstrate that reducing the mechanical mass of the sound porthole would improve the impedance match between the sound porthole and the air. We decided to use feedback control to change the mass of the sound porthole by Δm. In parallel with the controller described in Section 3.1, we applied the controller

$$F_{C,change\,mass} = \Delta m \hat{\ddot{x}}, \tag{5}$$

where we formed the acceleration estimate $\hat{\ddot{x}}$ by differentiating the velocity estimate $\hat{\dot{x}}$. Implementing this mass controller was simple and effective, and it was indeed typical of the advantages that one often encounters when implementing control systems using mechanical analogs. While reducing the mass was not passive, the control system was still stable since we knew that the sound porthole was guaranteed to have a minimum mechanical mass.

We tested our mass-change hypothesis in the time domain. The energy decay curves (EDC) of measured impulse responses from the loudspeaker to the microphone (see Figure 6) helped us illustrate how quickly the reverberation energy at the microphone decayed. We calculated the EDC of an impulse response *q(t)* using the tail integral of the squared impulse response at time *t*[11]:

$$\mathrm{EDC}\Big[q(t)\Big] \triangleq \int_t^{\infty} q^2(\tau)d\tau. \tag{6}$$

The blue solid line in Figure 8 shows how fast the energy decayed in the room when the control system was disabled. We observed that the energy initially decayed rapidly until about 10ms, after which point it decayed at an approximately constant rate. In contrast, when the reverberation enhancement system was enabled and the mass of the driver was decreased slightly, then the energy decayed more slowly as shown with the red dashed-dotted curve in Figure 8. However, when the mass of the driver was instead increased using feedback control, then the energy decayed at about the same rate as with the mass decreased, but the EDC settled in to this rate at a later time and lower level (see the green dashed curve in Figure 8). In other words, the configuration with the decreased mass clearly increased the intensity of the reverberation enhancement. We believe that this was because decreasing the sound porthole mass should have improved the impedance match with the air.

5 CONCLUSIONS AND FUTURE WORK

Room reverberation enhancement systems can be designed using feedback control with mechanical analogs. A feedback controller implements a passive connection between virtual acoustic spaces, which are realized using digital waveguide networks, and *sound portholes*. Because the feedback controller models a passive system, the loop gain can be made relatively large even in practical scenarios. Furthermore, the system does not need to be re-calibrated if the properties of the room change or even if moved to a whole new room. Finally, this method for designing room reverberation enhancement systems may generally result in more realistic reverberant sound because it implements the acoustical features of a system that could exist naturally in the physical world.

These attractive properties warrant future work to increase the control bandwidth of the system and the overall intensity of the effect. The first step is to design sound portholes that have mechanical impedances that are more closely matched to air than standard loudspeakers. We showed that decreasing the mass of a standard loudspeaker can help, but the mass is only one of the components contributing to the sound porthole mechanical impedance. The mass of the transducer should be physically reduced at the manufacturing stage anyway rather than as a design afterthought using feedback control. A related but alternative strategy would be to modify an electrodynamic microphone, replacing the standard voice coil with a dual voice coil that could dissipate enough heat to serve as a loudspeaker winding.

If the sound porthole's mechanical impedance can only be matched to the air's impedance over a limited band, then each sound porthole will only be effective in its own band. As a consequence, it might be best to plan to employ some sound portholes for controlling lower frequencies and other sound portholes for controlling higher frequencies.

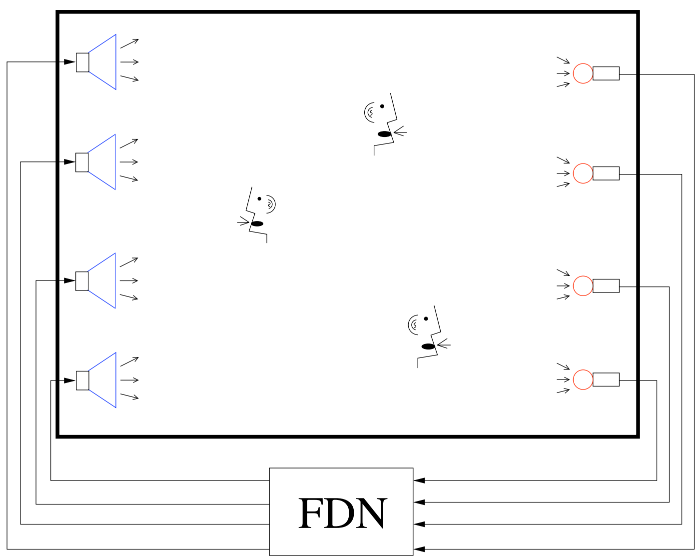

Fig. 1 - Example prior active room reverberation enhancement system.

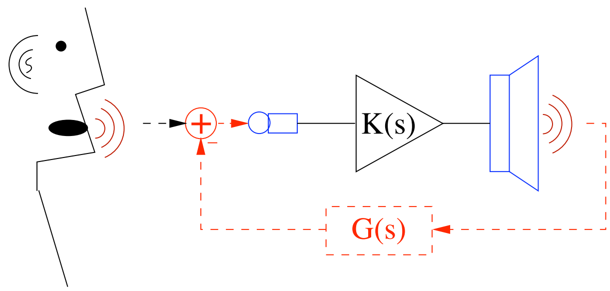

Fig. 2 - Person speaking into a microphone, whose signal is processed by an amplifier with transfer function K(s) and fed to a loudspeaker; acoustic path from loudspeaker to microphone represented by G(s).

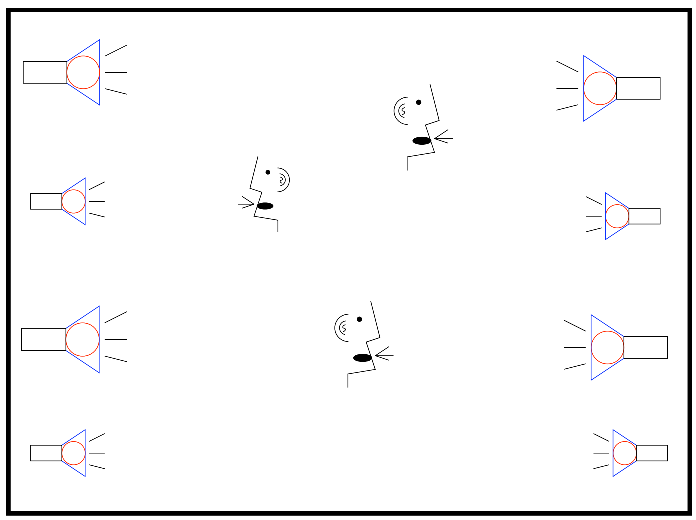

Fig. 3 - Room reverberation enhancement using sound portholes.

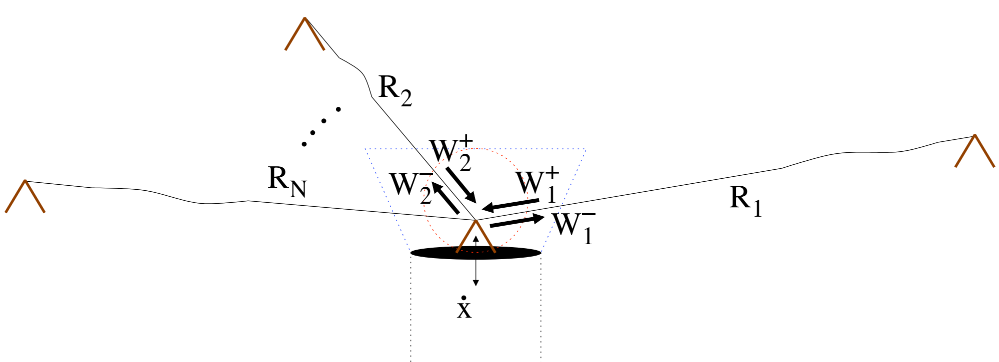

Fig. 4 - Mechanical analog of a sound porthole connected to N waveguides.

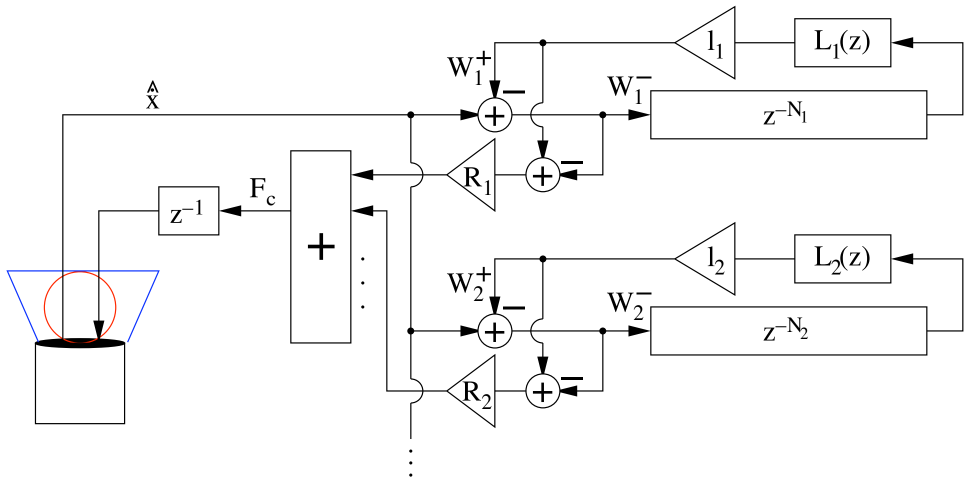

Fig. 5 - Signal flow graph for interfacing a digital waveguide network with a sound porthole.

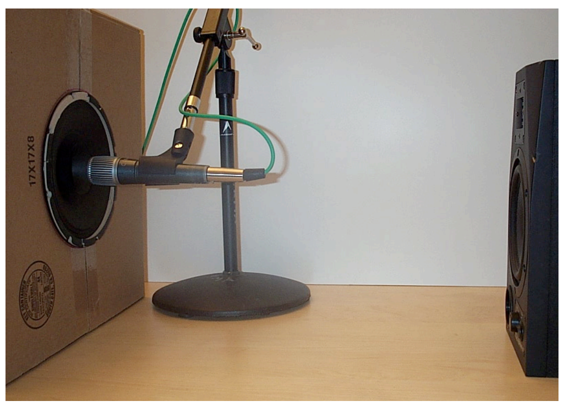

Fig. 6 - Measurement setup.

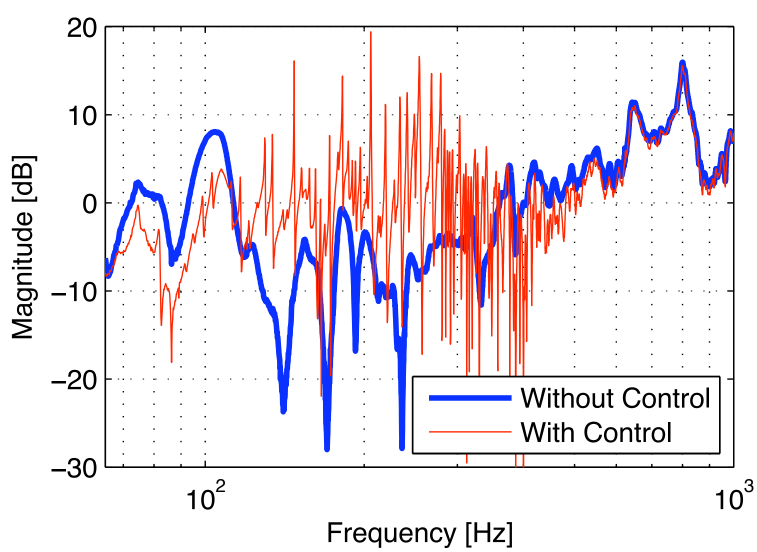

Fig. 7 - Magnitude response without control (thick blue line) and with control (thin red line).

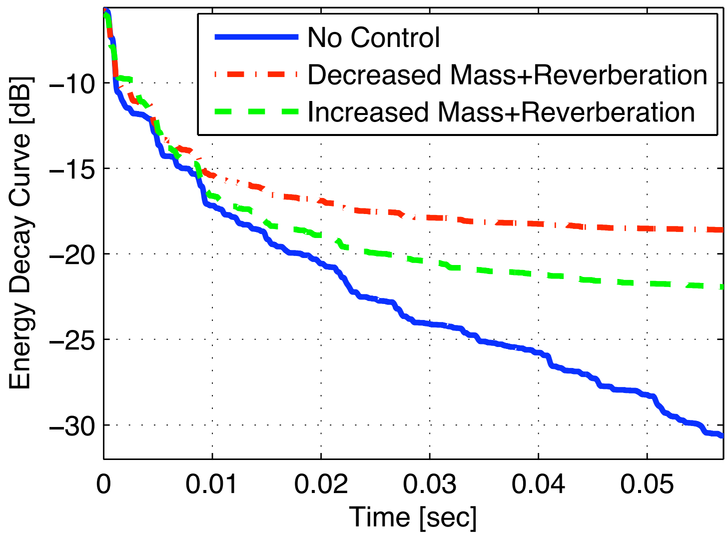

Fig. 8 - Energy decay curve with no control (blue), with reverberation control while the mass was decreased (red dashed-dotted), and with reverberation control while the mass was increased (green dashed).

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