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A Comparison of Signal Enhancement Methods for Extracting Tonal Acoustic Signals

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Introduction

The measurement of pure tone acoustic pressure signals in the presence of masking noise, often generated by mean flow, is a continual problem in the field of passive liner duct acoustics research. In support of the Advanced Subsonic Technology Noise Reduction Program, methods were investigated for conducting measurements of advanced duct liner concepts in harsh, aeroacoustic environments.

When performing acoustic liner tests in a flow duct facility, the researcher is faced with the task of optimizing two criteria. The first, and most obvious, criteria is to design the acoustic liner such that the maximum amount of sound absorption is achieved. The other criteria is to obtain a signal-to-noise ratio high enough for quality measurements. Obviously, if the measurements cannot be made with certainty, the development of improved acoustic liners will be inhibited. For grazing incidence impedance tests, the above two criteria are contradictory. As the liner absorptive capacity is increased, the signal-to-noise ratio at the downstream end of the duct (opposite side of liner from sound source) is decreased. For this reason, measurement methods are needed that are capable of extracting the portion of the measured acoustic pressure which is due to the sound source. This is especially difficult when the desired signal is buried beneath the broadband background noise generated by the presence of mean flow.

This report presents the results of a comparison study of three signal extraction methods (SEM) for acquiring quality acoustic pressure measurements in the presence of broadband noise (to simulate effects of mean flow). The performance of each method was compared to a baseline measurement of a pure tone acoustic pressure 3 dB above a uniform, broadband noise background.

Discussion

Baseline method

The selected signal extraction methods were compared with a "hard wired" signal extracted with an existing FFT analyzer, set to a 12.5 Hz bandwidth centered on a tonal signal 3 dB above a uniform, broadband noise spectrum. Initially, it was desired that this test be conducted in the presence of mean flow (in a flow impedance tube). However, changing the mean flow conditions (increasing the velocity) is likely to change the loading conditions on the acoustic drivers. Thus, there is no solid baseline against which to compare the results of the methods studied in the current research. For this reason, it was decided that the test would be conducted using additional acoustic drivers to simulate the acoustic field due to a mean flow.

Figure 1 provides a schematic of the instrumentation that was used to conduct the baseline test. As shown in figure 1, a pure tone (1 kHz) was fed through a power amplifier to an acoustic driver connected to the end of the flow impedance tube. A random noise signal was fed through a second power amplifier to another acoustic driver connected to the flow impedance tube. The respective magnitudes were set to achieve a 103 dB magnitude at the frequency of interest (1 kHz), with a broadband noise such that the signal-to-noise ratio was approximately 3 dB within the 12.5 Hz bandwidth centered on the tone.

Figure 2 provides a demonstration of the variability of measurements using this method. Five sets of data were obtained at each selected data acquisition duration (labeled as averaging time on chart) to determine the variability between measurements. The six choices for averaging time were selected to correspond with the data that will be presented for the three SEM's in this study.

As can be seen in figure 2, the magnitudes of the five sets of measurement signals converge to within ± 0.5 dB after 120 seconds of averaging time. However, the phase components have a range of $10°$ after averaging. Obviously, the results for less averaging time are even less acceptable. As will be shown in the following sections, the new SEM's perform significantly better than the baseline method.

A coherence-based method

The first SEM to be studied was the coherence-based method. This method was found to be quite successful in the extraction of tonal signals which were at least 9 dB below the background noise spectrum (S/N = −9dB). This is a significantly more stringent requirement than shown in the baseline test. However, this method is limited because it only allows for the extraction of the magnitude component of the acoustic pressure signal (the phase component is ignored). Regardless, it is important to note that this technique may indeed be the most efficient method when only the magnitude component is needed.

The underlying equation for this method, taken from reference 1, is

$$SPL_t = SPL_m + 10 \log \left[\gamma^2_{m,s} \right] \tag{1}$$

where SPL_t and SPL_m represent the "true" and measured sound pressure levels, and $\gamma^2_{m,s}$ represents the coherence between the measured signal and the pure tone source. A schematic of the instrumentation used to conduct the study of this SEM is provided in figure 3.

As indicated in figure 3, a random noise generator was used in these tests to simulate the effects of mean flow on acoustic pressure measurements. The random noise was filtered (low-pass cut-off set at 10 kHz) and amplified to a selected level. This signal was then passed through a scanner, which allowed it to be engaged or disengaged via computer control. The resultant signal was then fed to two power amplifiers and their respective acoustic drivers, which were mounted on the end of the flow impedance tube. Simultaneously, a pure tone output from an arbitrary waveform generator was passed through a potentiometer and a low-pass filter/amplifier to two different power amplifiers and their respective acoustic drivers (also mounted on end of flow impedance tube). The pure tone signal was also fed to an FFT analyzer, as was the signal measured by the measurement microphone.

A computer was used to control the hardware in the following sequence:

(1) Disengage random noise generator

(2) Set arbitrary waveform generator to desired frequency (0.5, 1.0, 1.5, 2.0, 2.5 or 3.0 kHz)

(3) Set amplification to achieve pure tone signal of 100 dB at selected frequency

(4) Measure magnitude of measurement microphone signal

(5) Engage random noise generator

(6) Set random noise generator amplification to achieve selected value (9, 3, -3, or -9 dB) of local (within 12.5 Hz bandwidth, centered on test frequency) signal-to-noise (S/N) ratio

(7) Measure source and measurement microphone power spectral densities and the coherence between them using a selected number of averages (25, 50, 100, 200, 400 or 800)

Although the baseline results were for S/N = 3 dB, data for the other S/N's were acquired to provide a better overall understanding of the capabilities of this method. The sequence for the number of averages was used to determine the rate of convergence to a "true" answer, which was assumed to be that determined from step 4 above. A comparison of the measured data is provided in figure 4, in which the error (extracted measurement microphone magnitude minus "true" magnitude) versus the number of averages is given for each of the test frequencies.

Consider first the results for a S/N of 3 dB. As shown in figure 4, the extracted data for this condition collapse to within ± 0.3 dB of the "true" magnitude after 800 averages. After only 200 averages, the results are within ± 0.4 dB. It should also be noted from figure 4 that when the S/N was -3 dB, the results after 400 averages were within ± 0.5 dB. These results are clearly an improvement over that achieved in the baseline tests. It must be noted again, however, that only the magnitude component is available via this method. It should also be noted that the FFT analyzer was operated in a new high-speed mode for each of these new SEM's. Because of this improvement, 800 averages can now be acquired in 2 minutes. The prior mode allowed for only 120 averages to be acquired in this amount of time.

A cross-spectrum-based method

The second SEM to be studied was based on a cross-spectrum method. Based upon the results of this study, this SEM was selected as the "best" method for extracting pure tones from within a broadband noise background. The underlying equations for this method, expanded from reference 2, are provided for completeness.

The following definitions will be used in the ensuing equations:

G_{ab}	cross-spectrum between a and b signals
$\overline{G}_{ab}$	averaged cross-spectrum between a and b signals
H_{ab}	transfer function of signal a to signal b
$n(t)$	time history of broadband contaminating noise
S_a	auto-spectrum of a signal
S_a^*	complex conjugate of auto-spectrum of a signal
SPL_a	sound pressure level of signal a, dB (re 20 μPa)
$u(t)$	time history of "true" acoustic signal (pure tone)
$x(t)$	time history of electronic source signal fed to acoustic driver
$y(t)$	time history of contaminated signal (pure tone plus broadband background noise)

The following equations can be used to extract the "true" acoustic signal $u(t)$ from the contaminated signal $y(t)$. By definition

$$G_{yx} = (S_u + S_n)S_x^* = G_{ux} + G_{nx} \tag{2}$$

$$\overline{G}_{yx} = \overline{G}_{ux} + \overline{G}_{nx} \tag{3}$$

Since S_n is not coherent with S_x^*, $\overline{G}_{nx}$ approaches zero after a sufficient number of averages. Thus, equation 3 can be rewritten as

$$\overline{G}_{yx} = \overline{G}_{ux} \tag{4}$$

It should be noted from this equation that the desired phase component of $\overline{G}_{yx}$ can be acquired simply by taking the phase component of $\overline{G}_{ux}$.

The transfer function of the "true" acoustic signal to the source, H_{ux}, can be derived as either

$$H_{ux} = \frac{S_u}{S_x} = \frac{S_u S_x^*}{S_x S_x^*} = \frac{G_{ux}}{G_{xx}} \tag{5}$$

or

$$H_{ux} = \frac{S_u}{S_x} = \frac{S_u S_u^*}{S_x S_u^*} = \frac{G_{uu}}{G_{xu}} \tag{6}$$

After a number of averages, we can combine equations 5 and 6 to get

$$\frac{\overline{G}_{ux}}{\overline{G}_{xx}} = \frac{\overline{G}_{uu}}{\overline{G}_{xu}} \tag{7}$$

Rewritten, this becomes

$$\overline{G}_{ux}\overline{G}_{xu} = \overline{G}^2_{ux} = \overline{G}_{xx}\overline{G}_{uu} \tag{8}$$

Combining equations 4 and 8 gives

$$\overline{G}^2_{yx} = \overline{G}_{xx}\overline{G}_{uu} \tag{9}$$

By inspection,

$$\overline{G}^2_{xy} = \overline{G}^2_{yx} \tag{10}$$

Thus,

$$\left|\overline{G}_{xy}\right| = (\overline{G}_{xx}\overline{G}_{uu})^{0.5} \tag{11}$$

If we convert our results to a logarithmic form, which more directly matches our measured data, we get

$$SPL_{uu} = 10\log\left|\overline{G}_{uu}\right| = 20\log\left|\overline{G}_{xy}\right| - 10\log\overline{G}_{xx} \tag{12}$$

The schematic of the instrumentation used to conduct the study of this SEM is the same as used for the study of the coherence-based method (figure 3).

Acquisition software was used to control the hardware in the same sequence as was given for the coherence-based method, with the following exceptions:

(1) At step 4, also record the phase between the pure tone source and the measurement microphone

(2) Replace step 7 with the following: Measure cross-spectral density between pure tone source and measurement microphone (magnitude and phase) and power spectral density of pure tone source signal

Analysis software was used to apply the above equations to the measured data to determine the magnitude and phase of the extracted signal.

A comparison of the measured data is provided in figure 5, in which the error (magnitude and phase components of extracted measurement microphone signal minus the "true" signal) versus the number of averages is given for each of the test frequencies. As can be seen from this figure, the data for a S/N of 3 dB are better than that measured for the baseline case when at least 400 averages are acquired. While the magnitude accuracy is observed to be only slightly better than the baseline, the phase accuracy is significantly improved. The phase data have a range of less than 4° centered around the target ("true" answer determined from modified step 4 above), as compared to a range of 10° for the baseline. In fact, after 800 averages the data for S/N's of -3 and -9 dB are generally more accurate than was the case for a S/N of 3 dB in the baseline study.

It should be noted that the ranges for each of the data charts have been set identical to allow for more simple comparisons. As a result, some of the outlying data has been clipped and is not shown. However, none of the outlying data is needed in the discussions provided in this report.

It is expected that this SEM can be further improved if the measurement signal is filtered with a narrow-band tracking filter prior to the computation of the cross-spectra. Due to time constraints, however, this supposition will have to be substantiated at a later time.

A time history signal enhancement method

The third signal extraction method studied was based on a signal enhancement method described in reference 3. The underlying equations are included below.

Let $x(t)$ and $y(t)$ represent the time histories of the portions of the measurement microphone signal which are due to the pure tone and random noise sources, respectively. The total time history $z(t)$ is equal to the combination of $x(t)$ and $y(t)$; i.e.

$$z(t) = x(t) + y(t) \tag{13}$$

If these time histories are subdivided into N synchronous blocks of 1024 samples ($x_k(t)$ and $y_k(t)$), as was done in the current study, averaged time histories can be computed as

$$\hat{z}(t) = \frac{1}{N}\sum_{k=1}^{N}\left(x_k(t) + y_k(t)\right)\Big|_{t=0}^{1023\ \Delta t} \tag{14}$$

where ˆ indicates an averaged quantity. By synchronous blocks, we mean that each block of data ($x_k(t)$ and $y_k(t)$) begins at a time where the pure tone source is at a positive-going zero-crossing.

If $x(t)$ and $y(t)$ are independent processes, as is the case in this study, equation 14 can be rewritten as

$$\hat{z}(t) = \frac{1}{N}\sum_{k=1}^{N} x_k(t) + \frac{1}{N}\sum_{k=1}^{N} y_k(t)\Big|_{t=0}^{1023\ \Delta t} \tag{15}$$

Since $y(t)$ represents a random noise signal, the second portion of equation 15 approaches zero as N goes to ∞, leaving

$$\hat{z}(t) = \frac{1}{N}\sum_{k=1}^{N} x_k(t)\Big|_{t=0}^{1023\ \Delta t} \tag{16}$$

i.e.; the resultant time history is dependent only on the desired portion of the signal.

An acquisition code was used to implement equation 15 for N = 25, 50, 100, 200, 400 and 800. This was done to determine the number of averages required to achieve a "clean" time history, from which an estimate of the "true" power spectral density can be determined by taking the FFT of the resultant time history.

A schematic of the instrumentation used to conduct the study of this SEM is provided in figure 6. The data acquisition routine used a digital signal processing chip to acquire two data channels simultaneously at a user-selected sample rate up to 100 kHz. For the current study, the sample rate was set to 10 kHz and two measurement microphones were used. Independent analyses (using the equations given above) were conducted for each measurement signal, and the results were compared to data acquired with the FFT analyzer. The pure tone signal at microphone 1 was set to be 3 dB above the local background noise. The pure tone signal at microphone 2 was measured to be 1.5 dB below that at microphone 1 when the random noise generator was disengaged. The difference in phase between the two microphones was measured to be 144.8°.

Figure 7 provides a comparison of the extracted signals using a range of 25 to 800 averages, as was done with the other SEM studies. After only 25 averages, the local S/N was significantly improved. This improvement increases with an increasing number of averages. Figure 8 provides another view of the same data for the test frequency (1 kHz). For convenience, lines have been drawn on the plots to correspond to the results at 800 averages. This was done to help indicate how fast the data are converging. It is interesting to note that the data converged quite well after a minimal number of averages. Note also that the difference between the two results (1.39 dB and 143.82°) is almost the same as was measured with the FFT analyzer with the random noise generator disengaged.

This method would appear to be very attractive for continued usage. However, it requires a two step process in which the data is first acquired and stored onto a storage media, and is then subdivided into a number of synchronous blocks for analysis. This procedure is time consuming, making it unattractive for regular usage. Nevertheless, this method may prove to be viable for cases where a large number of microphones are needed, since it can be conducted for a larger number of microphones at almost the same speed as for a few microphones.

Summary

The measurement of pure tone acoustic pressure signals in the presence of masking noise, often generated by mean flow, is a continual problem in the field of passive liner duct acoustics research. In support of the Advanced Subsonic Technology Noise Reduction Program, three signal extraction methods (SEM) were investigated for conducting measurements of advanced duct liner concepts in harsh, aeroacoustic environments: (1) a coherence-based method, (2) a cross-spectrum-based method, and (3) a time-history signal enhancement method. These methods were compared to a baseline data acquisition configuration, in which an FFT analyzer was used to read the spectrum directly.

Each of the three SEM's was shown to be at least as accurate as the baseline. The coherence-based method was shown to be quite efficient, and is recommended as the method of choice for cases where only the magnitude component is required. The cross-spectrum-based method was shown to be quite robust, both in accuracy and efficiency. Although not quite as efficient as the coherence-based method, the cross-spectrum-based method provides the phase component. It is thus recommended as the 'work-horse' method for regular data acquisition.

Because of instrumentation difficulties, the time-history signal enhancement method was tested for only a few selected conditions. The results of this testing indicated that this method is also capable of providing quality data. However, this method is time-consuming. It is thus recommended that this method be used only when more than three microphones are to be measured simultaneously.

References

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3. Meirovitch, L.: "Analytical Methods in Vibrations," The Macmillan Company, New York, 1967.

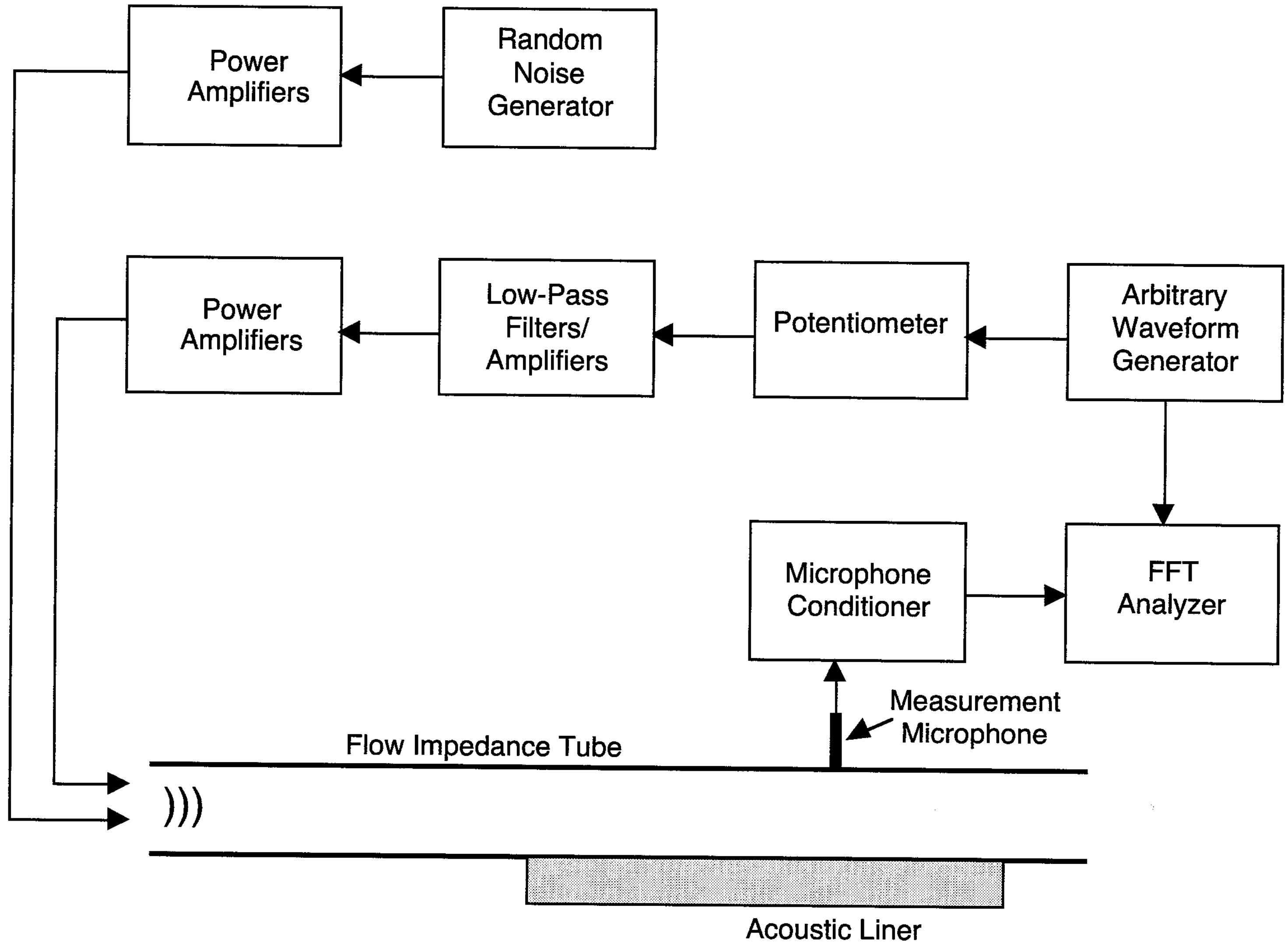


Figure 1. Schematic of instumentation used in baseline study

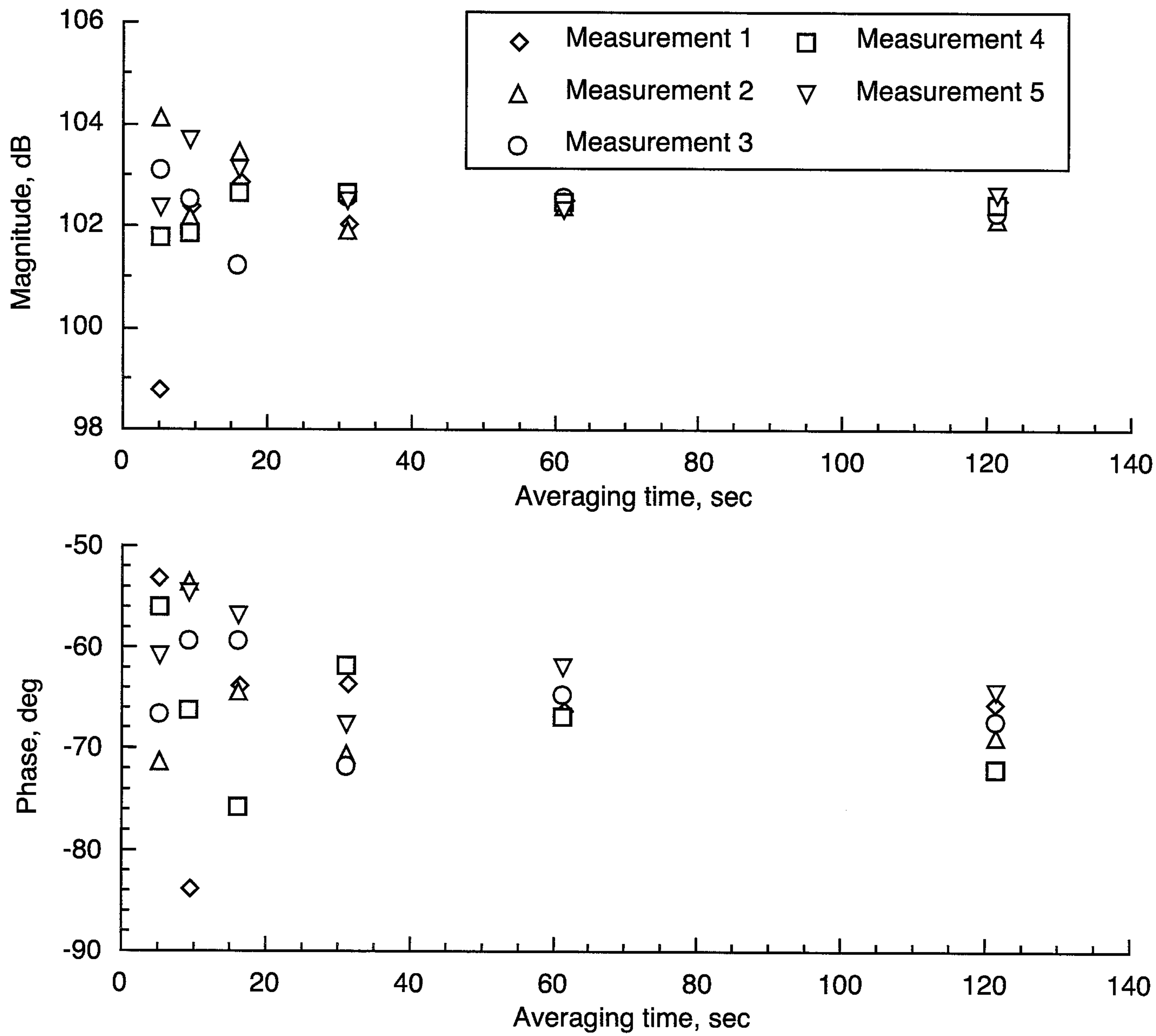


Figure 2. Demonstration of variability of measured signals using baseline method

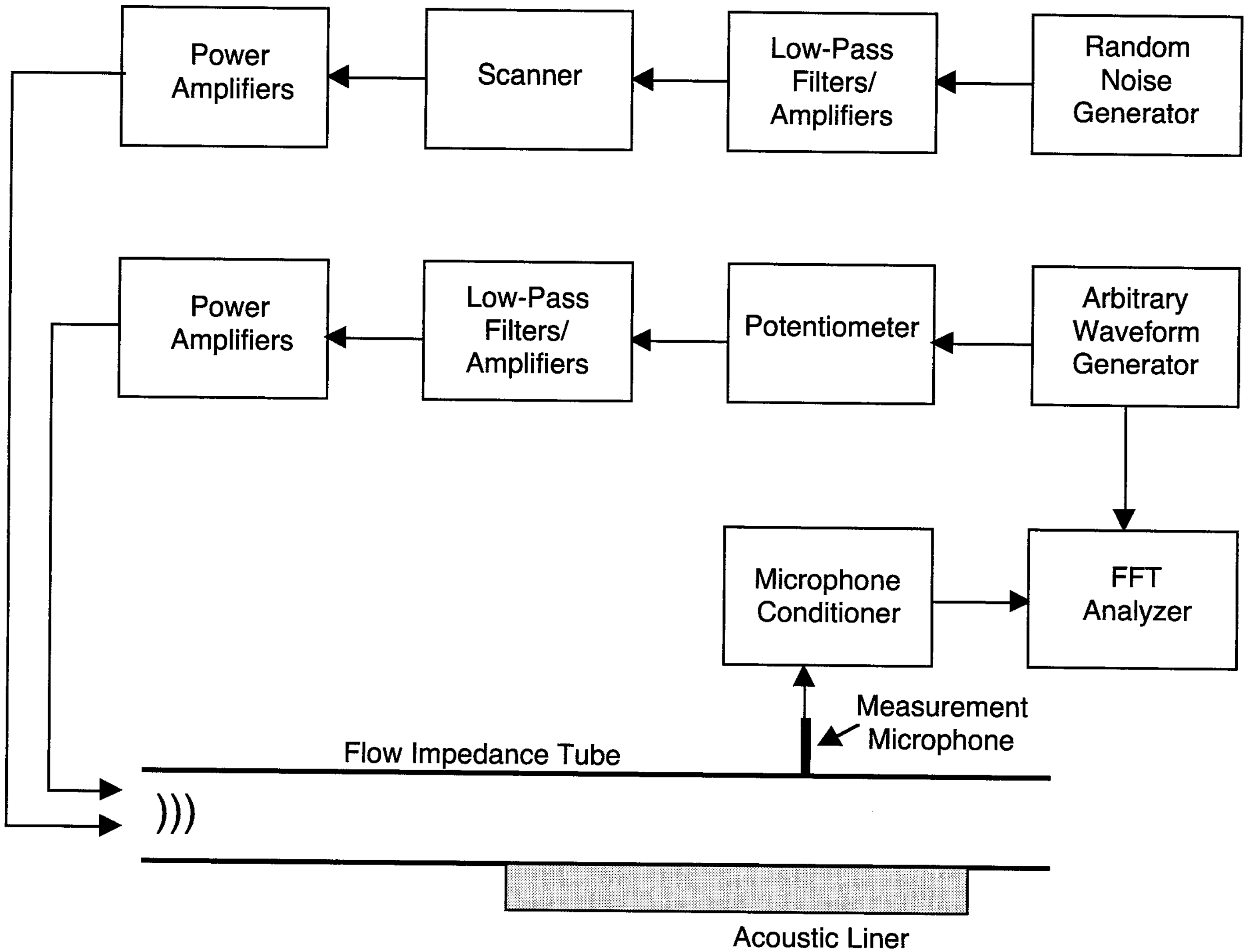


Figure 3. Schematic of instumentation used in studies of coherence-based method and cross-spectrum-based method

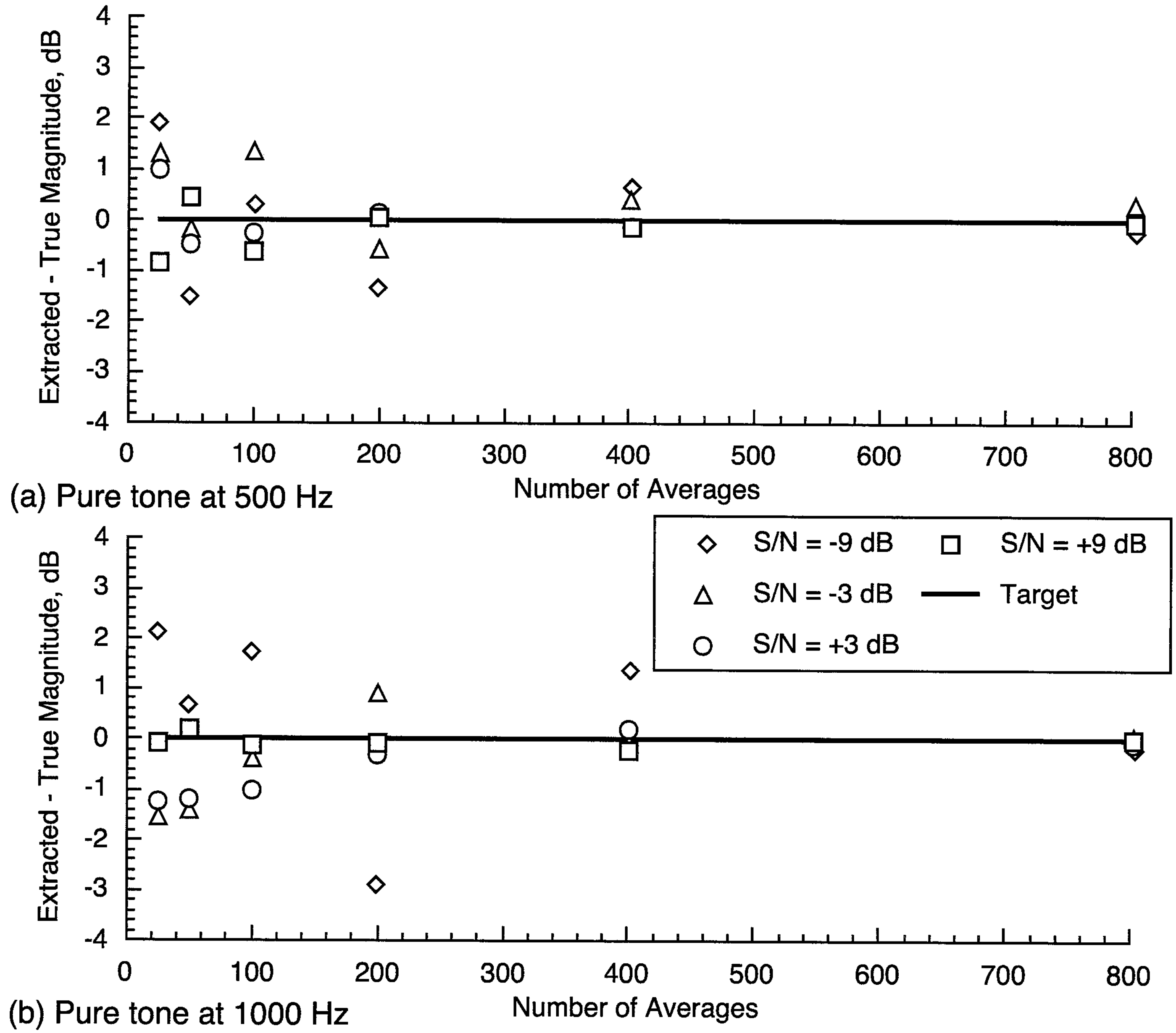
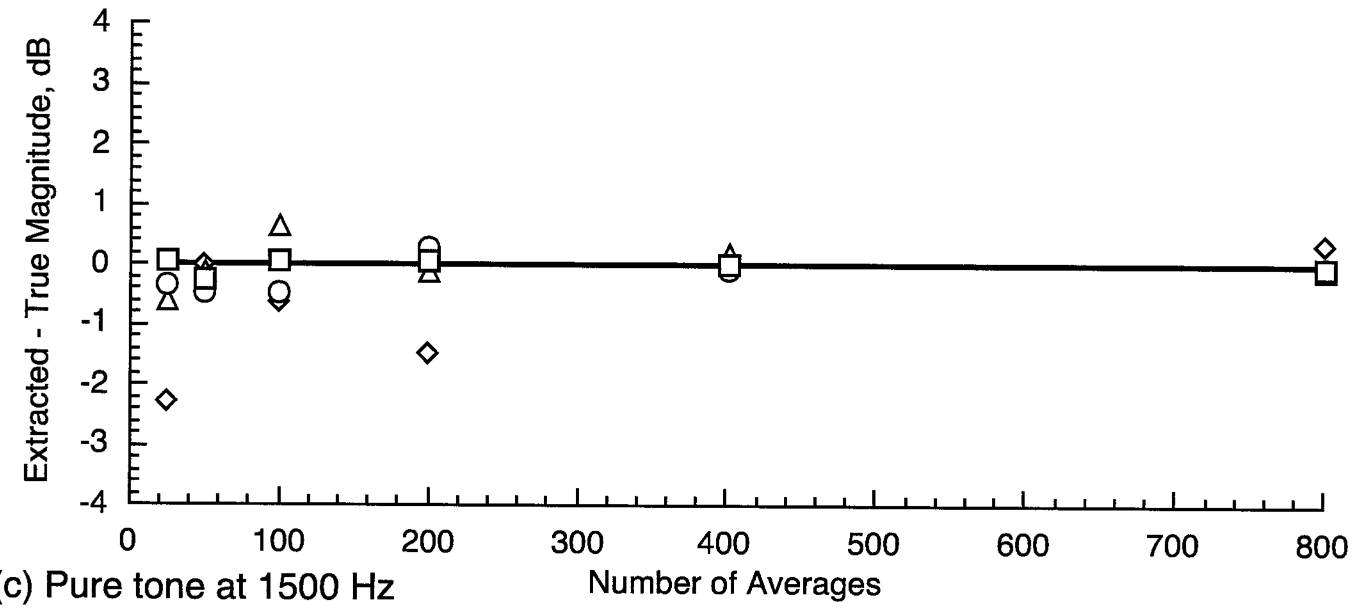
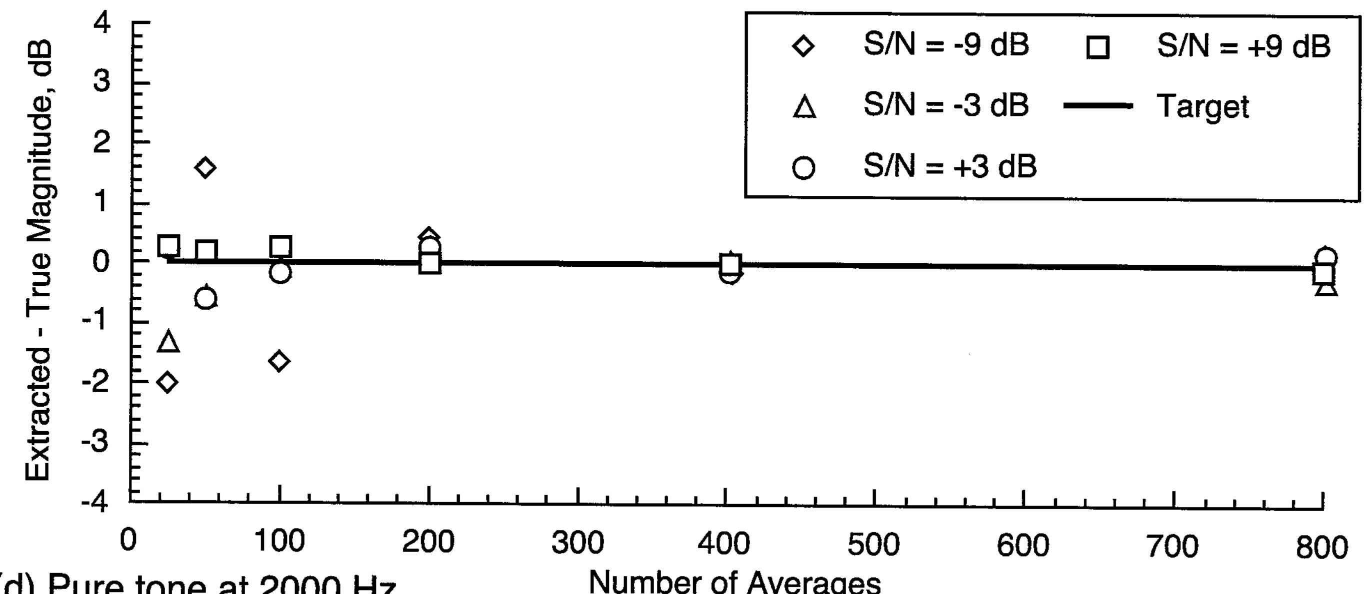


Figure 4. Comparison of errors (extracted magnitude - 'true' magnitude) for four signal-to-noise ratios using coherence-based method

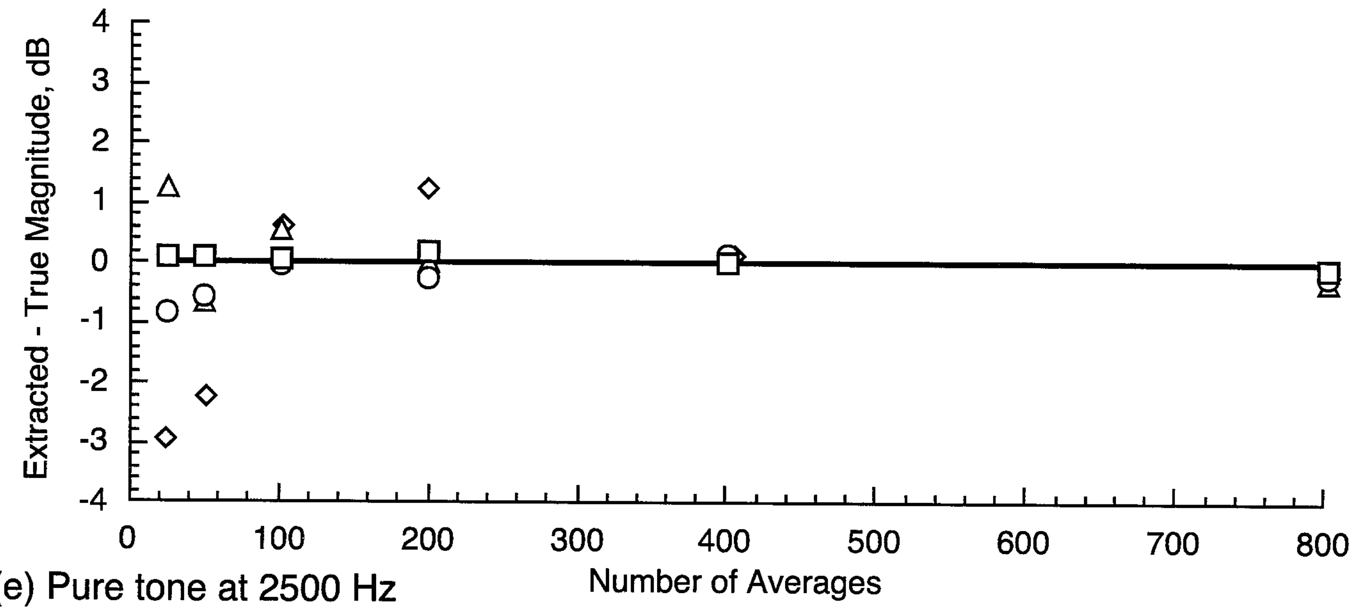


(c) Pure tone at 1500 Hz

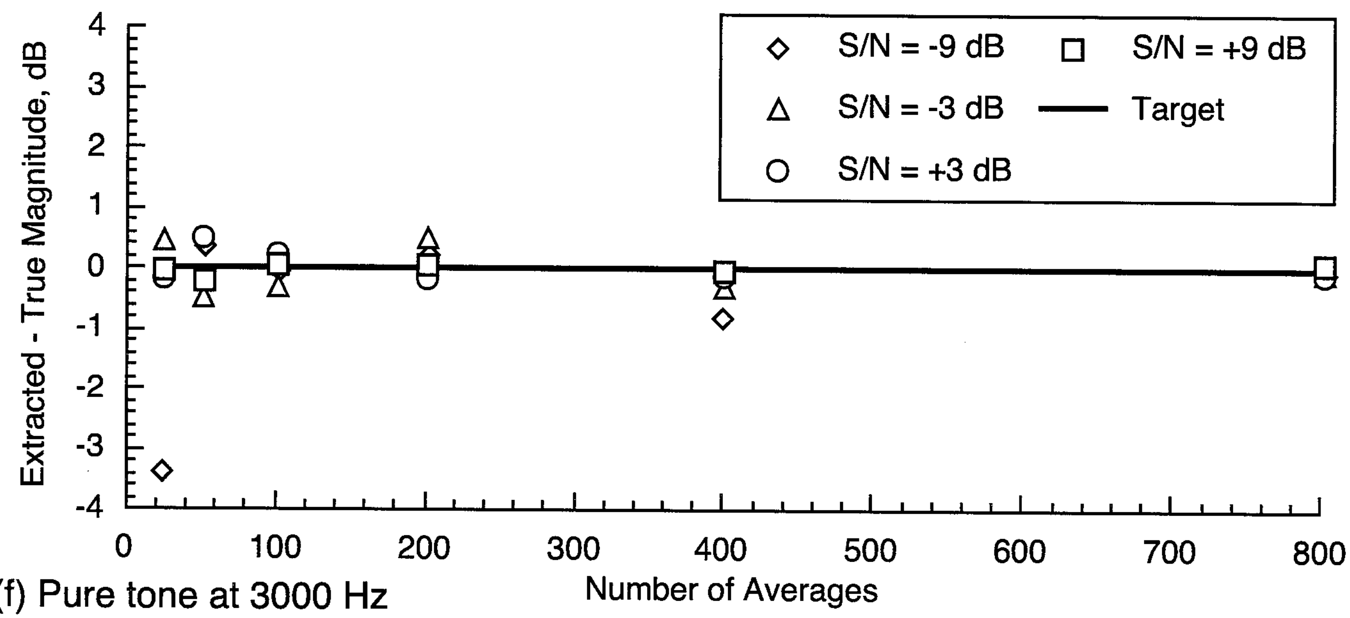


(d) Pure tone at 2000 Hz

Figure 4. (Continued)

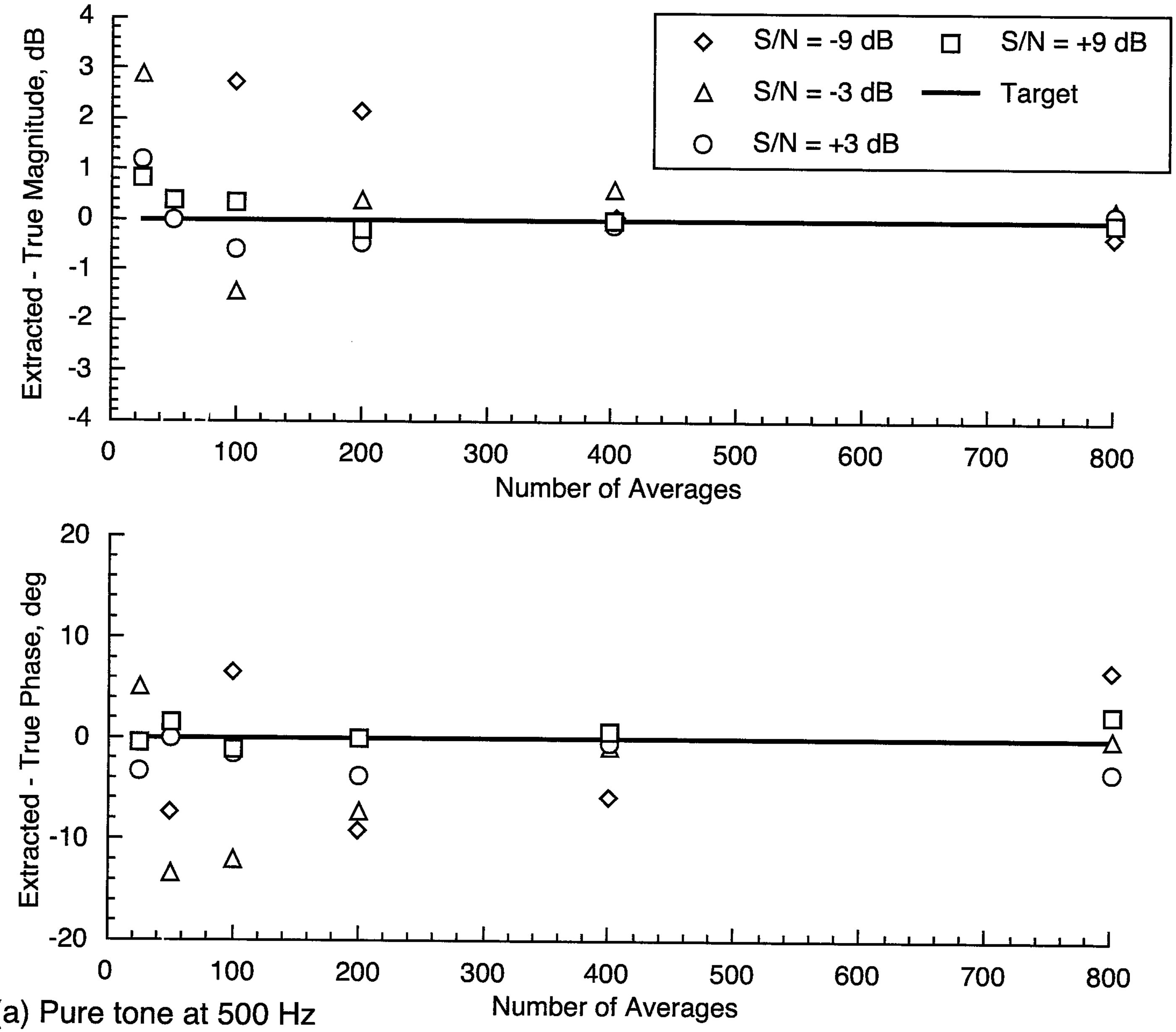


(e) Pure tone at 2500 Hz



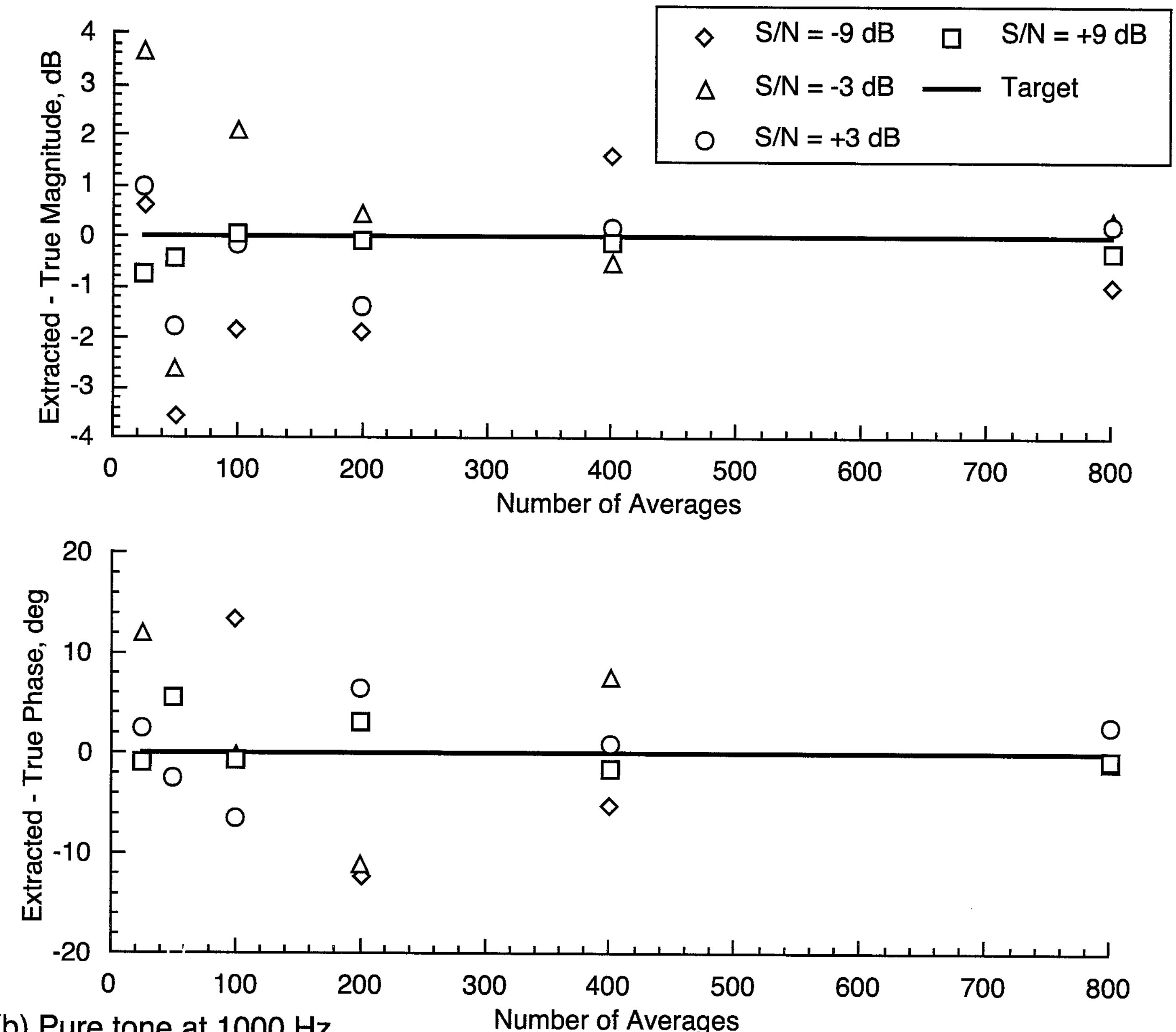
(f) Pure tone at 3000 Hz

Figure 4. (Continued)



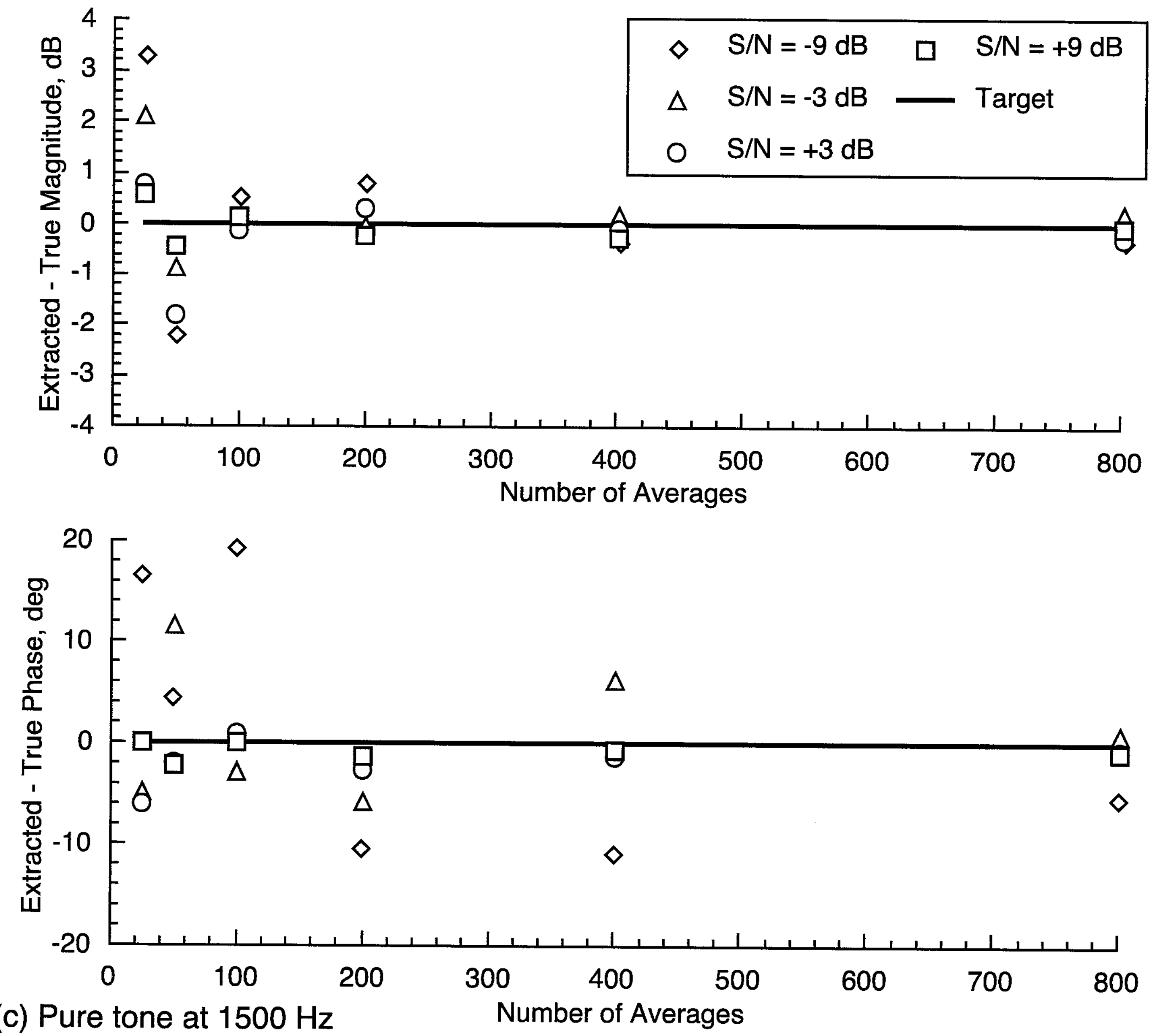
(a) Pure tone at 500 Hz

Figure 5. Comparison of errors (extracted signal - 'true' signal) for four signal-to-noise ratios using cross-spectrum-based method



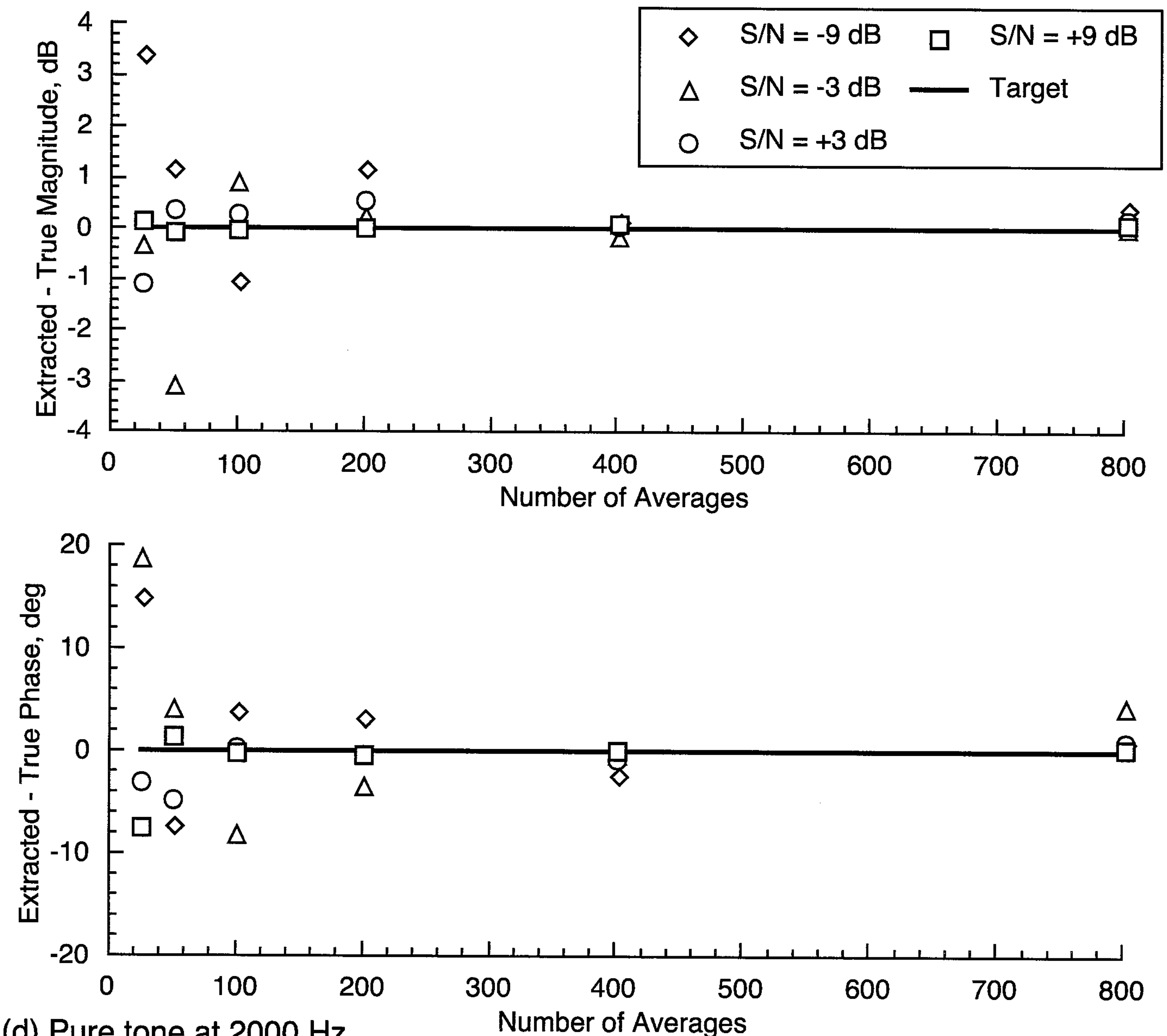
(b) Pure tone at 1000 Hz

Figure 5. Continued



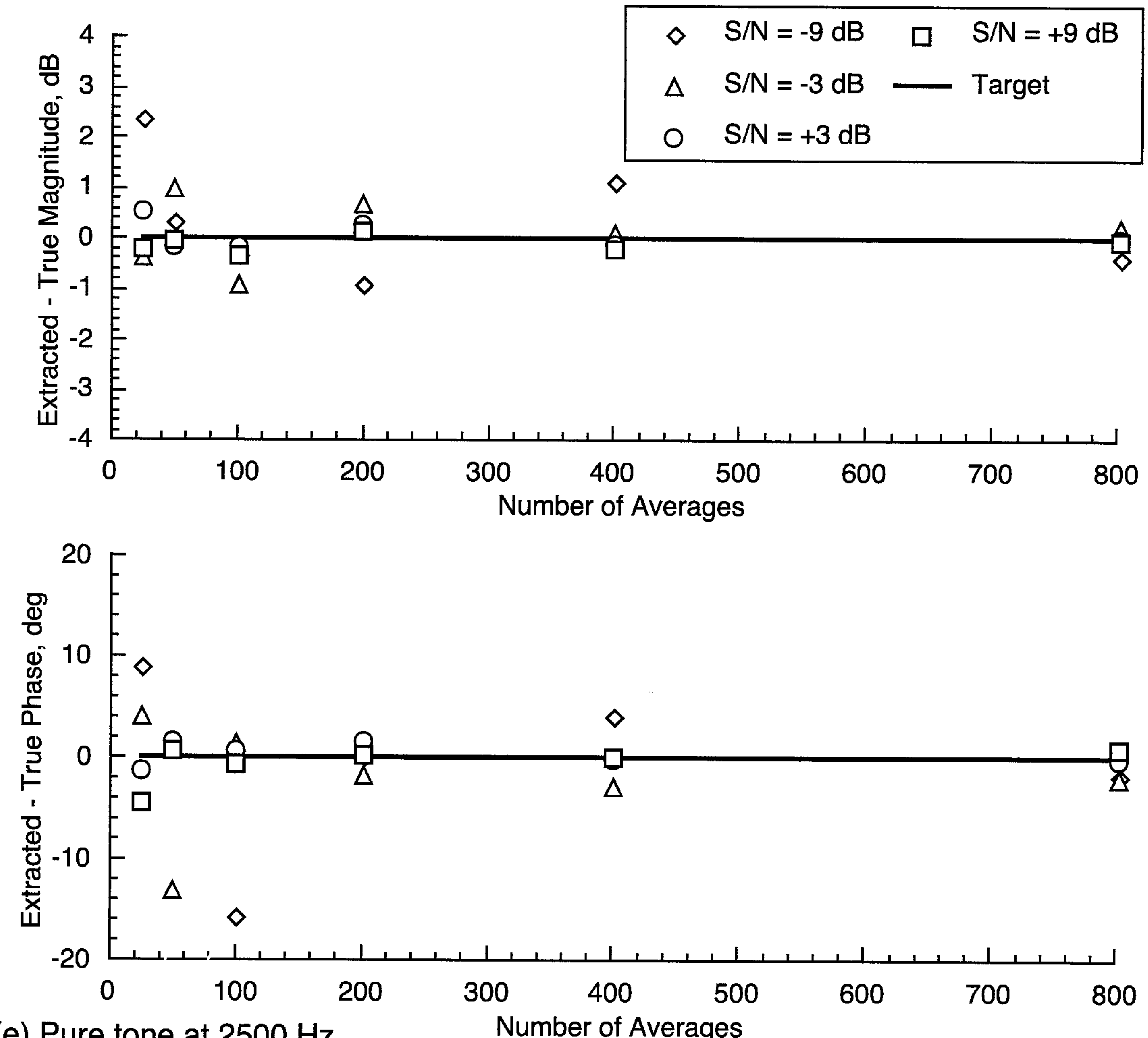
(c) Pure tone at 1500 Hz

Figure 5. Continued



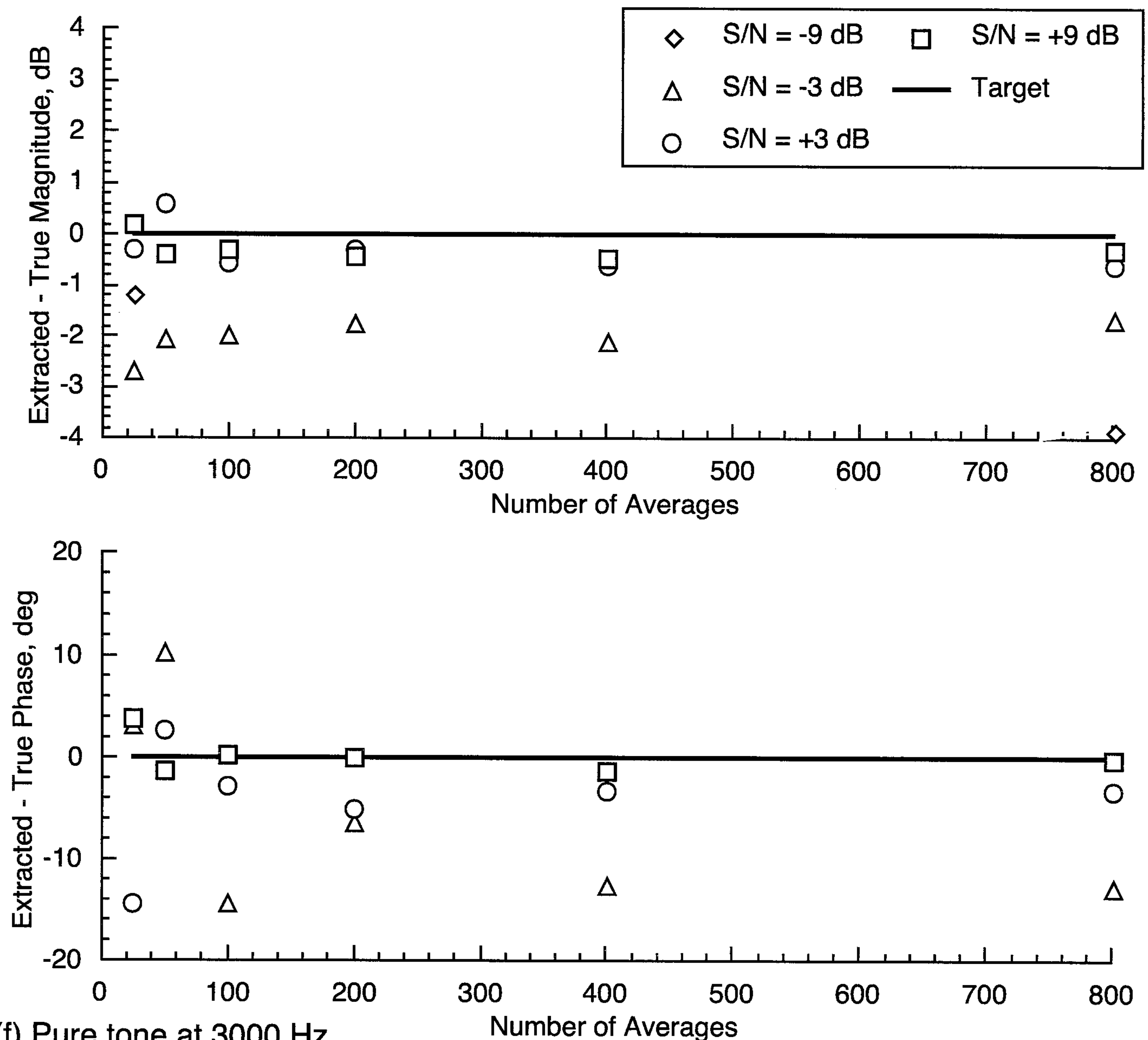
(d) Pure tone at 2000 Hz

Figure 5. Continued



(e) Pure tone at 2500 Hz

Figure 5. Continued



(f) Pure tone at 3000 Hz

Figure 5. Continued

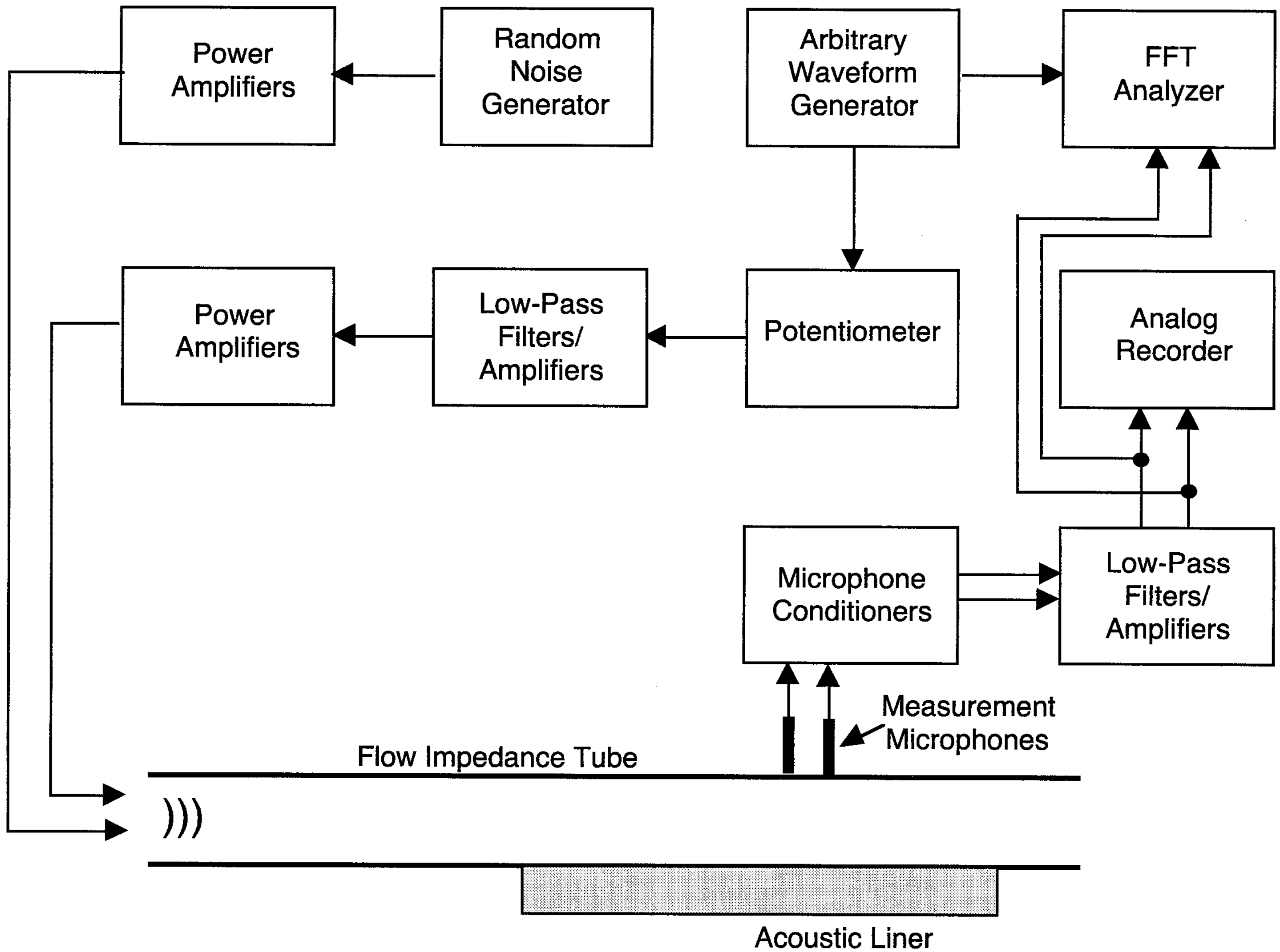


Figure 6. Schematic of instumentation used in study of time history signal enhancement method

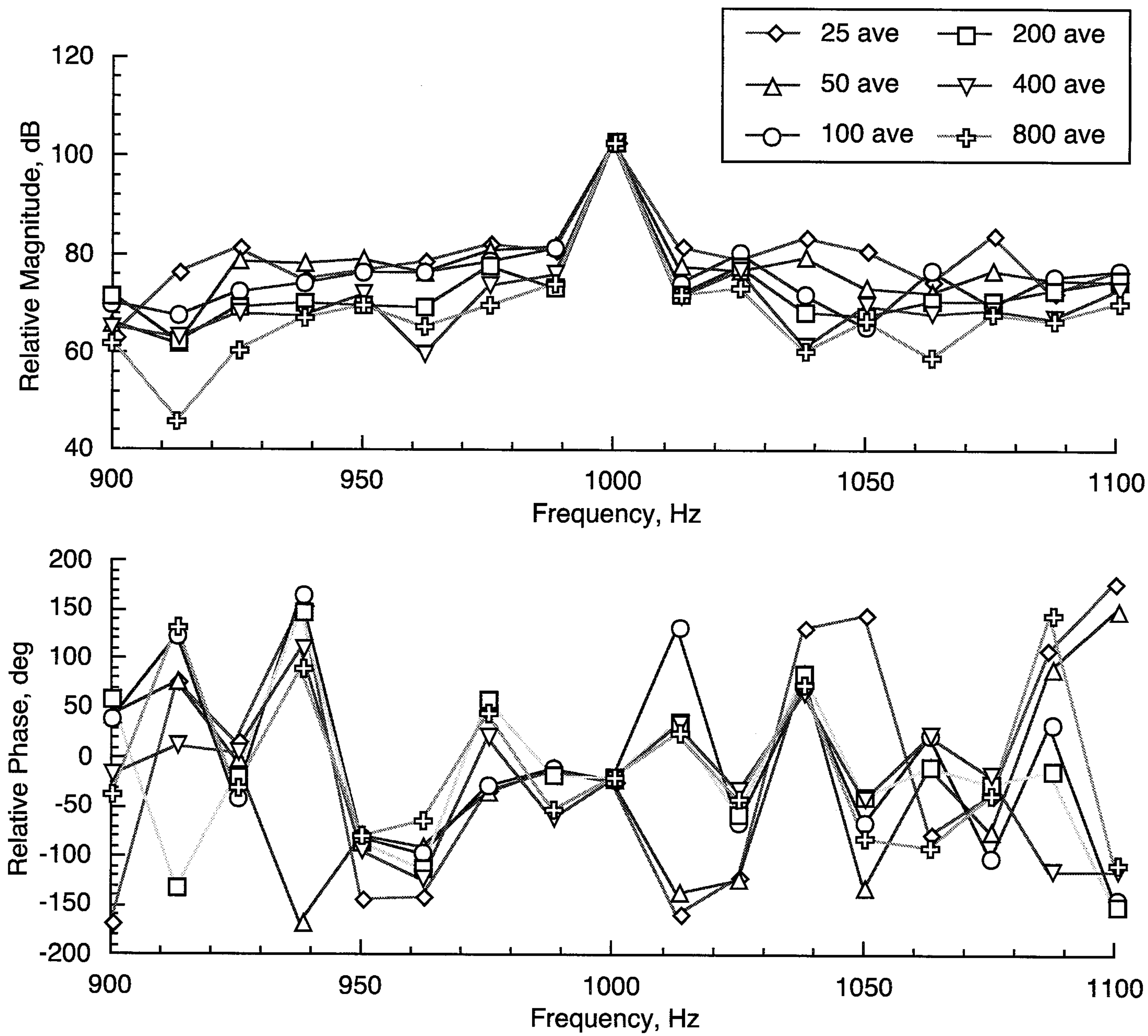


Figure 7. Comparison of extracted signals for six sets of averages using time history signal enhancement method

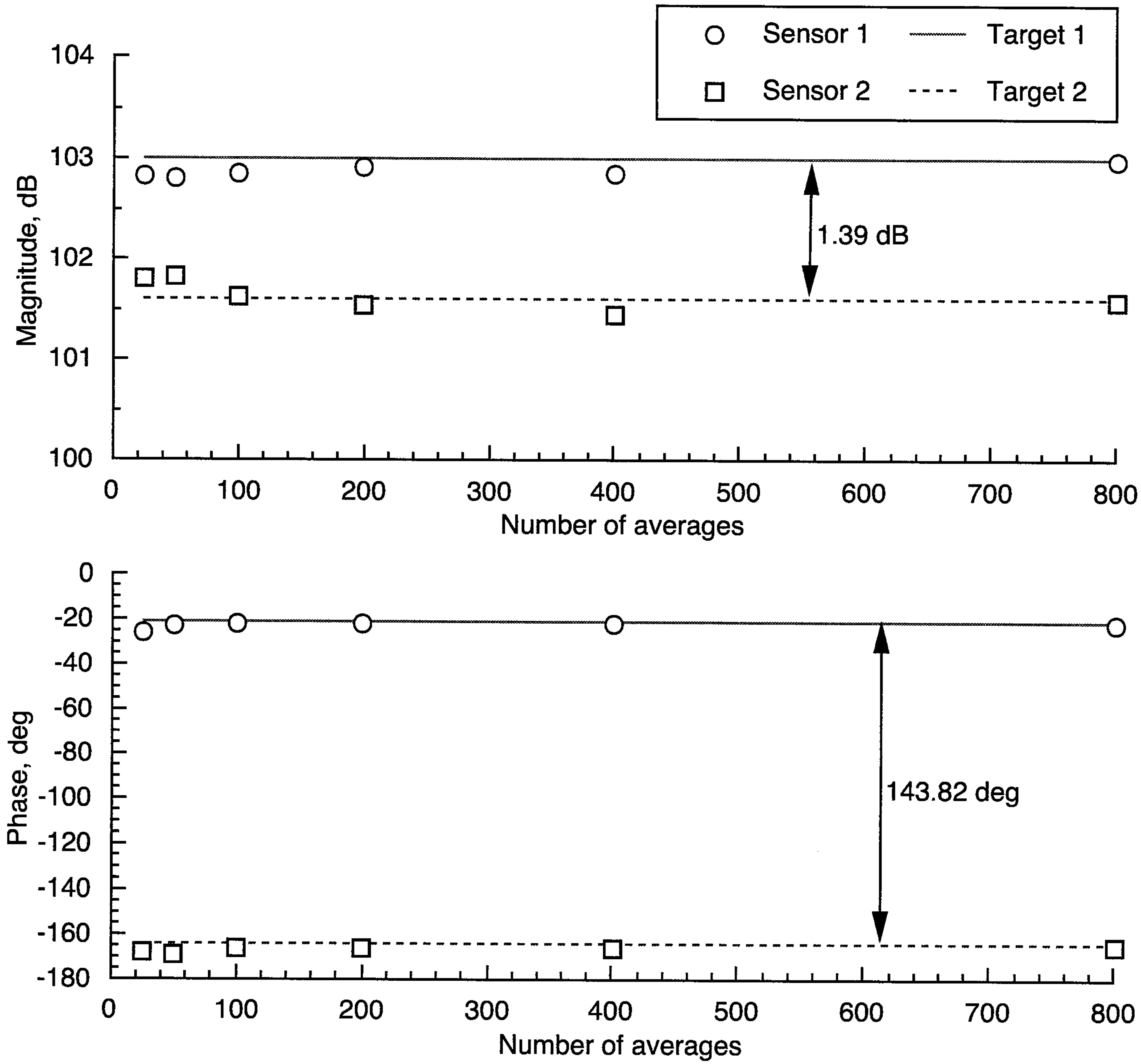


Figure 8. Comparison of extracted signals for two sensors using time history signal enhancement method

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