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SangGon Cha, Andrey G. Troshin

SQ engineering, Dream Building 160-11,Garak-dong, Songpa-gu, Seoul, South Korea
atroshin@yahoo.com

Active noise control barrier for national road in South Korea: Part I Preliminary noise study and electro-acoustic hardware design

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Traffic noise annoyance in dwelling area like private houses and apartment is becoming very important issue nowadays in South Korea and need to be solved. In this paper frequency analysis of the traffic noise near highways was performed. Based on sound pressure levels analyses and theirs frequency content specific requirements for component of active noise control systems are formulated. Active compensational loudspeaker, microphone front end and DSP design were analysed, considered and proposed. Required components specifications and design solutions for electro-acoustic hardware are verified using experimental technique.

Keywords: noise barrier, noise levels, ANC (active noise control), active loudspeaker, DSP platform, adaptive algorithms.

1. INTRODUCTION

Traffic noise annoyance in dwelling area like private houses and apartment is becoming very important issue nowadays in South Korea. Due to local specifics namely the limited dwelling space area since Korea is small country with high population density the traffic noise abatement using proper planning and placing houses faraway from highways and local roads is difficult to provide.

However the intensity of traffic increases and at the same time noise rating for apartments requires providing the total noise level of 45 dBA during day time and 35 dBA at night. The traffic noise abatement can be performed using various techniques. One way is a passive sound proofing which requires the very heavy structures to block low frequency noise. The noise barrier technology among others is also used very often in South Korea.

Traffic noise comprises several components as car engine noise, tire-road interaction noise and aerodynamic flow noise. Car engine noise could contribute to the low frequency noise spectrum due to engine vibrations and exhaust gas pulsation whilst tyre-road interaction noise and aerodynamic flow noise form a stochastic broadband spectrum.

To reduce the low frequency noise the active noise control technology might be appeal to be implemented. Let review the existing technology for active barrier. Theoretical basis for active noise control was proposed by Nelson and Elliot [1] while numerous practical implementations were developed by C. Hansen and S. Snyder [2]. Detail algorithms implementation for ANC was described by S. Kuo and D. Morgan in [3]. Combined passive and active noise control technology with efficiency of 10…20 dB in the range 50…400 Hz can be implemented as active silencer for ducted fan

[4]. Another idea how to use active noise control to provide the local silent zone in apartments was proposed and reported by Sen Kuo. [5].

Also active noise control technology was studied and implemented in order to increase transmission loss of the apartment windows in Germany [6], [7]. Feed forward and feedback active noise control algorithms based systems were investigated.

Active noise control technology can be used for generation so-called soft edge along the noise barrier to increase low frequency transmission loss. The ANC implementation for active noise barrier was reported in [8]–[12].

It should be noted here that ANC technology possesses advantages and drawbacks like any another technology. Using ANC it is possible to avoid implementation of the very huge and heavy structures for blocking low frequency unwanted noise. Efficiency of ANC could be very high both for broadband and tonal low frequency components. Some authors were reported 20…40 dB reduction index.

The technology limitations are connected with frequency range. In other words it related to wavelength size and complexity of the acoustic field. The typical cross-over frequency which separates frequency range for active and passive noise reduction technology is around 500…1000 Hz.

Indeed to make the proper reconstruction of acoustic field the separation between compensational loudspeakers should be equal to half of wave length. However if frequency increases a wavelength becomes shorter therefore the number of controlling channels also increases. It raises the total cost of the ANC solution and complexity of the system controlling and handling.

The following research and development tasks will be discussed and addressed in this part of paper: study of noise frequency content, general requirement to ANC hardware platform, active compensational loudspeaker design, and solution for error and reference microphone front-end.

2. ROAD NOISE ANALYSIS

It is very important to make frequency content analyses in order to determine a presence of prominent low frequency components in the noise spectrum otherwise subjective efficiency of the ANC would be very small due to Fletcher-Monson law. A frequency content analyses using narrow band spectrum technique was performed. Results are depicted in Fig. 1. Doing analyses of the traffic noise levels we can conclude that in the low frequency range there are prominent low frequency components. The critical bands can be determined from Fig. 1. as 25…100 Hz, 125 Hz…175 Hz~ 225 Hz…300 Hz 250 Hz…325 Hz and 450 Hz. Using SPL data for the traffic noise it can be concluded that cancellation loudspeakers should have acoustic sensitivity at least of 90…100 dB per 1 Watt of applied electrical power to radiate the same sound pressure levels as a primary noise source providing the destructive interferences of two waves – source wave and wave generated by control loudspeaker. Also to provide reliable long term operation the active speaker should have some margin of electrical and acoustic power [16].

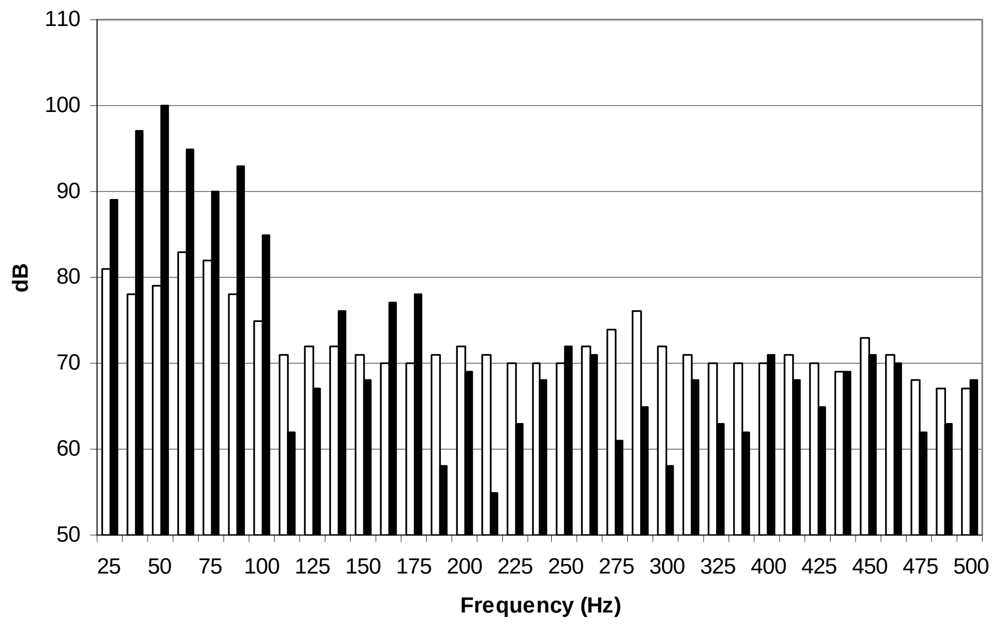

Fig. 1 1/24 octave traffic noise spectrum:

Black color – high way 1 meter from road surface asphalt, 8 lanes heavy trucks and buses Cheonan town, white color – high way 1 meter from the road surface asphalt 6 lanes cars Geumcheong town

3. FORMULATION OF REQUIREMENTS TO DESIGN OF ANC SYSTEM ARCHITECHTURE

Implementing the active noise control it should be noted that unlike very simple 1-D cases of ducted fan and active headset we are generally dealing with 3-D sound field. There are several designing problem that should be solved during industrial implementation of the Active Noise Control technology. They can be briefly summarized as follows:

- where compensational loudspeakers should be placed?
- where error microphones should be placed?
- where reference microphones should be placed if feed forward control supposed to be implemented?
- what type of control has to be implemented: feedback, feed forward or hybrid?
- what type of the algorithm is most efficient *in situ*?
- what type of DSP platform expedient to be used?

The tasks mentioned above will be solved during experimental evaluation of ANC performances in Part II further on.

4. DEVELOPMENT OF SYSTEM COMPONENTS

4.1 General requirement to DSP for ANC system

Consider the general requirement to parameter of ANC DSP system as sampling rate, number of filter taps and delays in electro-acoustic hardware.

It is well known from the sampling theorem that sampling rate should be at least two times higher than highest frequency of the interest. However P. Vitarius *et al.* reported research results that sampling frequency should be 4 times higher for better accuracy in acoustic measurements [13]. For better modeling an impulse response of primary propagating path $h(\tau)$ of the acoustic signal from reference to error microphones the following equality has to be fulfilled as:

$$\delta T_{pp} = N \cdot f_s , \quad (1)$$

where δT_{pp} is the length of the primary path, N – number of filter taps, f_s – sampling frequency.

For longer primary propagating path the higher sampling frequency have to be selected to reduce the number of adaptive filter taps and computational cost as well. The time required for processing an adaptive filter quotients can be estimated as:

$$\delta T_{dsp} = \frac{N-1}{2 f_s} . \quad (2)$$

It also should be noted here about causality problem which means that total delay in electronic and electro-acoustic path should be less than propagating time in primary path. It can be written as:

$$\delta T_{adc} + \delta T_{dsp} + \delta T_{dac} + \delta T_{speaker} + \delta T_{power\_amp} < \delta T_{pp} , \quad (3)$$

where δT_{adc} is the time delay due to ADC, δT_{dsp} – computational time due to DSP, δT_{dac} – time delay due to DAC, $\delta T_{speaker}$ – time delay due to control speaker, δT_{power_amp} – time delay due to power amplifier.

Problem with delay due to acoustic hardware can be alleviated using so-called Filtered LMS algorithm (FxLMS) suggested by Sen-Kuo and D. Morgan. During off-line modeling and using the test signal the delay can be determined and compensated by algorithm. So that Eq.(3) can be re-written as:

$$\delta T_{adc} + \delta T_{dsp} + \delta T_{dac} < \delta T_{pp} . \quad (4)$$

To minimize the final terms $\delta T_{adc} + \delta T_{dac}$ in inequality (4) the low group delay codecs have to be chosen. Also so-called causality problem in acoustic terms must be taken into account. It can be written as:

$$\frac{1}{f_s} < \frac{L}{c} , \quad (5)$$

where: f_s is the sampling frequency L – propagating distance (distance between error and reference mikes), c – speed of sound.

The final inequality means that sampling time $t_s = \frac{1}{f_s}$ should be less than sound propagating time from reference microphone point to error one.

4.2 Microphones and front-end

Performances of the ANC system are also strongly dependent on quality of the reference signal. Regarding positioning of the reference microphone it should be placed as much as closer to the noise source. However for the moving source in the case of the noise barrier this condition can not be fulfilled. Therefore it might be useful to place the reference microphone outside the barrier from the traffic side.

Analog front-end should provide the signal conditioning like impedance matching between microphone output, converting co-axial non-symmetrical microphone output to balanced line and providing an appropriate gain to transmit high level analog signal from reference microphones to DSP board since reference microphone suppose to be placed before barrier and far from the main electronic block. An example of microphone front-end schematic and its dual-channel practical design is shown in Fig. 3. Front-end provides 20 dB gain in a frequency range 20 Hz…1 kHz should be powered by dual power supply ranged from ±5 V до ±15 V and provides balanced audio output.

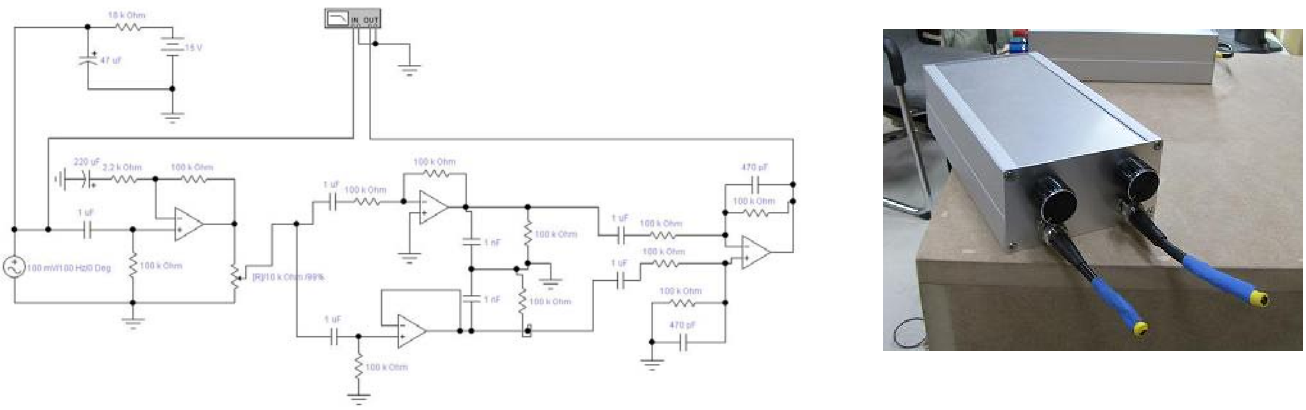

Fig. 2. Schematic of low-cost (5–10$) microphone preamplifier and its embodiment

Another consideration of the microphone design is the cost. Since system suppose to be multi channel due to long barrier it is expedient to choose multimedia or hearing aid miniature electret microphones rather than DJ or studio recording ones due to their relatively high cost.

Electret microphones also appeal to be used due to theirs flat response in the low frequency range starting from 20 Hz to 1 kHz. Those frequencies are of the interest in the case of active noise control. Also environmental factors like temperature, rain and humidity for microphones must be considered and taken into account during custom ANC system design due to local climate specifics since temperature range is wide and summer rainy season also very prominent.

4.2 Active canceling loudspeaker

The generated by DSP algorithm compensation signal should be boosted and radiated by loudspeaker. Despite plenty of different active loudspeaker design and models which are available nowadays on the market theirs direct implementation in ANC system is doubtful. For outdoor ANC system some special requirements have to be fulfilled. For example the loudspeaker should be water and snow proofed taking into consideration climate in Korea. Another requirement that speaker should generate long term sound pressure levels at least 90…100 dB referred to 20 μPa. Also the frequency range should be started from 40…50 Hz.

Most of the low cost multimedia subwoofer can not provide those specifications while Hi-End one is much more expensive. That is why special attention should be also focused to speaker design.

Using Thiele and Small (T/S) parameters it is possible to calculate appropriate speaker cabinet volume. Technique how to measure speaker T/S driver parameters by means of dual channel FFT analyzer with help of speaker electrical impedance curve can be found in [13]. Having at hand measured equivalent volume V_{as}, resonance frequency f_s and total quality factor Q_{ts} of the speaker driver the low cut-off frequency, speaker compartment volume and low frequency FRF bump can be determined. Results of the experimental testing for speaker driver samples are collected to Table 1.

Table 1. Speaker driver test results

Model number for speaker driver	N 128	N 1240
Diameter, cm	27	27
Total Harmonic Distortion (THD)	0.3% (–50 dB)	1% (–40 dB)
Speaker driver sensitivity, dB/Watt m	96	98
Driver resonance frequency, f_s, Hz	33	45
Calculated box volume, V_b, L	33	37
Equivalent volume V_{as}, L	23	13

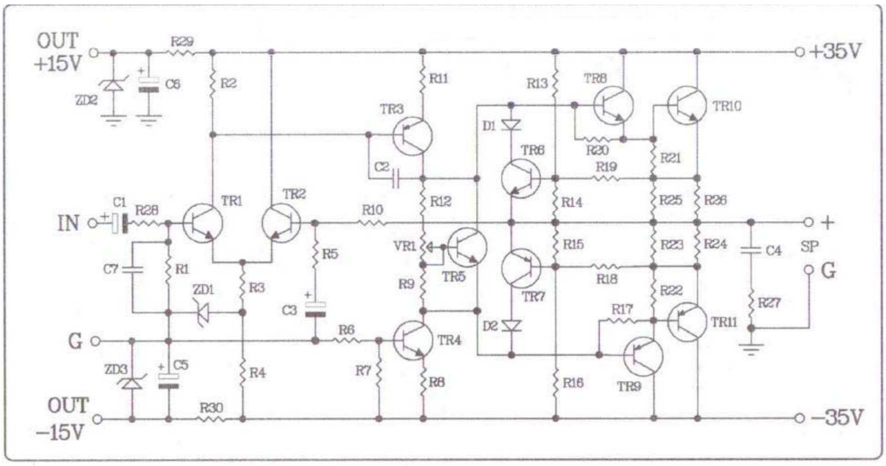

Fig. 3. Schematic of low-cost (~15$) power amplifier

Fig 4. Active compensation loudspeaker design example with power amplifier

The low cost power amplifier speaker schematic shown in Fig. 3 and practical design of active speaker using the amplifier is shown in Fig. 4. Additional low power dual power supply ±15 V in power amplifier scheme can be used to provide DC voltage for microphone front-end. The amplifier provides flat frequency response in the range 20Hz…1 kHz, gain factor 27 dB and output power 50 Watt at 8 Ohm with THD = 0.07 %. Maximum input voltage is 1 V RMS.

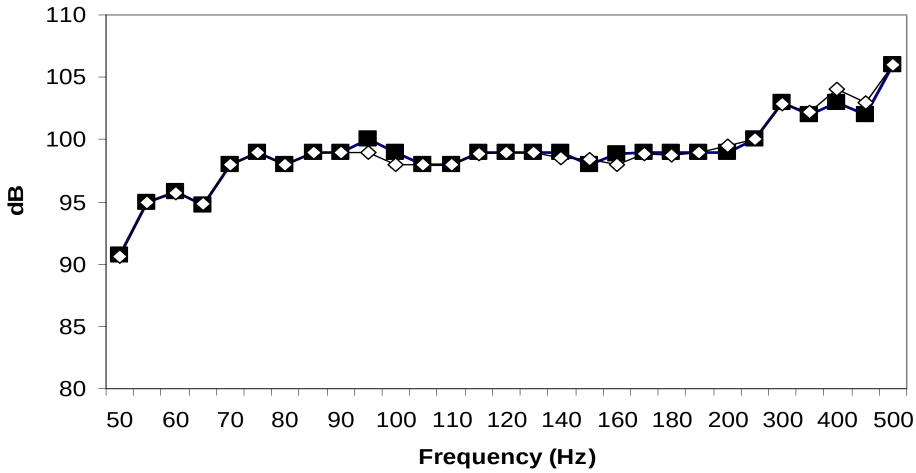

Fig 5. FRF of designed active speaker, power amplifier input voltage 1 V RMS. Effect of the plastic rain protection cover: white mark – plastic cover on, black mark – plastic cover off

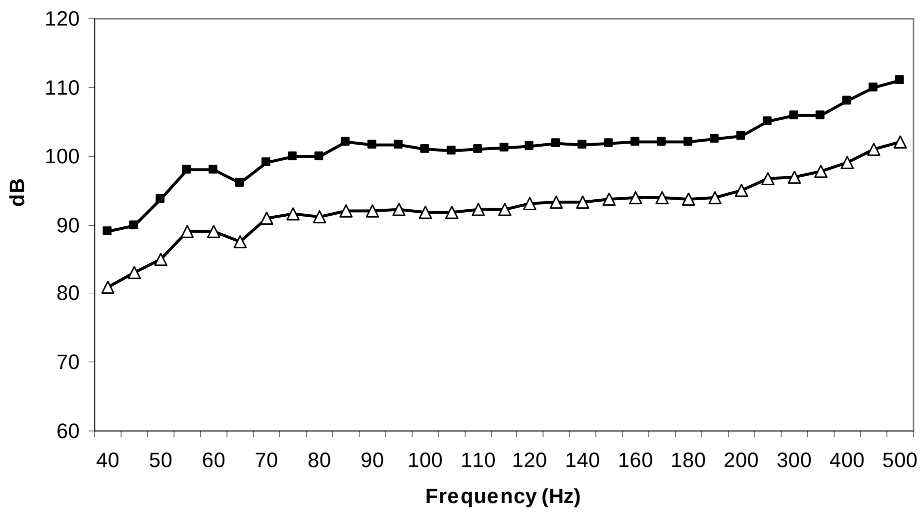

Fig 6. FRF of designed active speaker, rectangular black marker – power amplifier input voltage 1V RMS, triangle white marker – power amplifier input voltage 0.3V RMS

It can be seen from Fig. 5 and Fig. 6. that active speaker FRF with different levels of input voltage applied to embedded power amplifier poses high sensitivity which is enough for anti noise generation and influence of the rain and snow protection cover on its FRF is negligible. Based on test data we can conclude that proposed design of active loudspeaker and microphone front-end are applicable for ANC barrier system.

CONCLUSIONS

As a result of conducted research and experiments the traffic noise analysis was performed. The sound pressure levels and critical frequency bands required for design the canceling loudspeakers of active noise control system of traffic noise were determined. The general requirements to multi channel DSP system for ANC as well as technical and cost considerations to electro acoustic components of the active noise control system for traffic noise were formulated. Dual channel low-cost microphone front-end with balanced output and high effective compensational active loudspeaker with sensitivity of 90…100 dB in frequency range 40…500 Hz were designed, manufactured and tested.

It was shown experimentally that influence of rain protection cover on speaker frequency response is negligible. Designed on preliminary stage of the project and R&D the electro acoustic components of the system are met to formulated requirements and can be implemented in active noise barriers which will be used for control of traffic noise propagated to apartments and dwelling area.

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