Using Active Sound Intensity Control to Increase Transmission Loss

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ABSTRACT
Actively controlling sound transmission loss (TL) is difficult because it is not possible to measure its instantaneous value. TL measurements require complex experimental procedures involving post processing in the frequency domain. The current approaches for TL improvement via active noise control consist of controlling pressure or particle velocity at the reception side. However, local attenuation in near field or highly reverberant acoustic fields may not improve the TL. In this work an adaptive frequency domain controller that minimizes the sound intensity computed from microphone and particle velocity sensor signals measured at the same location at the reception side is proposed. The proposed method (ASIC-PU) is compared with classical active noise control strategies such as FX-LMS in the time and frequency domain. A hybrid actuator ("smart pillow") consisting of a piezoelectric patch attached to a curved Plexiglas plate embedded in polyethylene foam is used to increase the TL of an aluminum panel. The panel is placed in a plane wave tube where the TL can be measured. Experimental results show that, at lower frequencies, where the assumption of an anechoic acoustic termination is less realistic, ASIC-PU can outperform conventional ANC methods.

Keywords: Sound, Insulation, Transmission. I-INCE Classification of Subjects Number: 37.7

1. INTRODUCTION
Sound Transmission Loss (TL) of panels can be evaluated by measuring the incident sound power at one side of the panel and the sound power transmitted to the other side. The panel is usually placed between a reverberant room, where the incident sound power can be measured, and an anechoic room, where the transmitted noise can be measured (1). In the simpler case of normal incidence, a plane wave tube can be used. The TL is defined for transmission to an anechoic termination. Given that the practical implementation of an anechoic termination is not simple, a two-load method is normally used (2).

Increasing sound TL of panels using active noise control (ANC) is not an easy task, because the TL cannot be directly measured. Recent works have used directly measured physical quantities involved in the TL evaluation in strategies to control indirectly the TL (3,4,5,6). Different configurations have been tested to try to improve the TL using Least-Mean-Square (LMS) controllers in time or frequency domain, \textit{i.e.}, measuring sound pressure, particle velocity, or both simultaneously (7,8).

This work shows a new approach to maximize the TL with an active control strategy which drives a smart foam actuator ("smart pillow") developed by the authors in previous work (3). This controller acts to minimize the acoustic energy transmitted through a panel. This controller is called Active Sound Intensity Control (ASIC) and was developed by one of the authors and collaborators in previous works (9,10,11). This controller estimates the transmitted sound intensity and acts to minimize it. The transmitted sound intensity estimation is computed from measurements of sound pressure and particle velocity. Both can be measured simultaneously with special sensors (12) or by a sound intensity probe consisting of two phase-matched microphones (PP).

The ASIC controller consists of an algorithm similar to the traditional frequency domain LMS
controller (13), but using two error signals simultaneously to evaluate the sound intensity, which is minimized in the adaptive algorithm.

The main goal of this work is to investigate different algorithms for transmission loss control using a hybrid actuator. This analysis is performed by comparing the increase in TL achieved with each controller.

In section 2 an experiment carried on in a plane wave tube to calibrate the particle velocity sensor is described. In this experiment, the Prony Method, especially adapted for three equally spaced measurement points was used to precisely estimate the wavenumber, the pressure and the particle velocity at any point along the tube. Section 3 presents a brief review of the Filtered-X LMS control algorithms in the time and frequency domain and the ASIC-PU algorithm used in this work. Section 4 describes the experiments performed in a plane wave tube. The TL achieved with the different controllers is compared and, finally, in section 5, the results are presented and some conclusions are drawn.

2. VALIDATING THE PARTICLE VELOCITY MEASUREMENT

The particle velocity sensor is not as easy to calibrate as a microphone. The sensor gives a signal that is linearly related to the particle velocity, but the calibration factor may depend on frequency. In this work an experiment was performed in a plane wave tube with three equally spaced microphones to calibrate the particle velocity sensor. Using Prony's approach, it is possible to determine the wavenumber, the sound pressure, and the particle velocity at any position along the tube, hence with no previous measure of the air temperature or atmospheric pressure to estimate the sound propagation velocity.

The tube used in the experiment has a circular cross section with 6 in. diameter and is made of PVC. At one end a loudspeaker was placed and, at the other end, a foam insert with a spherical surface approximately 80mm thick was placed inside the tube in order to reduce the sound reflection. A scheme of the setup is illustrated in Figure 1, where the lengths are indicated.

![Figure 1 – Plane wave tube setup scheme.](image)

Figure 1 – Plane wave tube setup scheme. $A_1$ and $A_2$ are traveling plane sound wave amplitudes and $x_i (i = 0, 1, 2)$ indicate the longitudinal position of the microphones.

The sound pressure at any position inside the tube can be written as:

$$\hat{p}(x_i) = A_i e^{-iKx_i} + A_j e^{+iKx_i}$$

(1)

Using the fact that $\Delta x_1 = \Delta x_2 = \Delta x$ and making $s_1 = -iK$ and $s_2 = iK$, Eq. (1) can be simplified to

$$\hat{p}_i = \sum_{j=1}^{2} A_j e^{s_j x_i}$$

(2)

where $x_0 = 0, x_1 = \Delta x$ and $x_2 = 2\Delta x$. If $e^{s_j \Delta x} = V_j$, the sound pressure will be given by

$$\hat{p}_i = \sum_{j=1}^{2} A_j V_j^i$$

(3)
The Prony method (14) can be applied to this problem to determine both unknown values, i.e., the complex wave amplitude and the wavenumber $K$. The main advantage of this method is the experimental measurement of the wavenumber, which makes it unnecessary to estimate it based on the environmental conditions. Cases where the viscous loss is relevant due to the large distances involved were not tested (when the wavenumber becomes complex), but the formulation is not restricted to real wavenumbers.

Using Prony’s method, one can write:

$$\sum_{i=0}^{2} \beta_i \hat{p}_i = \sum_{j=1}^{2} A_j \sum_{i=0}^{2} \beta_j V_i' = 0$$  \hspace{1cm} (4)

The next step is to determine the roots of the polynomial equation in Eq (4). Making $\beta_2 = 1$ without loss of generality, the follow relation can be found:

$$\beta_0 \hat{p}_0 + \beta_1 \hat{p}_1 = -\hat{p}_2$$  \hspace{1cm} (5)

Now, if the second term of Eq. (4) is equal to zero the values of $V_j$ that make this new equality true can be found. Remembering that $e^{s_j \Delta x} = V_j$, $S_1 = -iK$ and $S_2 = iK$, the roots of $V_j$ are:

$$V_1 = e^{-iK \Delta x}$$  \hspace{1cm} (6)

$$V_2 = e^{iK \Delta x} = \frac{1}{V_1}$$  \hspace{1cm} (7)

It is possible to assemble a quadratic equation that has roots of the type of Eq. (6) and Eq. (7), as

$$(s - V_1)(s - \frac{1}{V_1}) = s^2 - (V_1 + \frac{1}{V_1})s + 1 = 0$$  \hspace{1cm} (8)

Remembering that $\beta_2 = 1$ and developing the sum within the second part of the equality in Eq. (4) (which must be equal to zero), one can obtain:

$$\beta_0 + \beta_1 V + \beta_2 V^2 = V^2 + \beta_1 V + \beta_0 = 0$$  \hspace{1cm} (9)

By comparing Eq. (8) and Eq. (9), the values $\beta_1 = V_1 + \frac{1}{V_1}$ and $\beta_0 = 1$ are obtained. At this point, the only one unknown is $\beta_i$, but with Eq. (5) it is possible to write:

$$\beta_i = \frac{-\hat{p}_2 - \hat{p}_0}{\hat{p}_1}$$  \hspace{1cm} (10)

Thus, summarizing, measuring the sound pressure at the points $\hat{p}_0$, $\hat{p}_1$, and $\hat{p}_2$ it is possible to determine the value of $\beta_i$. After that, it is possible to determine the roots of the quadratic equation given by $V^2 + \beta_1 V + 1 = 0$, which determines the experimental wave number since $V_{1,2} = e^{\pm iK \Delta x}$. Reorganizing the last relation, the experimental wavenumber is given by:

$$K = \left| \frac{\ln V}{i \Delta x} \right|$$  \hspace{1cm} (11)

The complex amplitudes of the wave equation (Eq. (1)) can be extracted by solving the matrix equation given by:

$$\begin{bmatrix}
I & I & e^{-iK \Delta x} & e^{iK \Delta x} \\
0 & 0 & e^{-iK \Delta x} & e^{iK \Delta x} \\
0 & 0 & e^{-2iK \Delta x} & e^{2iK \Delta x}
\end{bmatrix} \begin{bmatrix}
\hat{A}_1 \\
\hat{A}_2
\end{bmatrix} = \begin{bmatrix}
\hat{p}_0 \\
\hat{p}_1 \\
\hat{p}_2
\end{bmatrix}$$  \hspace{1cm} (12)

To validate this formulation, a test was performed inside a 6 in. PVC tube with a loudspeaker at one end and three equidistant holes to hold microphones at the mid span of the tube. The side opposite to
the speaker was closed with melamine foam to reduce the reflected waves at the end of the tube. The tube is illustrated in Figure 2.

![Experimental setup for wavenumber estimation using Prony’s method.](image)

Figure 2 - Experimental setup for wavenumber estimation using Prony’s method.

After the experimental estimation of the wavenumber, a Microflown particle velocity sensor was positioned at a known position relative to the microphone position to measure the particle velocity. So, the measured particle velocity can be checked with the particle velocity determined using the experimental wavenumber and wave amplitudes.

![Comparison between theoretical and experimental values of the wavenumber.](image)

Figure 3 - Comparison between theoretical and experimental values of the wavenumber.

The comparison between the experimental and theoretical wavenumber is shown in Figure 3. The theoretical value was obtained by dividing the angular frequency by the sound velocity propagation in the air (approx. 342 m/s in lab conditions of pressure, temperature and humidity). Near 1 kHz an angular coefficient inversion occurs in the experimental curve. This effect is due to spatial aliasing, when half of the wavelength is equal to the distance between microphones. Manually correcting for aliasing makes it possible to reach a good convergence between the theoretical and experimental curves until the tube cut-off frequency (after this frequency, non-planar waves propagate along the tube, around 1.3 kHz for this tube diameter).

For the comparison between the particle velocity estimated via the Prony method and the particle velocity measured with the Microflown particle velocity probe, this last sensor was positioned at the same cross section of the second microphone (170mm away from the other two microphones). The result is shown in Figure 4.

For the frequency range between 1Hz and 900Hz the difference between the two particle velocities estimates is only in the amplitude, which can correct by a calibration factor near of 3. Note that no calibration gain was applied to the particle velocity probe measurement (calibration factor equal to 1). This result shows that in this frequency range the Microflown particle velocity probe signal is highly linearly proportional to the particle velocity.
3. CONTROL SCHEMES

The main objective of this paper is to investigate the behavior of the ASIC-PU strategy in a TL control problem. To achieve this, a comparison between the performance of the ASIC control and that of other traditional strategies was made. The other strategies were the classical Filtered-X LMS with one and two error signals, both in the time and frequency domain.

The block diagram of the Filtered-X LMS algorithm, shown in Figure 5, is the basis of all the controllers tested in this paper. The only change between the Filtered-X and the ASIC-PU control strategies is in the adaptive block, which has the adaptive algorithm equation changed.

In Figure 5, \(x(n)\) indicates a reference signal, \(e(n)\) the signal measured by the error sensor, \(W(z)\) it is a Wiener filter in which the gains are adjusted by the LMS algorithm (represented by LMS in the figure). \(S(z)\) means the secondary path acoustic system (the acoustic path between the control actuator and the error sensor) and, finally, \(\hat{S}(z)\) is an off-line estimation of the secondary path.

The first variation of Filtered-X LMS tested used two error sensors simultaneously. The control diagram box is similar to the one in Figure 5, but with two different error signals feeding the LMS algorithm (a pressure signal and a particle velocity signal) and, consequently, two off-line estimations of the secondary path used to filter the reference signal.

\[
W(n + 1) = W(n) + \frac{\mu x'(n)e(n)}{|x'(n)|^2}
\]  

(13)

Here the future gains values, expressed by \(W(n+1)\), depend on the actual gain values, \(W(n)\), and on a normalized relation between the reference filtered by the secondary path, \(x'(n)\), the instantaneous
error signal, $e(n)$, and the parameter $\mu$ that controls the convergence step and the stability of the algorithm. Due to the normalization, the parameter $\mu$ must be between 0 and 1. All adaptive filters experimentally tested were of 16th order and the algorithm step size was equal to 0.002.

For two error sensors, the adaptive normalized equation becomes:

$$W(n+1) = W(n) + \mu \left( \frac{x'_1(n)e_1(n)}{|x'_1(n)|^2} + \frac{x'_2(n)e_2(n)}{|x'_2(n)|^2} \right)$$  \hspace{1cm} (14)

In Eq. (14) the indices of $x'$ and $e$ indicate the error sensor. $x'_i$ means the reference signal filtered by the secondary path for the error microphone and $e_i$ is the instantaneous pressure measured by the microphone, while index 2 indicates that the signal is measured with the Microflown particle velocity sensor and the secondary path is for this sensor.

The second version of the Filtered-X LMS controller is the frequency domain (with one and two error sensors also). In order to control in the frequency domain, the filtered reference and error signal must go through a Fast Fourier Transform (FFT) before getting into the filter or adaptation block. Consequently, the control law must go through an Inverse Fast Fourier Transform (iFFT) to transform back to the time domain and be applied to the secondary path, as illustrated in Figure 6.

![Figure 6 - Frequency domain Filtered-X LMS block diagram.](image)

Although both implementations are very similar, the control action of each one is significantly different. While in the time domain the control will act on the whole frequency range selected in the secondary path estimation, in the frequency domain the controller will concentrate the control power on some discrete frequencies selected by the secondary path estimation and by the set of parameters defined by the size of the data block and the sample rate of the DSP system where the control is running, combined in Equation (13), which determines at which frequencies the frequency domain controller will act:

$$f_i = \frac{i}{2NdT}$$  \hspace{1cm} (15)

where $f_i$ indicates the $i$-th frequency ($i$ can vary from 1 to $N/2$), $N$ represents the data block size and $dT$ is the sampling interval.

There are two advantages in the use of the frequency domain FX-LMS controller. The first is that it concentrates all the control effort to attenuate just a few discrete frequencies, instead of distributing it over all frequencies in the control range. The second is that the frequency domain implementation will not recognize the existence of frequencies different from the ones given by Eq. (13). In other words, it can be said that the frequency approach is less sensitive to external disturbances in the error signal, as these disturbances (noise) tend to occur at different frequencies, not affected by the control law. For the frequency control with two error sensors, the adaptive algorithm was changed in the same way as the time domain algorithm.

The last strategy is the ASIC control. This controller is based on the frequency domain LMS and works in a similar way, but instead of minimizing the instantaneous error in the two error sensors, this strategy computes the acoustic intensity and aims at minimizing it. The adaptive equation is given by (9):
The meaning of each variable in Eq. (16) is the same as for the Filtered-X LMS in the time domain when applied with two error sensors. The operator $\Re$ indicates the real part of a complex number and $*$ indicates the complex conjugate.

4. EXPERIMENTAL SETUP

To compare these strategies of indirect TL control, an experiment was performed. A thin aluminum plate was clamped at the mid span of a plane wave tube with square cross section, and a “smart pillow” was placed next to this aluminum plate. The “smart pillow” presents low acoustic TL at low frequencies, but a considerable control action, what makes the assembly performance dependent of the control strategy. The waveguide scheme is presented in Figure 7 and the “smart pillow” is shown in Figure 8.

In order to compute the sound TL, it is assumed that the flare at the end of the plane wave tube of Figure 7 filled with foam is perfectly anechoic and that only incident normal plane waves reach the test sample (2). In this case one needs to evaluate only four transfer functions between the four microphone locations $M1$, $M2$, $M3$ and $M4$ and the reference signal (signal sent to the loudspeaker). Details of the techniques for TL measurement can be found in the literature (15). Pressure microphones GRAS® model 40AQ have been used.

Adopting a complex exponential representation and the origin of the coordinate system at the sample surface, the sound pressure and particle velocity in the frequency domain can be expressed as:

\[
P(x, \omega) = \begin{cases} 
A(\omega) e^{-jKx} + B(\omega) e^{jKx}, & x < l \\
C(\omega) e^{-jKx} + D(\omega) e^{jKx}, & x > l
\end{cases} \tag{17}
\]

\[
V(x, \omega) = \begin{cases} 
\frac{A(\omega) e^{-jKx} + B(\omega) e^{jKx}}{\rho_0 c}, & x < l \\
\frac{C(\omega) e^{-jKx} + D(\omega) e^{jKx}}{\rho_0 c}, & x > l
\end{cases} \tag{18}
\]
where \( \omega \) is the angular frequency, \( K \) is the wave number, given by \( K = (\omega / c - ja) \) (note that \( K \) can be complex to add viscoelastic and thermal dissipation to the pressure and velocity), and \( l \) is the length of the first section of the tube, upstream from the sample. Here, we assume \( \alpha = 0 \). \( A, B, C \) and \( D \) are complex amplitudes of the plane wave components (\( A \) describes the incident wave, \( B \) the wave reflected from the sample, \( C \) the transmitted wave and \( D \) the wave reflected by the waveguide termination).

To compute the TL of a sample, the reflected wave (represented by \( B \) in Eq. (17) and Eq. (18)) is not considered. Besides, assuming that the waveguide termination is perfectly anechoic, the coefficient \( D \) was neglected.

The coefficients \( A \) and \( C \) can be expressed as a function of the measured transfer functions at positions 1 to 4, respectively, as:

\[
A = \frac{j(H p_1 R e^{j\kappa_1} - H p_2 R e^{j\kappa_2})}{2 \sin[k(x_1 - x_2)]}, \quad C = \frac{j(H p_3 R e^{j\kappa_3} - H p_4 R e^{j\kappa_4})}{2 \sin[k(x_3 - x_4)]}
\]  

(19)

where \( H_{PnR} \) represents the complex Frequency Response Function (FRF) measured between the noise source (reference signal) and the pressure at microphone \( n \). The sound transmission loss is defined by:

\[
TL = 10 \log_{10} \left( \frac{W_t}{W_i} \right)
\]  

(20)

where \( W_i \) and \( W_t \) represent the incident and transmitted sound power, respectively. For a perfect anechoic termination (\( D=0 \)), assuming normal incident plane waves, the sound TL is given by:

\[
TL = 10 \log_{10} \left( \frac{H_{P4R}}{H_{P3R}} \right)
\]  

(21)

As error sensors, a general propose microphone (01dB model MCE212) and a particle velocity sensor (Microflown®) were positioned 50mm downstream from the “smart pillow”. Both were fixed at the same support, with the sensing parts aligned, as shown in Figure 9.

![Figure 9 - Microphone and particle velocity probe at the same support with sensitive parts aligned.](image)

4.1 Secondary Path Identification

To implement any of the acoustic control strategies listed above it is necessary to perform the secondary path identification, which consists of measuring the FRFs of the acoustic system from the control signal input to the error sensor output and estimating a linear model that presents a similar response in the desired frequency range, which is the frequency range where the controllers will be able to act. Reconstruction filters and anti-aliasing filters must be used in the identification step to include the delay imposed by these elements in the system response.

Two measures of the system response were performed with an HP35650 spectrum analyzer. To identify the linear model, the Matlab® identification toolbox named PEM (Prediction Error estimate for linear or non linear Models) was used. An 18th order system state space model (the A matrix is 18x18) for the frequency range from 128Hz to 640Hz was estimated with analogical signals sampled at 2048 Hz sampling rate.

The comparison between the measured responses and the identified model responses is shown in Figure 10. Note that it is not necessary to perform a sensor calibration, since all control strategies adopted in this work need only a signal highly correlated with the error signal, not the accurate error signal. So, the sensor measurements are presented in Volts, not in Pascal or m/s, as it should be for sound pressure and particle velocity, respectively. According to results in Figure 10, the controllers should work as expected between approximately 110Hz and 680Hz.
4.2 Controller Implementation

As discussed above, frequency domain controllers only work at frequencies that obey the relation given in Eq. (15). To establish a common criterion for evaluating the control strategies, only frequencies in the identified range that could be modified by the action of the frequency domain controllers were tested. So, a periodic disturbance (reference) composed by the sum of sinusoids from 64Hz to 768Hz was used.

The diagram presented in Figure 11 shows the electrical connections of the experimental setup. The signal was generated by a dSPACE® 1003 system and, after passing through a reconstruction filter (an 8th order Butterworth low pass filter from Frequency Devices®, model 900, with cutoff frequency adjusted to 1 kHz) to eliminate the effects of digital/analogical conversion, it was fed to an amplifier connected to the plane wave tube primary source at the same time that it got into the dSPACE® 1104 to be the reference for the control strategies.

As for the reference signal, the control signal also needs to pass through a reconstruction filter with the same configuration before it is fed to the “smart pillow”. The error sensors were connected to ICP conditioners and anti-aliasing filters (also with the same characteristics of the reconstruction filters) and then to the dSPACE® 1104, where the control strategy was loaded and run.

For each controller, after 5 seconds for the adaptive control strategy to reach the optimum values of the FIR filter gains, a spectrum analyzer measured the sound pressure at the points M1 to M4 to evaluate the FRF between the pressure at these points and the electrical signal that was feeding the primary source amplifier. This step was repeated four times to use only one 40AD GRAS® microphone placed at different positions. So, the same input channel of the spectrum analyzer was used with the same anti-aliasing filter and ICP conditioner, which guaranteed that there was neither phase error nor calibration error, since the TL is given by a ratio.

Using these FRFs, Eq. (19) and Eq. (21) were used to compute the TL. While the FRFs were
measured, the adaptive characteristic of each controller was disabled in order not to change the acoustic system while the spectrum analyzer was measuring. In all the tests the positions of the error sensors remained the same. Therefore, the identification step was not repeated. The only thing that changed among the tests was the control strategy. Sample frequency, adaption time, convergence step and filter order (number of gains of the FIR filter) were kept equal for all implementations.

5. EXPERIMENTAL RESULTS

The TL results for all control strategies tested, with all possible error sensor combinations, are presented in Figure 12, where the bold blue line indicates the TL of the clamped aluminum plate with the smart pillow attached but without control action (only the passive TL).

![Figure 12 - Compilation of the TL obtained with different active controllers.](image)

The poorest TL (the closest result to the passive TL) was obtained using the time domain Filtered-X LMS with only a particle velocity probe as error sensor. As expected for a time domain strategy, this configuration improved the TL only at some frequencies (usually the lower ones, which show a higher contribution for the mean square error). Using the microphone as an error sensor and still a time domain Filtered-X LMS, the TL presented better results (the overall TL grows, but the strategy did not work for all the frequencies).

The frequency domain Filtered-X LMS and the ASIC-PU strategies presented the best results. This can be justified by the conversion to the frequency domain. While in the time domain the controller objective is to minimize the mean square error, in the frequency domain each gain of the FIR filter is adjusted to minimize the error at a specific frequency. So, if the FIR filter has 16 adjustable gains, they can be set to independently minimize the error measured at 16 different frequencies. The frequencies that the frequency domain algorithms can control are defined by the relation expressed in Eq. (15).

With the results above 640Hz being disregarded (due to the identification process that stopped at 640Hz), it is possible to say that the behavior of all frequency domain strategies was similar, except at 320Hz, where the ASIC-PU presented a poor result. These controllers act at all frequencies, improving the TL. Although the ASIC-PU presented a good performance at 128Hz, the frequency domain Filtered-X LMS using the particle velocity probe or a Microphone and a particle velocity probe as error sensor presented a better behavior overall in the frequency range analyzed.

Considering a global level of TL (after a logarithmic sum of the TL obtained at each frequency) it is possible to say that the ASIC-PU was more effective than the others. Table 1 presents the sum of the TL values in dB, in a way to establish which controller presented the best performance in the controlled frequency range.

6. CONCLUSION

In this work an Active Sound Intensity Controller (ASIC-PU) was compared with traditional acoustic control strategies based on the Filtered-X LMS algorithm in the time and frequency domain, using a microphone or a particle velocity sensor or both to indirectly increase the sound transmission...
loss. Since ASIC-PU minimizes the sound intensity, it was expected that this strategy would act better than others to maximize the sound transmission loss, which is a logarithmic ratio of the incident and transmitted sound power. The experimental results showed that the frequency domain Filtered-X LMS presented the best results in this application when analyzed frequency by frequency, using a particle velocity sensor and a microphone, or both. The control of the instantaneous sound intensity presented the most effective control of the TL when a global level of TL was considered.

Table 1 - Results (in dB) of sound transmission loss and the global sound transmission loss of the experiment.

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