MEASUREMENT OF THE DYNAMIC TRANSFER CHARACTERISTICS OF MULTIBAND SIGNAL PROCESSING SYSTEMS BY TIME DELAY SPECTROMETRY

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ABSTRACT

Multiband broadcast signal processing systems have defied measurement of their dynamic transfer characteristics under the heavy processing ratios commonly in use. The high noise immunity of Time Delay Spectrometry and the averaging capability of a microprocessor based TDS analyzer are combined the investigate the dynamic complex transfer characteristics of several popular processing systems. Measurements will be demonstrated with near inaudibility of the test signal.

BROADCAST SIGNAL PROCESSING

Broadcast signal processing systems are a special combination of audio and radio frequency signal processing. It is common for the audio signal feeding the transmitter to be split into as many as six frequency bands, compressed, peak limited, and even expanded within those bands, recombined, and sent to the transmitter. An additional stage of peak limiting is often incorporated within the modulation system. FM systems (including television audio) use pre-emphasis. For these systems, this final peak limiter is applied to pre-emphasized audio (or with pre-emphasis in the side chain) to prevent overmodulation. With amplitude modulated transmitters, asymmetrical peak limiting is often employed.

The complexity of these systems has always raised questions about their static and dynamic characteristics. Since these signal processing systems are, by their very nature, changing their transfer characteristics with the signal applied, use of audio test signals alone as excitation will not yield their response to a dynamic signal (i.e., program material). If, however, a test signal could be mixed with program material such that the test signal is small in comparison to the program material, the processing system will be responding to the program material rather than to the test signal.

Specialized broadcast signal processing systems thus fall into two distinct groups. The first consists of devices inserted directly into the audio chain between studio output and transmitter input. The signal appears as audio at both the input and output of this type of processor. Figures 1 a and 1 b are examples of this group of processor. The second group of processors are built into the broadcast station's stereo generator. Because the two functions are combined into a single unit, signal processing may occur both before and after generation of the stereo signal. Both single band and multiband processing are used. Signal processing after the stereo generation stage has certain advantages, and this approach has come into widespread use. The input signal to this type of processor is discrete left/right audio, but the output consists of the matrixed and modulated main channel L+R and stereo L-R sub-carrier used for FM broadcasting. The audio output also is pre-emphasized. Figure 2 illustrates this type of processor.

SPECTRAL ENERGY DISTRIBUTION FOR BROADCAST AUDIO

Broadcast audio consists primarily of two basic kinds of sources. The first consists of the music and commercials produced outside the radio facility. This material is normally compressed and peak limited with the pre-emphasis and dynamic range limitations of the phonograph record or analog tape recorder in mind before reaching the broadcast facility. Although its high frequency headroom is considerably greater, material released on compact disc is often processed in about the same way. The spectral content and dynamic characteristics of all of these materials are generally well controlled for the nature of the program material.

The second kind consists of live announcers and field audio from sources such as news and sports. In some broadcast facilities (primarily the larger ones) this material is also processed, but not usually as well as the commercially recorded music and commercials. In many broadcast facilities it is not processed at all prior to reaching the main processor; its spectrum is weighted toward the midrange and has more dynamic range.

The spectrum of the first type of program material at the input to the broadcast processor approximates pink noise for the frequency range between 50 Hz and 1.5 KHz, and pink noise with an additional 3 dB/octave rolloff for the range between 1.5 KHz and 15 KHz. Since pink noise can be thought of as white noise with a 3 dB/octave rolloff, the 1 KHz-15 KHz region may also be seen as rolled off 6 dB/octave as compared to white noise. Figure 3 is the spectrum of this material as it would be indicated on a constant percentage bandwidth real time analyzer.

At the output of the broadcast processor, the spectrum is essentially the same as at the input except that the previously unprocessed audio looks more like the commercially recorded material and the peak to average ratio is reduced by several dB.

TIME DELAY SPECTROMETRY AND NOISE IMMUNITY

Time Delay Spectrometry has been thoroughly documented by its inventor, Richard C. Heyser, and others. [1, 2, 3, 4, 5] It is a powerful method for the analysis of two-port networks, and one of its major advantages is its noise immunity. Another is that it is valid for non-linear systems. The noise immunity is a function of the signal bandwidth, measurement bandwidth, and the length of time of the measurement.

The immunity of TDS energy-frequency measurements to white noise is given by

$$S/N = 10 \operatorname{Log10}[(\text{Program Bandwidth}) / (\text{Measurement Bandwidth})] \quad (1)$$

For critical bandwidth, (maximum frequency resolution for a given sweep rate) the measurement bandwidth is equal to the square root of the sweep rate and (1) may be expressed as

$$S/N = 10 \operatorname{Log10}[(\text{Measurement Time}) \times (\text{Program Bandwidth})] \quad (2)$$

If the interfering signal were pink noise, band-limited at 50 Hz and 15,000 Hz, the above expression would represent the signal to noise ratio at mid band (866 Hz) and the immunity at any frequency would be improved by 3 dB/octave of increasing frequency. For the spectral distribution of program audio, the mid-band point is approximately 500 Hz. The signal to noise degrades by 3 dB/octave of decreasing frequency down to 50 Hz, improves by 3 dB/octave of increasing frequency to about 1 KHz, and improves by 6 dB/octave above 1 KHz.

THE TECHRON TDS SWEEP

The Techron implementation of TDS [4, 6] employs an especially useful variation of the linear sweep. To eliminate the perturbations in response which would otherwise be obtained by sweeping through or near zero frequency, two sweeps (one a cosine sweep, the other a sine) through zero frequency may be used. The negative frequency portion of the sweeps extends to only one quarter of the frequency of the positive going sweep. A sweep to provide analysis between zero and 1,000 Hz would thus sweep the range from -250 Hz to 1,000 Hz twice in succession. An added benefit of the technique for this application is that the measurement time in the low frequency portion of the spectrum is doubled, improving the signal to noise ratio by 3 dB for those low frequencies.

AVERAGING AND NOISE IMMUNITY

This noise immunity may be further improved by averaging. If the signal delay through the medium is constant, complex (i.e. magnitude and phase) averaging may be used, and the improvement is given by

$$S/N = 10 \operatorname{Log}_{10}(\text{Nr of sweeps}) \qquad (3)$$

The averaging may also be thought of as simply increasing the measurement time, and that new time simply substituted into the corresponding expression (2). This lengthening of the measurement time required for averaging and the desired frequency resolution for the measurement set some practical restrictions on the sweep times and filter bandwidths which will be used in the analyzer. The author has found bandwidths of about one-third critical bandwidth to be good compromises for practical measurements. Figure 4 shows the signal to noise ratio (noise immunity) to be expected from a TDS measurement using a complex average of 64 sweeps to 15,000 Hz at a sweep rate of 2,000 Hz per second.

If the travel time through the system changes with successive sweeps, complex averaging will lose non-correlated data and scalar averaging must be used. In that instance, the improvement in noise immunity due to averaging will be approximately one half of that provided by complex averaging.

THE TEST METHOD

Time delay spectrometry thus allows recovery of a test signal in the presence of a considerable amount of noise. Currently available implementations of TDS systems are based on sine waves swept at a linear rate through the portion of the audio spectrum to be investigated. The TDS sweep signal is mixed with program audio at a level well below program peaks, so that the test signal is a negligible part of the input to the processing system.

Since the signal processing system is changing its transfer characteristic (gain) on an instantaneous basis, the amplitude response to this sweep will vary with time. The measured response will thus be a function both of the program applied at the instant the sweep is at each frequency and the overall response of the system. If many measurements were made of the system, each would be different.

If many sweeps could be averaged, the instantaneous variations in system gain due to applied program material would be averaged out and the average response of the system obtained. The Techron implementation of TDS, developed by Gerald Stanley and using software created by Heyser, allows this type of measurement. The Techron system was also well documented at the time of its introduction.

The software allows averaging of either scalar or complex data using four, sixteen, or sixty four sweeps. If complex averaging is used, the averaged data may be stored on disk and printed to hardcopy. Scalar averaged data may not currently be written to disk and may not be used at the nearly inaudible levels of which complex averaging is capable. Both types of data may be differenced to other data for measurement of gain reduction or comparison of different devices.

With practical values of sweep rates and bandwidths, noise immunity on the order of 36 dB is available at midfrequencies, improved by 18 dB with the complex averaging of 64 sweeps. If the program audio has a form factor of 10 dB, the overall noise rejection would allow mid-frequency sweeps at nearly 60 dB below 100% modulation. By 100 Hz, however, the noise immunity for the same level of test signal has degraded by 10 dB.

With highly compressed pop music formats, good data may be obtained with the sweep 60-70 dB below program peaks. At these levels the sweep will be quite well masked by program material except during periods of silence ("dead air"). For all practical purposes, dead air does not exist in such formats, and measurements can be made on the air with the audience unaware of the sweep. With less compressed formats, the high frequency portion of the sweep (above about 2 KHz) must be further reduced in level to make it inaudible. Measurement in less compressed formats and those with more dead air require that the test signal be at a lower level.

PRE-EMPHASIZED MEASUREMENT

Some means is therefore needed to improve the noise immunity at low frequencies. Several approaches could be taken. A pre-emphasized test signal could be used, with the sweep having a 3 dB/octave rolloff applied to it prior to its being mixed with program material, and a corresponding 3 dB/octave de-emphasis applied to the processed signal prior to analysis. The Techron system contains such a rolloff filter which may be switched into the test oscillator signal path, although current software does not support it.

For the spectral energy distribution of broadcast audio, an alternative pre-emphasis network might be a 75-150 microsecond 6 dB/octave low pass applied between the generator and the device under test, with a corresponding de-emphasis network at the input to the analyzer. An additional 3 dB/octave low pass network for low frequencies would be useful but more difficult to implement. It should operate between the range of 100 Hz and 1 KHz, shifting to a 6 dB/octave high pass at 50 Hz. Figure 5a is the desired response of the generator's network. Such a pre-emphasized measuring system has not yet been implemented by the author. Figure 5b is the signal to noise ratio to be expected from these pre-emphasized measurement systems. The sweep signal to the simple 75 microsecond network was raised 10 dB at the input to the filter. The signal from the combined network was assumed to be unchanged from the flat measurement condition.

Alternatively, it might be desirable to split the spectrum into two or more segments for analysis. Thus, data for the range from 1,000 to 15,000 Hz is obtained from one set of averaged sweeps and combined with a second set for the range of 30 - 1,000 Hz. The experimental work done by this author uses this method of measurement. Figure 6 shows the signal to noise ratio of two possible setups of to analyze from 50 Hz to 1,000 Hz and 2,000 Hz respectively, using 64 sweeps of 19 seconds each.

OPERATING LEVELS

In order not to be audible in the transmitted audio, the TDS sweep at high frequencies must be at least 70 dB below 100% modulation level. If gain reduction of 10 dB occurs with processing, it must be 80 dB down at the processor input. With the 150 microsecond pre-emphasis suggested above, the sweep could be maintained at 60 dB below 100% modulation at the input and still be expected to provide good data down to a low frequency limit of about 150 Hz. Use of the additional 3 dB/octave low frequency pre-emphasis could improve that low frequency limit to about 50 Hz.

Audibility is a particularly important factor, since in major cities, most stations do not sign off for maintenance but rather maintain standby transmitters and even antennas. Many others will do so for no more than one night a year and there are usually many more important projects needing the four hours available.

CHOOSING TDS PARAMETERS AND INTERPRETING THE DISPLAY

Any choice of TDS measurement parameters defines certain limitations on the measurement. These limitations must be clearly understood to avoid making incorrect interpretations of the results of any measurement. As in any swept frequency measurement, the choice of a sweep rate and measurement bandwidth will define the frequency and time resolution of the data. This well known relationship is given by

$$\text{(Frequency Resolution)} = \frac{\text{(Sweep Rate)}}{\text{(Bandwidth)}} \tag{4}$$

and

$$\text{(Time Resolution)} = \frac{\text{(Bandwidth)}}{\text{(Sweep Rate)}} = \frac{1}{\text{(Frequency Resolution)}} \tag{5}$$

There are several effects of these relationships which must be kept in mind in the course of setting up for and interpreting the measurements. First, the response of most systems will be zero at zero frequency. First, the low frequency limit of a given measurement is one half of the frequency resolution for that measurement, and any details of that response which are more closely spaced that the frequency resolution will be smoothed. Second, the time resolution may be thought of as a time window, during which the data will be taken. Any system response which occurs outside that window, either before or after, will be rejected by the analysis filter, and any response within that window will appear to have occurred at the same time.

A limitation of the Techron system, used in this work, is that the data is stored in only 400 discrete frequency registers. Since TDS uses a linear sweep, this means that when converted to a log display these points are sometimes not close enough together to yield the desired resolution, even though the above relationships might indicate otherwise. One must thus be careful not to misinterpret a smoothness in, or lack of, a low frequency response for a given measurement.

EXPERIMENTAL WORK

The proposed measurement method was implemented using several popular broadcast signal processing systems as the device under test. The test setup of Figure 7 was used for the Orban Optimod 8100, a device known to use considerable amounts of all pass filtering in its processing. Since the 8100 is a stereo generator which provides composite, pre-emphasized audio at its output, a 75 microsecond de-emphasis network was needed to provide "flat" audio for analysis. Additional measurements were performed on other processing systems which do not perform pre-emphasis. The same setup was used for those devices, but the 75 microsecond filter was omitted.

The method of dividing the spectrum into two parts for analysis was used. As is seen from the documentation included with the data, sweep rates were such that a complete set of data (i.e., two sets of averaged sweeps) were obtained in about 45 minutes. The sweep signal was mixed with program at a level that was nearly inaudible, except during periods of silence. The program material used for the test was a compact disc reissue of a popular contemporary recording of acoustic music having a wide variety of sound dynamics. The systems under test were adjusted so that they were doing relatively large amounts of signal processing, representative of their use in competitive pop music formats. A tape recording of the mixed combination of program material and test signal will be included in the presentation. Data from these sweeps is shown in Figures 8A and 8B.

Another set of data was taken, using single sweeps for each of the two frequency bands at the same level of test signal but without program material. Since this level was well below the threshold of the signal processing, these sweeps represent the steady state transfer function of the device under test. The first sweeps, with program material, were subtracted from those obtained without program. The resulting data are the average dynamic gain reduction of the device, and are shown in Figures 9A and 9B.

The phase responses were then displayed and processed to obtain the group delay, with and without processing. The resulting phase data are shown in Figures 10A and 10B, and in differenced form in Figures 11A and 11B. The group delay data is shown in Figures 12A and 12B. The data for processed and unprocessed audio (i.e., with and without program audio present) show near perfect correlation, differing only in the presence of noise at the low frequency extremes for each set of sweeps for the averaged data. Note that the high frequency sweep begins to get noisy below about 2 Khz, but that it merges very well with the corresponding data taken below 2 Khz. An analysis of the low frequency data for the Optimod seemed to indicate a surprising loss of low frequency response below about 100 Hz. Further analysis of the group delay data showed, however, that the unit's all pass filter characteristic was producing so much delay that the frequencies below 100 Hz were present at normal amplitude, but were outside the time window of the TDS analysis filter. A new set of sweeps were made at the time offset indicated by the group delay measurements for low frequency information, and confirming data having good noise immunity obtained. The results of these new low frequency sweeps is shown as Figures 13A and 13B, which were a measurement of a single sweep only.

Another set of measurements was made with the same test setup but with the input level to the Optimod 8100 adjusted to provide considerably more gain reduction. Figure 14A shows the averaged amplitude response with and without program. Figure 14B is the gain reduction. Since this measurement extended to 20 KHz, the stereo generator's 15 KHz low pass filter is clearly visible in the response. It did not show up in the previous series of measurements because that sweep extended to only 15 KHz. The program material used for both this measurement and for the previous series was the same.

Figure 15A shows the low frequency response of a typical compressor set for an indicated 6 dB of gain reduction. Figure 15B is its phase response, both with and without signal. Note that there is no significant difference between the two curves.

Figure 16 shows the gain reduction of a popular processor which combines the functions of compression, limiting, and AGC in a single unit. It was adjusted to provide an indicated 10 dB of gain reduction for the measurement. The same program material was also used for this measurement.

AUDIBILITY OF THE TDS SWEEP SIGNAL

A listening test was set up by the author wherein program audio (a contemporary music FM broadcast station) was adjusted to provide 95 dB program peaks at the listening position. The TDS sweep signal was then added to the listening room and increased in level until audible. That level was then measured in the absence of program. At 30 dB SPL, the sweep was clearly audible at high frequencies (above about 5 KHz) in periods of moderate modulation density. During periods of low modulation density (disc jockey speech) it was clearly audible at frequencies as low as 800 Hz. At no time was it audible at the lower frequencies. This result supports the feasibility of a pre-emphasized TDS sweep.

The program material used as input for the device under test was analyzed for energy content using the averaging function of the Techron TEF analyzer. It is believed to be representative of typical popular music in its spectral content. [7]

Spectral distributions were also obtained by monitoring the program material with a one-third octave real time analyzer. Three types of measurements were made. First, the RMS value of program peaks (actually, fast responding RMS values) for a three minute segment of highly dynamic music were accumulated in memory. Second, an accumulation of long-term integrated levels was performed. Third, the long-term averaged levels were observed for the same period and the general shape of the curve noted. A plot was made of the data, and the second set of data subtracted from the first. This differenced plot shows a relatively constant peak to average ratio of 10 dB below 1 KHz, decreasing to 6 dB between 1 KHz and 10 KHz. It is likely that this additional high frequency amplitude compression is due to the pre-emphasized peak limiting used to provide for recording pre-emphasis. All three sets of data from the real time analyzer show the same trends as that obtained from the FFT.

SUMMARY AND CONCLUSIONS

Time delay spectrometry is a very useful tool for the evaluation of signal processing systems under dynamic operating conditions. Good data may be obtained quickly if the audibility of the sweep can be tolerated (as when the test is run in the laboratory) and more slowly if the sweep must not be audible (as in a broadcast station which is on the air). Pre-emphasized measuring techniques provide the most efficient method of obtaining data. The ability of the system to reject interference to the measurement from program material is up to 70 dB, depending upon the length of time allowed to obtain data.

ACKNOWLEDGEMENTS

The author would like to thank Richard Heyser for his encouragement and for the software used to accomplish these results; and Harry Priester of the American Broadcasting Company, and John Freberg of WXRT for the loan of equipment for measurement.

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5) Biering, H. and Pedersen, O. Z. , "System Analysis and Time Delay Spectrometry, Parts I and II", Technical Review No. 1, No. 2, 1983, Bruel & Kjaer Instruments, Marlboro, Mass.

6) Bartlett, Bruce, "Testing Microphones With Time Delay Spectrometry", Paper presented at 76th Convention of Audio Engineering Society, October, 1984

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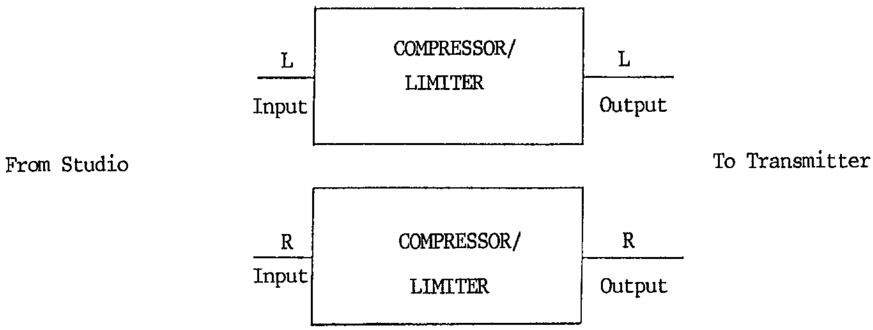


FIGURE 1A

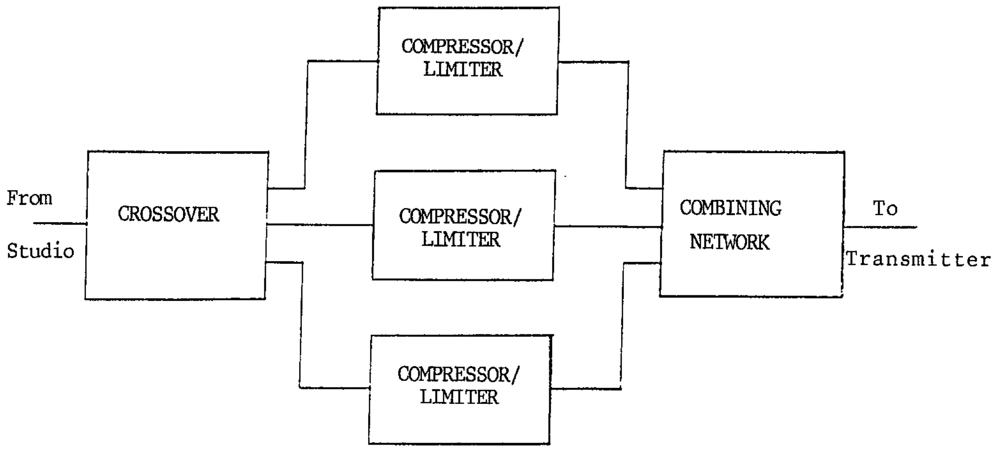


FIGURE 1B

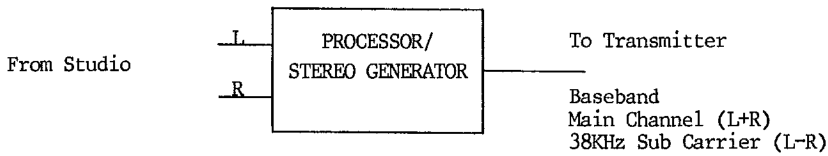

From Studio
L
R
PROCESSOR/
STEREO GENERATOR
To Transmitter
Baseband
Main Channel (L+R)
38KHz Sub Carrier (L-R)

FIGURE 2

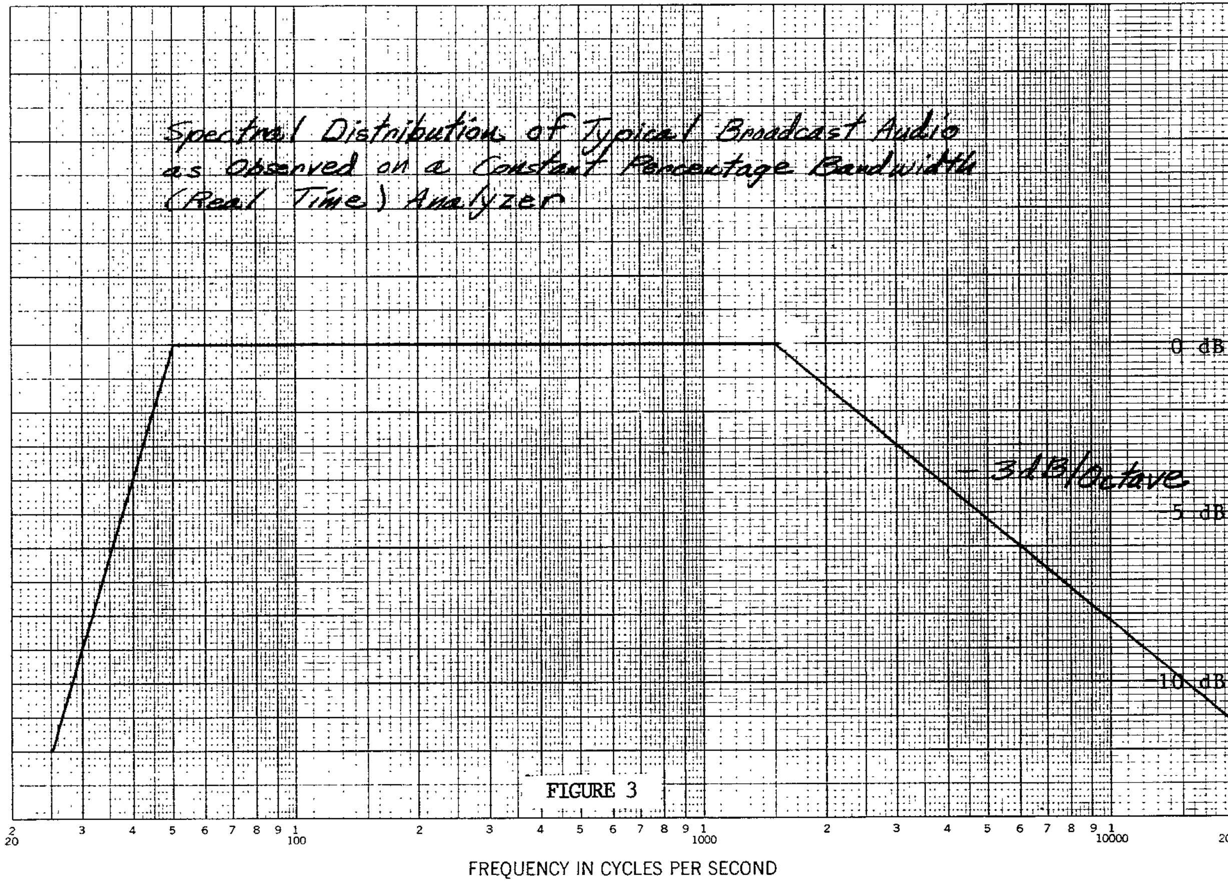

Spectral Distribution of Typical Broadcast Audio as Observed on a Constant Percentage Bandwidth (Real Time) Analyzer
0 dB
-3 dB/Octave
-5 dB
-10 dB
2
20
3
4
5
6
7
8
9
1
100
2
3
4
5
6
7
8
9
1
1000
2
3
4
5
6
7
8
9
1
10000
2
20000
FREQUENCY IN CYCLES PER SECOND

FIGURE 3

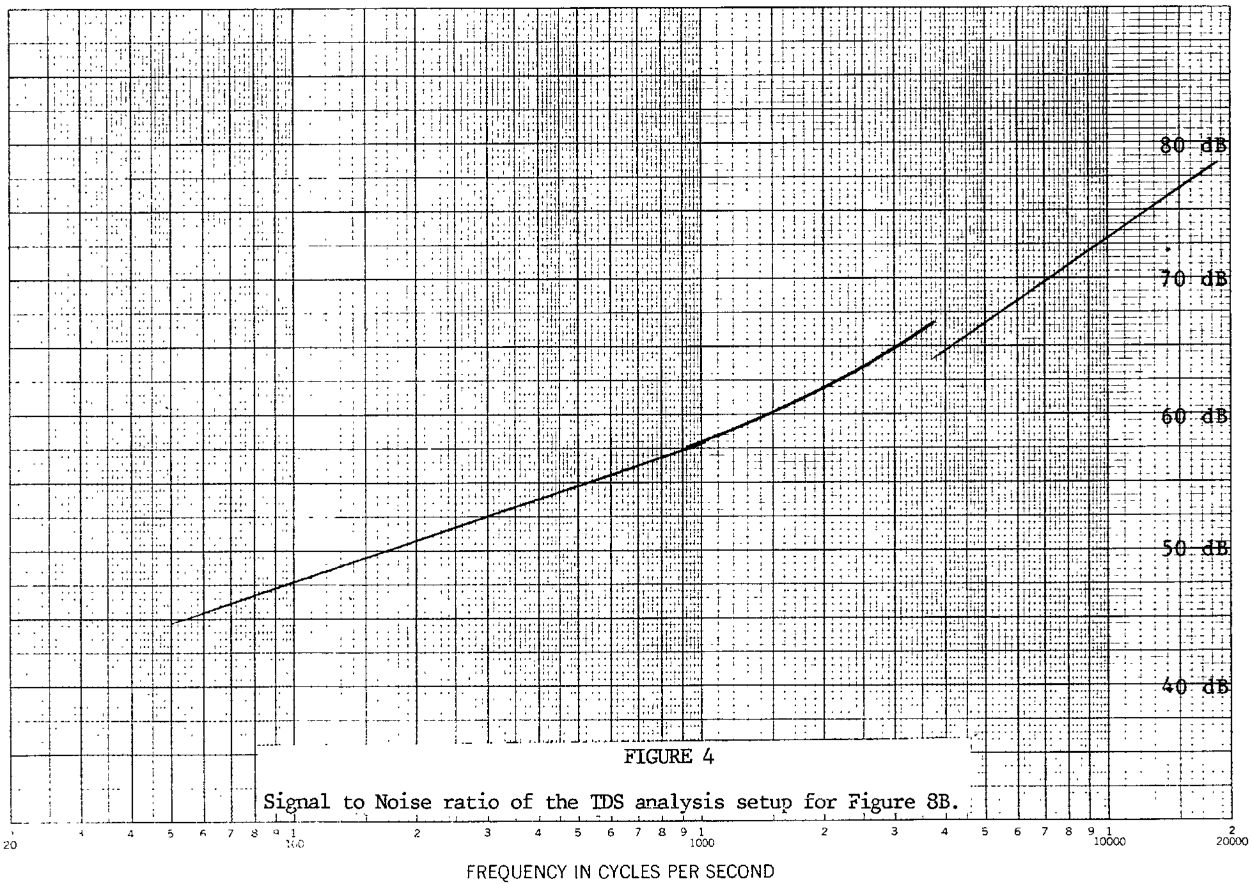


FIGURE 4

Signal to Noise ratio of the TDS analysis setup for Figure 8B.

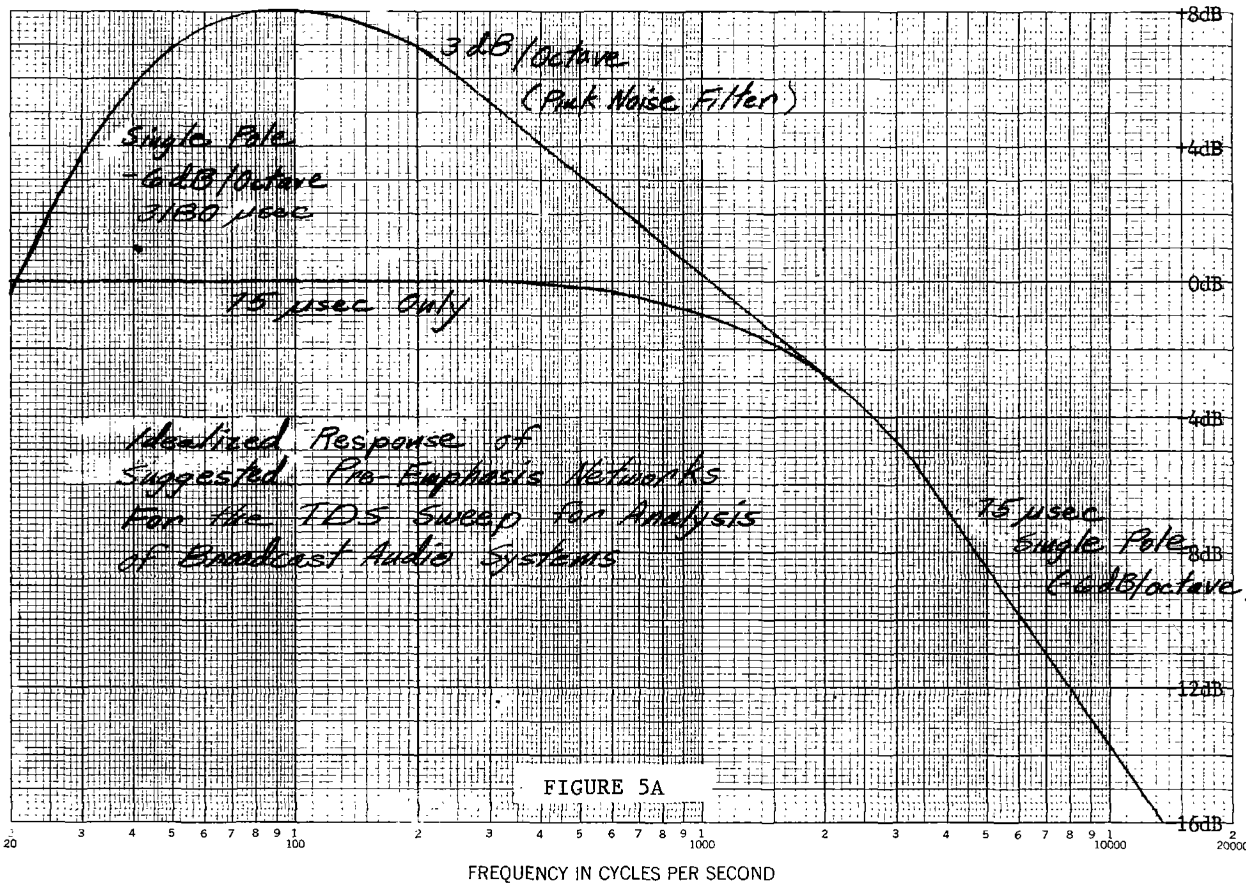

3 dB/Octave
(Pink Noise Filter)
Single Pole
-6 dB/Octave
3180 µsec
75 µsec Only
Idealized Response of
Suggested Pre-Emphasis Networks
For the TDS Sweep for Analysis
of Broadcast Audio Systems
75 µsec
Single Pole
(-6dB/octave)
+8dB
+4dB
0dB
-4dB
-8dB
-12dB
-16dB
20
100
1000
10000
20000
FREQUENCY IN CYCLES PER SECOND

FIGURE 5A

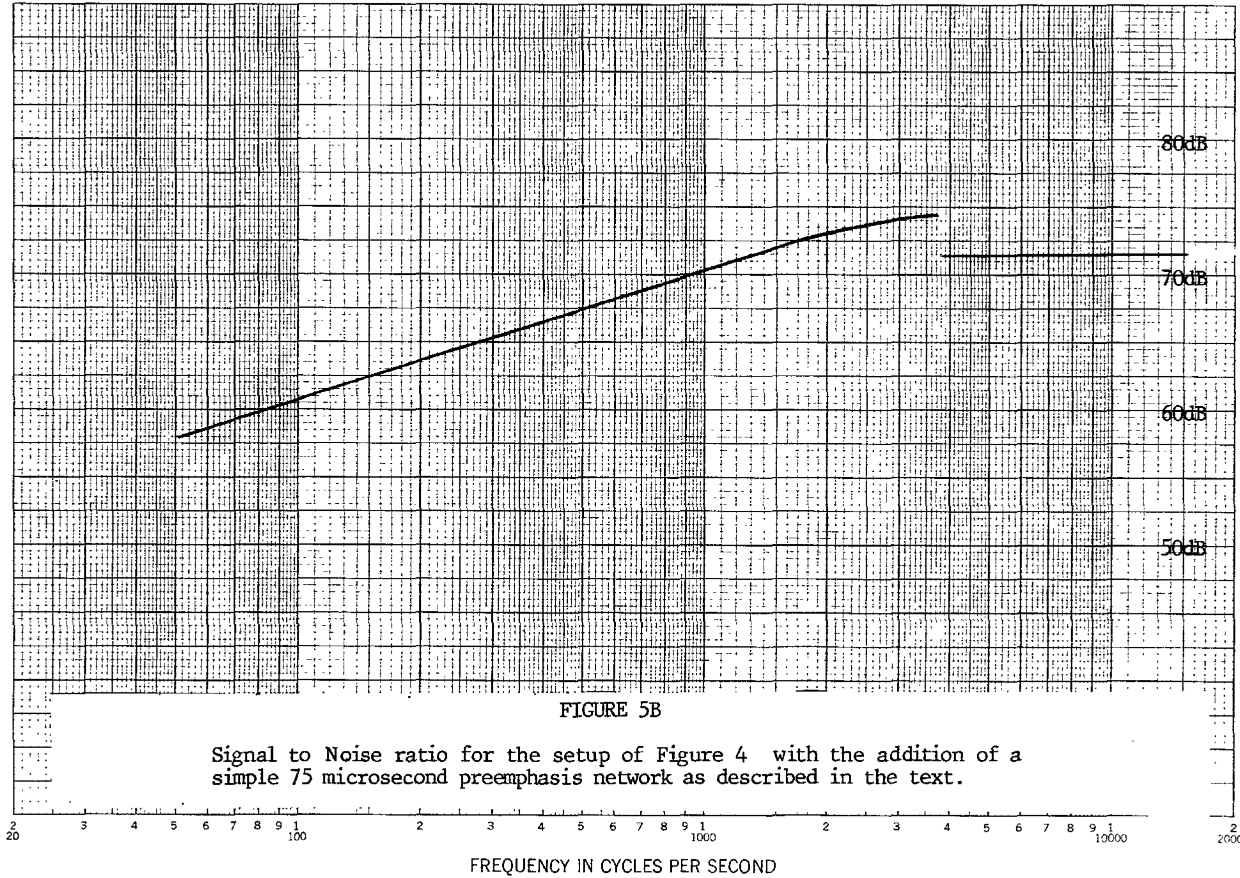


FIGURE 5B

Signal to Noise ratio for the setup of Figure 4 with the addition of a simple 75 microsecond preemphasis network as described in the text.

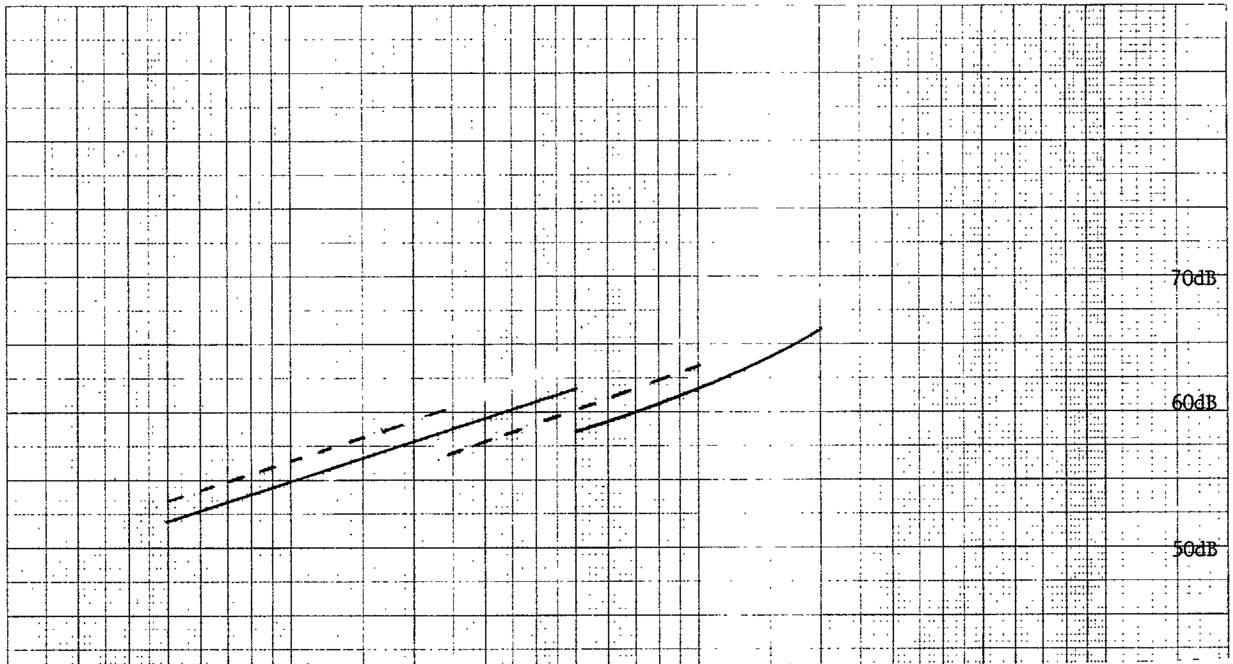
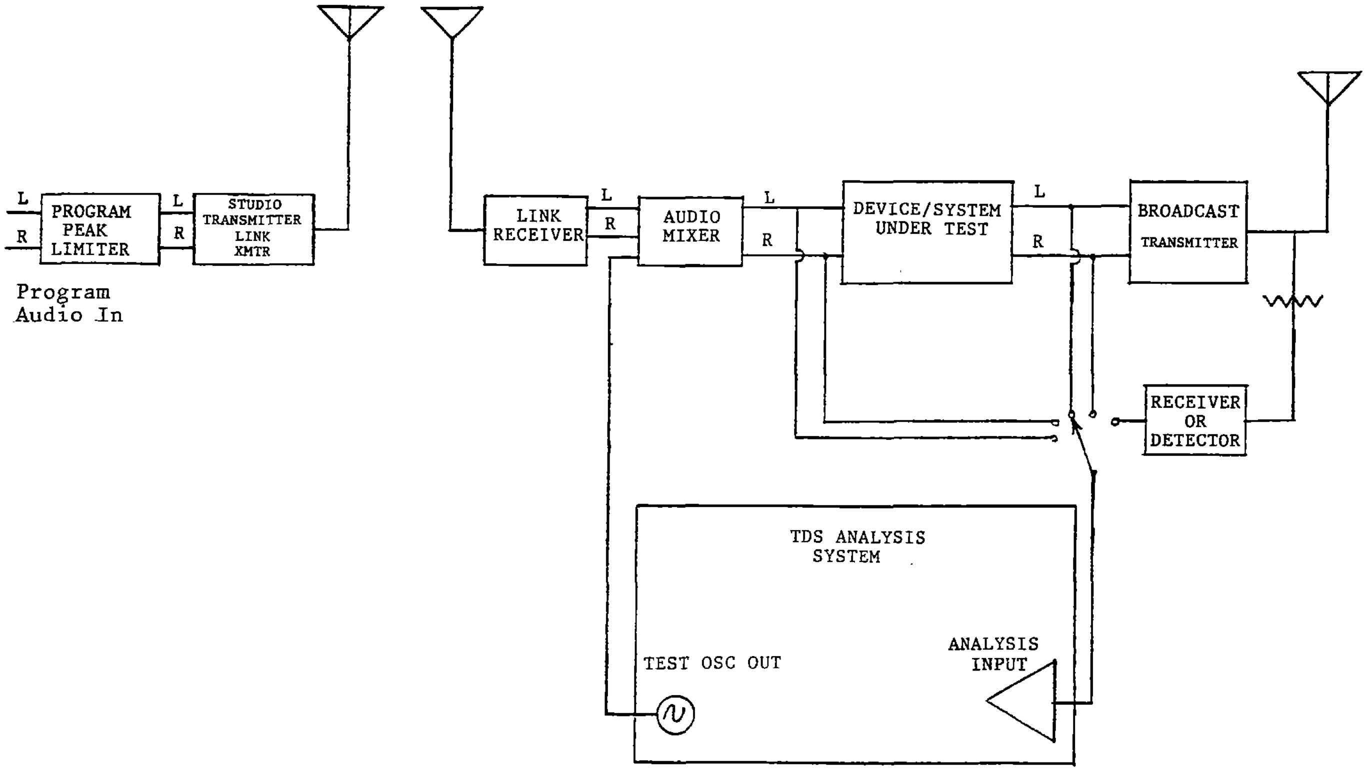


FIGURE 5C

Signal to noise ratio for the TDS analysis setup of Figure 8A. The dashed curve is the recommended low frequency setup for better resolution and noise.



TEST SETUP AT A TYPICAL BROADCAST FACILITY

FIGURE 6

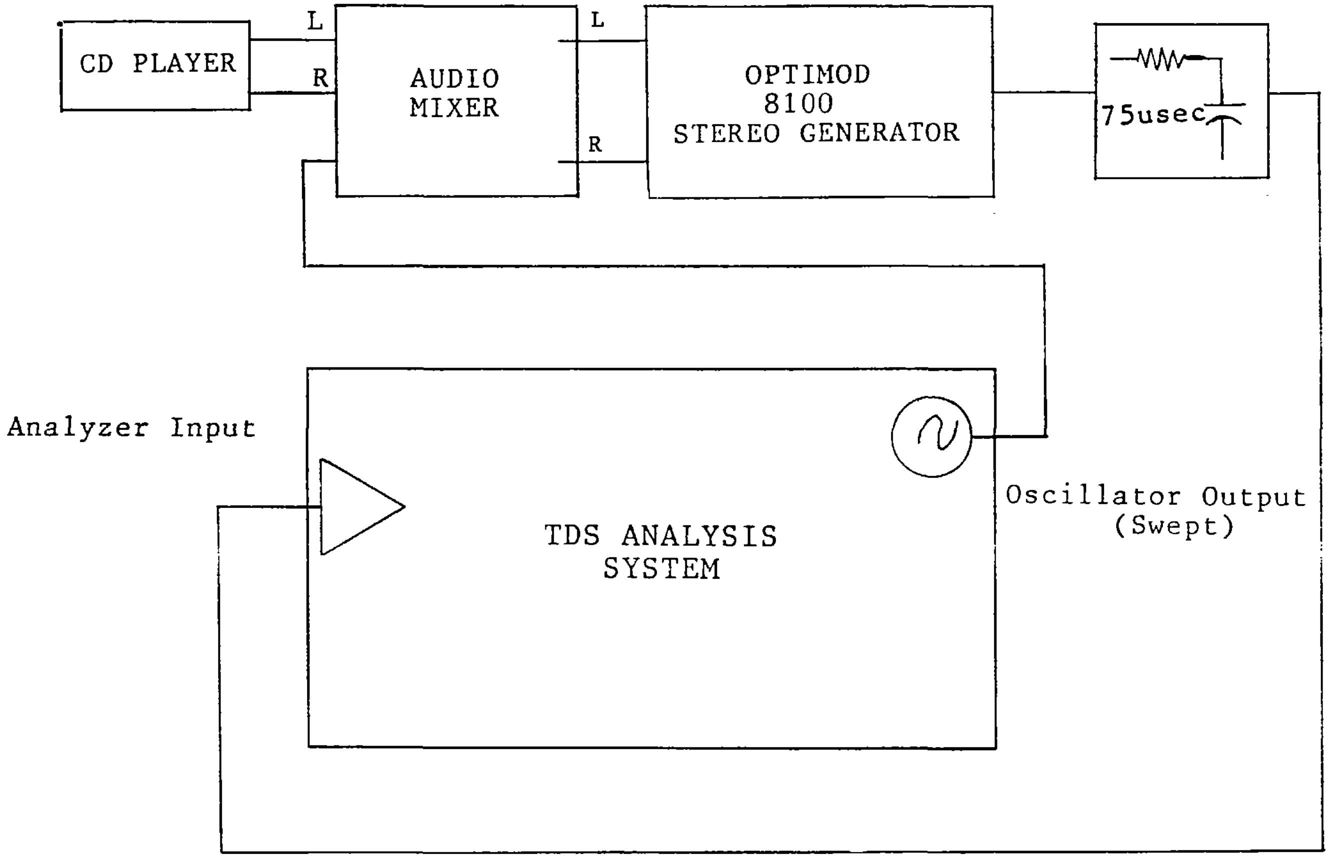
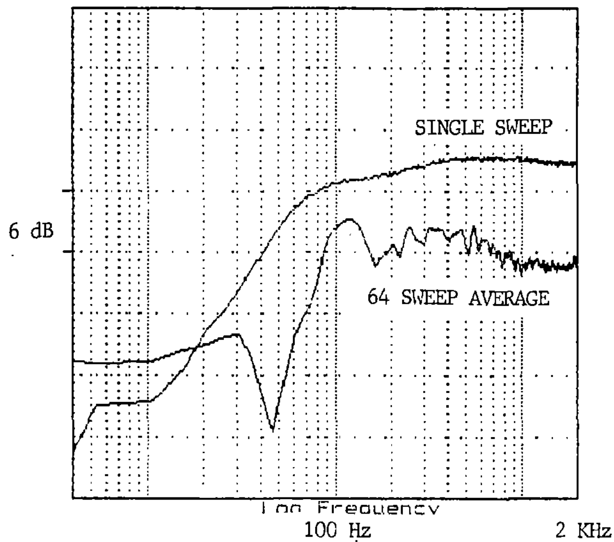

CD PLAYER
L
R
AUDIO
MIXER
L
R
OPTIMOD
8100
STEREO GENERATOR
75usec
Analyzer Input
TDS ANALYSIS
SYSTEM
Oscillator Output
(Swept)

FIGURE 7

Mag. vs Hz (EFC) of OPTIMOD 8100 #2
By JIM BROWN
On 10/3/85
At S. E. A. LAB



Vertical: 6dB/div with base of display at -52.8dB
0dB is located at .775 Volt

Horizontal: Auto 4.00Hz to 2000.24Hz
Log freq axis (2.7decades)

Resolution: 25 milliseconds & 39.97Hz

Time of test: 371 microseconds, 4.2318E-01 Feet

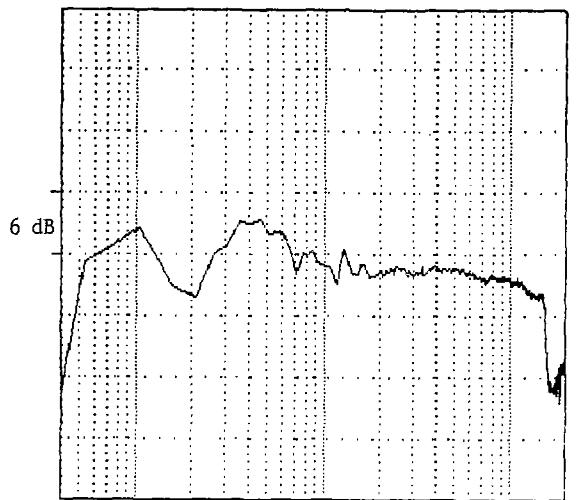
Sweep Rate & Bandwidth: 199.85Hz/Sec & 5.00Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 8A

Amplitude response of the processor at low frequencies, with and without signal. The time of the measurement is the 371 microseconds found to be the signal delay for the processor at high and medium frequencies.

Mag. vs Hz (EFC) of OPTIMOD 8100 #2
By JIM BROWN
On 10/3/85
At S. E. A. LAB



Vertical: 6dB/div with base of display at -52.8dB
0dB is located at .775 Volt

Horizontal: Auto 40.00Hz to 19998.10Hz
Log freq axis (2.7decades)

Resolution: 10 milliseconds & 98.87Hz

Time of test: 371 microseconds, 4.2318E-01 Feet

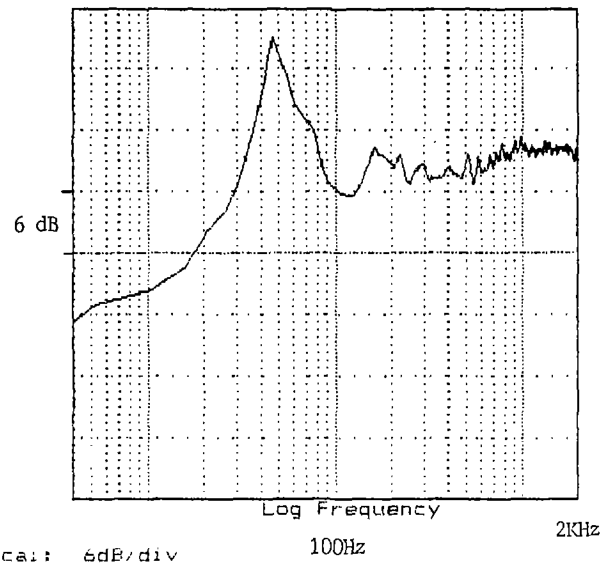
Sweep Rate & Bandwidth: 1977.45Hz/Sec & 20.00Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 8B

Magnitude response of the processor with signal, as measured by the 64 sweep complex average.

Mag. vs Hz (EFC) of OPTIMOD 8100 #2
By JIM BROWN
On 10/3/85
At S. E. A. LAB



Vertical: 6dB/div
Differenced data

Horizontal: Auto 4.00Hz to 2000.24Hz
Log freq axis (2.7decades)

Resolution: 25 milliseconds & 39.97Hz

Time of test: 371 microseconds, 4.2318E-01 Feet

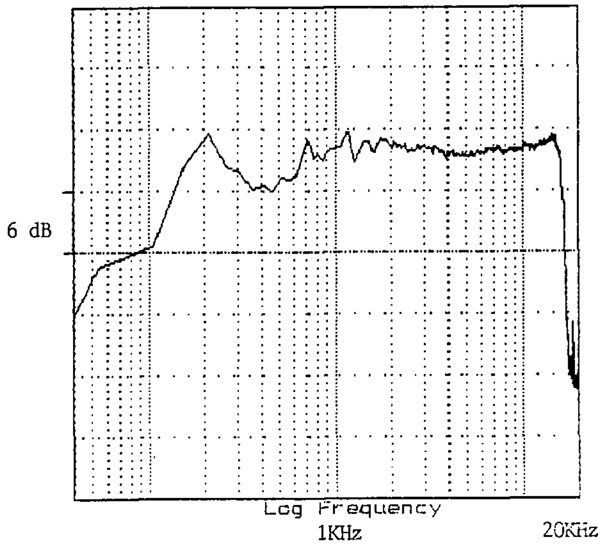
Sweep Rate & Bandwidth: 199.85Hz/Sec & 5.00Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 9A

The gain reduction of the processor at low frequencies. The single sweep without signal is differenced to the 64 sweep complex average.

Mag. vs Hz (EFC) of OPTIMOD 8100 #2
By JIM BROWN
On 10/3/85
At S. E. A. LAB



Vertical: 6dB/div
Differenced data

Horizontal: Auto 40.00Hz to 19998.10Hz
Log freq axis (2.7decades)

Resolution: 1.1538E+01 Feet & 98.97Hz

Time of test: 371 microseconds, 4.2318E-01 Feet

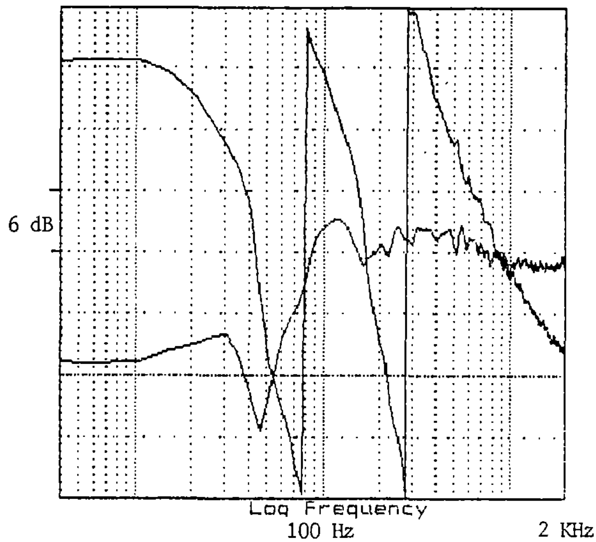
Sweep Rate & Bandwidth: 1977.45Hz/Sec & 20.00Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 9B

Gain Reduction for the processor. The single sweep without signal is referenced to the 64-sweep complex average.

Mag. vs Hz (EFC) of OPTIMOD 8100 #2
By JIM BROWN
On 10/3/85
At S. E. A. LAB



Vertical: 6dB/div with base of display at -52.8dB
0dB is located at .775 Volt

Horizontal: Auto 4.00Hz to 2000.24Hz
Log freq axis (2.7decades)

Resolution: 25 milliseconds & 39.97Hz

Time of test: 371 microseconds, 4.2318E-01 Feet

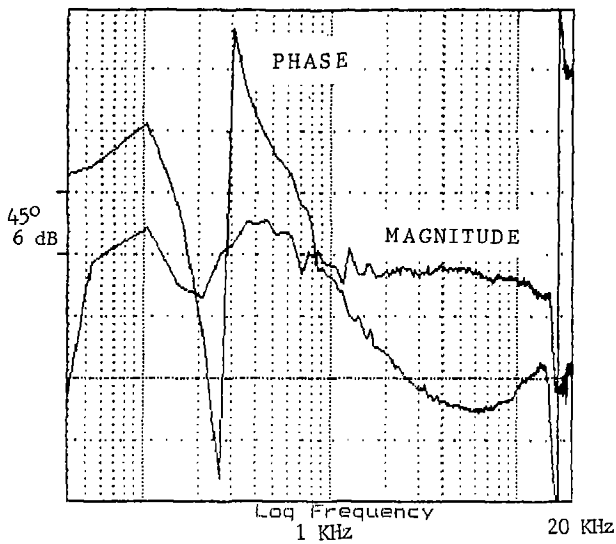
Sweep Rate & Bandwidth: 199.85Hz/Sec & 5.00Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 10A

Amplitude and phase response of the processor at low frequencies. The apparent oscillation of the phase data at low frequencies is actually very high phase shift, plotted at multiples of the 360 degrees which is displayed. This is a 64 sweep complex average, with program.

Mag. vs Hz (EFC) of OPTIMOD 8100 #2
By JIM BROWN
On 10/3/85
At S. E. A. LAB



Vertical: 6dB/div with base of display at -52.8dB
0dB is located at .775 Volt

Horizontal: Auto 40.00Hz to 19998.10Hz
Log freq axis (2.7decades)

Resolution: 1.1538E+01 Feet & 98.87Hz

Time of test: 371 microseconds, 4.2318E-01 Feet

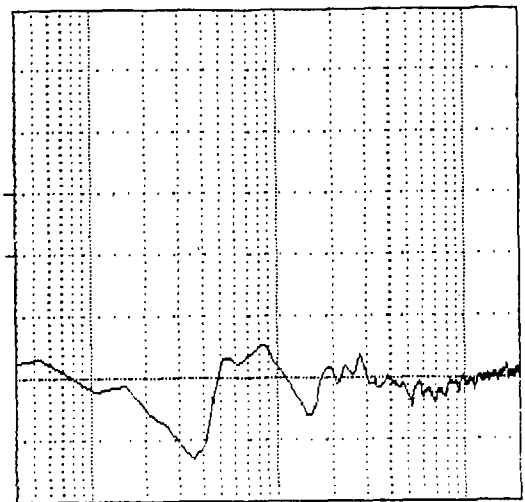
Sweep Rate & Bandwidth: 1977.45Hz/Sec & 20.00Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 10B

Magnitude and phase response of the processor with signal as measured by the 64 sweep complex average. The apparent oscillation of the data at high and low frequencies is really the phase being plotted at multiples of the 360 degrees which are displayed on axis.

Phase vs Hz (EFC) of OPTIMOD 8100 #2
By JIM BROWN
On 10/3/85
At S. E. A. LAB



Vertical: 45 degrees/div.
0 degrees is at the dashed horizontal line.

Differenced data

Horizontal: Auto 4.00Hz to 2000.24Hz
Log freq axis (2.7decades)

Resolution: 25 milliseconds & 39.97Hz

Time of test: 371 microseconds, 4.2318E-01 Feet

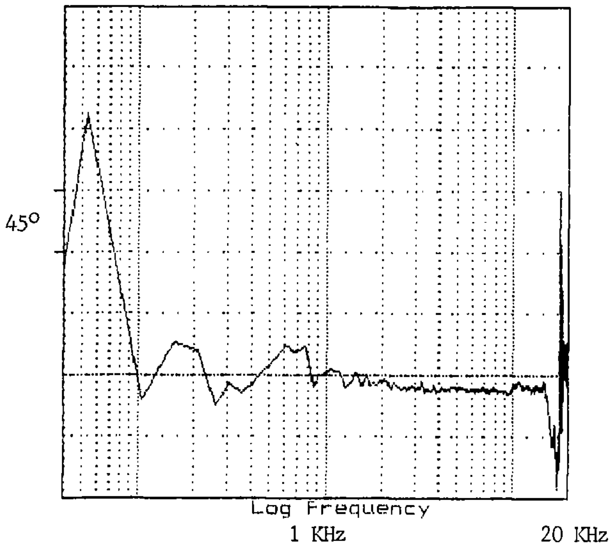
Sweep Rate & Bandwidth: 199.85Hz/Sec & 5.00Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 11A

The difference between the phase response of the processor with and without signal. The complex average is differenced to the single sweep without signal. The measurement is subject to considerable noise below 100 Hz, partially because of the signal delay at those frequencies and partly due to the reduced noise immunity.

Phase vs Hz (EFC) of OPTIMOD 8100 #2
By JIM BROWN
On 10/3/85
At S. E. A. LAB



Vertical: 45 degrees/div.
0 degrees is at the dashed horizontal line.

Differenced data

Horizontal: Auto 40.00Hz to 19998.10Hz
Log freq axis (2.7decades)

Resolution: 10 milliseconds & 98.87Hz

Time of test: 371 microseconds, 4.2318E-01 Feet

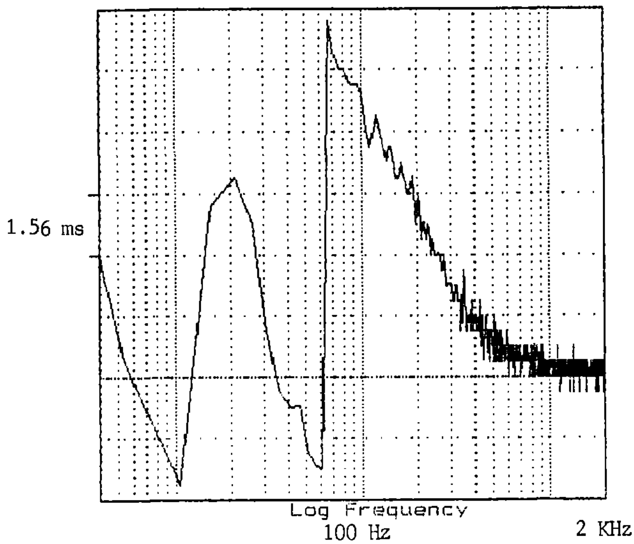
Sweep Rate & Bandwidth: 1977.45Hz/Sec & 20.00Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 11B

The difference between the phase response of the single sweep without signal and the 64 sweep complex average.

Group Delay of OPTIMOD 8100 #2
By JIM BROWN
On 10/3/85
At S. E. A. LAB



Vertical: Delay in microseconds, 1562 uSec/div
zero delay is at the dashed horizontal line.

Horizontal: Auto 4.00Hz to 2000.24Hz
Log freq axis (2.7decades)

Resolution: 25 milliseconds & 39.97Hz

Time of test: 371 microseconds, 4.2318E-01 Feet

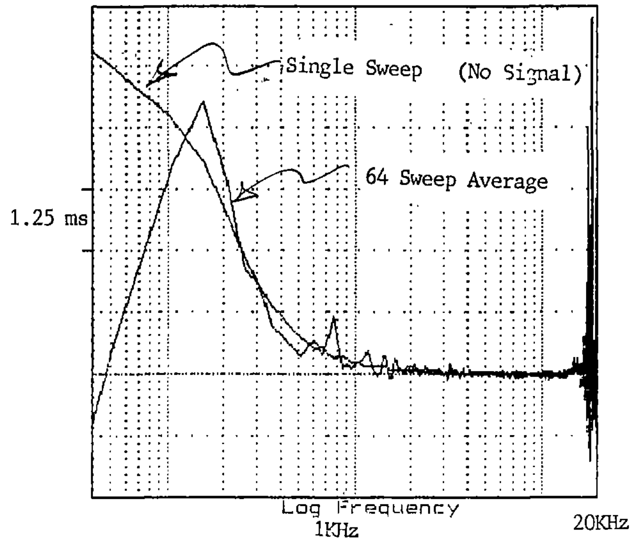
Sweep Rate & Bandwidth: 199.85Hz/Sec & 5.00Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 12A

Group delay for the processor from the low frequency sweep. The apparent oscillation of the data at low frequencies is really the computer plotting that data off axis above the display. Eight divisions of delay should thus be added to the displayed data in the range between 30 Hz and 60 Hz. Note that the frequency resolution of this data is 40 Hz, so that data below about 30 Hz has little significance.

Group Delay of OPTIMOD 8100 #2
By JIM BROWN
On 10/3/85
At S. E. A. LAB



Vertical: Delay in microseconds, 1250 uSec/div
zero delay is at the dashed horizontal line.

Horizontal: Auto 40.00Hz to 19998.10Hz
Log freq axis (2.7decades)

Resolution: 10 milliseconds & 98.87Hz

Time of test: 371 microseconds, 4.2318E-01 Feet

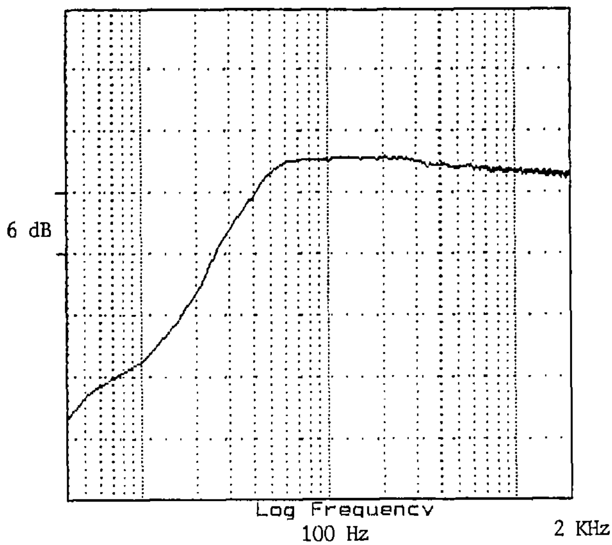
Sweep Rate & Bandwidth: 1977.45Hz/Sec & 20.00Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 12B

Group delay for the processor from the high frequency sweep. The time reference for this data is the 371 microseconds found to be the signal delay through the device at high and medium audio frequencies.

Mag. vs Hz (EFC) of OPTIMOD 8100 #2
By JIM BROWN
On 10/3/85
At S. E. A. LAB



Vertical: 6dB/div with base of display at -52.8dB
0dB is located at .775 Volt

Horizontal: Auto 4.00Hz to 2000.24Hz
Log freq axis (2.7decades)

Resolution: 25 milliseconds & 39.97Hz

Time of test: 8371 microseconds, 9.5492E+00 Feet

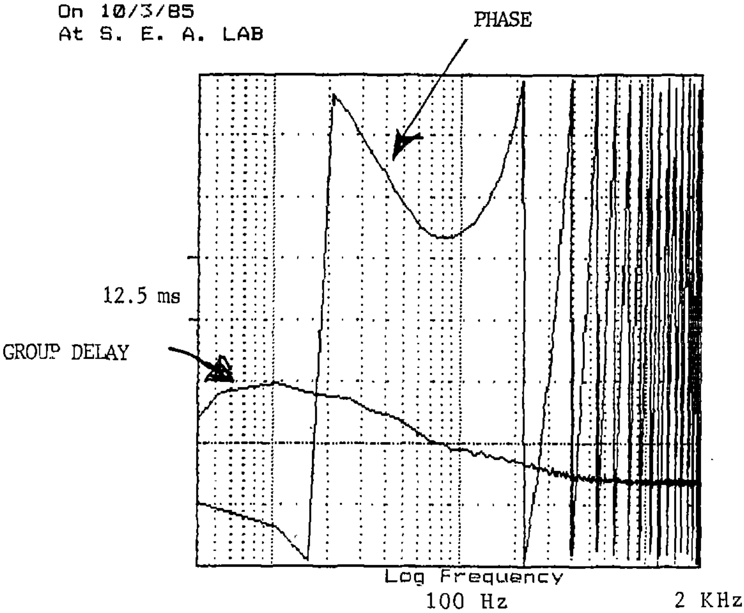
Sweep Rate & Bandwidth: 199.85Hz/Sec & 5.00Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 13A

Amplitude response of the processor with the time of measurement changed to include the delayed low frequency response. Note that the mid-frequencies are starting to move out of the time window. This data is from a single sweep with no signal.

Group Delay of OPTIMOD 8100 #2
By JIM BROWN
On 10/3/85
At S. E. A. LAB



Vertical: Delay in microseconds, % 12499 uSec/div
zero delay is at the dashed horizontal line.

Horizontal: Auto 4.00Hz to 2000.24Hz
Log freq axis (2.7decades)

Resolution: 25 milliseconds 39.97Hz

Time of test: 8371 microseconds, 9.5492E+00 Feet

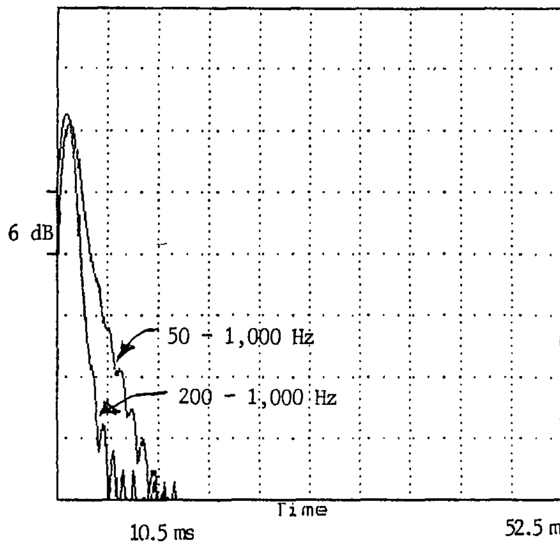
Sweep Rate & Bandwidth: 199.85Hz/Sec & 5.00Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 13B

The phase response, and its negative first derivative, the group delay for the measurement of Figure 13A. Note that this measurement shows the low frequencies are delayed considerably more than the 8 msec time chosen for the measurement, so that some low frequency data is still falling outside the analysis time window. The amplitude response at those low frequencies is thus higher than indicated, but with a significant time offset from the major part of the program audio.

Amplitude ETC of OPTIMOD 8100 #2
By JIM BROWN
On 10/3/85
At S. E. A. LAB



Vertical: 6dB/div with base of display at -55.8dB
0dB is located at .775 Volt

Horizontal: 0 microseconds or 0 Feet to
52630 microseconds or 60.0381 Feet
scale: 1.6415E+01 Feet/inch or 6.4626E+00 Feet/cm.
14389 microseconds/inch or 5665 microseconds/cm.

Line Spacing: 131.905 microseconds or .150471 Feet
Line Width: 179.391 microseconds or .204641 Feet

Sweep rate: 99.92Hz/Sec

Sweep range: 50.33Hz to 997.98Hz

Window file name: A:HAMMING.W8T

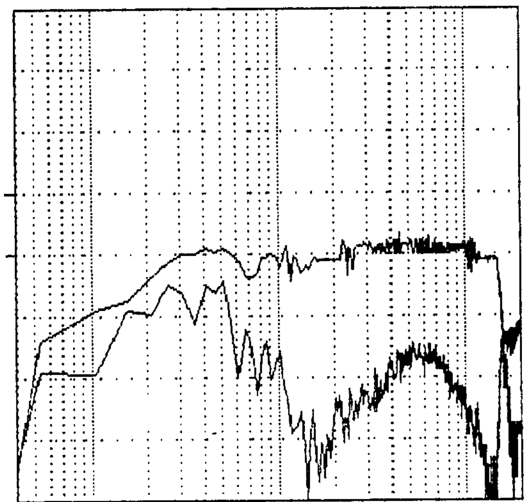
Input configuration: Channel 1 Balanced
with 24dB of input gain & 15dB of IF gain.

FIGURE 13C

Energy Time Curves for the processor taken without program material, but at the same level as during program tests. Note that the sweep which covers only the region above 200 Hz is more compact, indicating the lower frequency energy is arriving later. Since the TDS sweep is linear, the Energy-Time Curve emphasizes the higher frequency octaves. This measurement confirms the findings made using the data of curves 13A, 13B, 12A, and 12B.

OPTIMOD #2

By JIM BROWN
On 10/8/85
At MADISON BROADCAST CLINIC



Vertical: 6dB/div with base of display at -58.8dB
0dB is located at .775 Volt

Horizontal: Auto 40.00Hz to 19998.10Hz
Log freq axis (2.7decades)

Resolution: 10 milliseconds at & 98.87Hz

Time of test: 371 microseconds, 4.2318E-01 Feet

Sweep Rate & Bandwidth: 1977.45Hz/Sec & 20.00Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

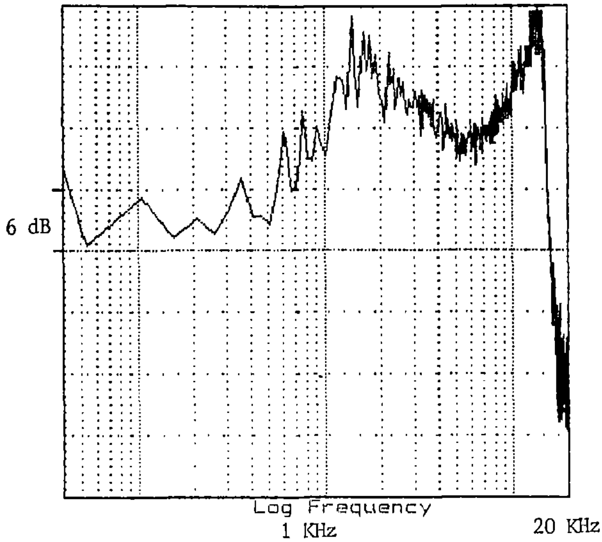
Remarks:

FIGURE 14A

Amplitude response with and without signal, with a much higher drive signal to the processor. The processor is doing much higher compression. The sweep with signal is a 64 sweep complex average.

OPTIMOD #2

By JIM BROWN
On 10/8/85
At MADISON BROADCAST CLINIC



Vertical: 6dB/div
Differenced data

Horizontal: Auto 40.00Hz to 19998.10Hz
Log freq axis (2.7decades)

Resolution: 10 milliseconds & 98.87Hz

Time of test: 371 microseconds, 4.2318E-01 Feet

Sweep Rate & Bandwidth: 1977.45Hz/Sec & 20.00Hz

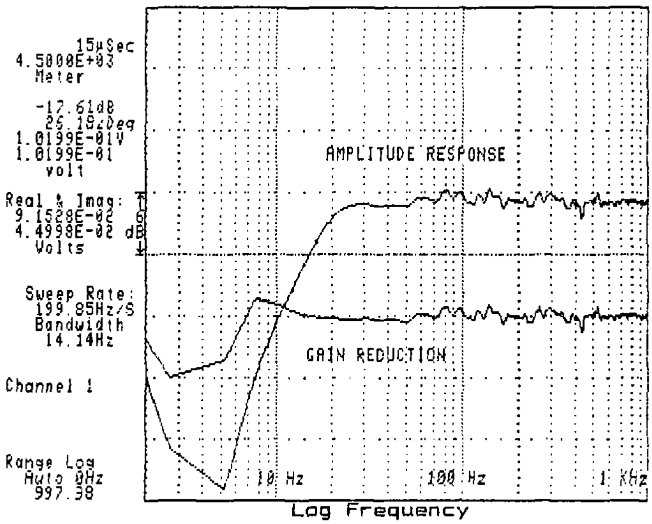
Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

Remarks:

FIGURE 14B

The difference between the two sets of data of Figure 14A, which represents the gain reduction of the processor.

Mag. vs Hz (EFC) of AUDIO & DESIGN VOCAL STRESSOR
By JIM BROWN
On 10/19/85
At S. E. A. LAB



Vertical: 6dB/div with base of display at -46.8dB
0dB is located at .775 volt

Horizontal: Auto 2.00Hz to 997.98Hz
Log freq axis (2.7decades)

Resolution: 70 milliseconds & 14.14Hz

Time of test: 15 microseconds, 4.5000E+03 Meter

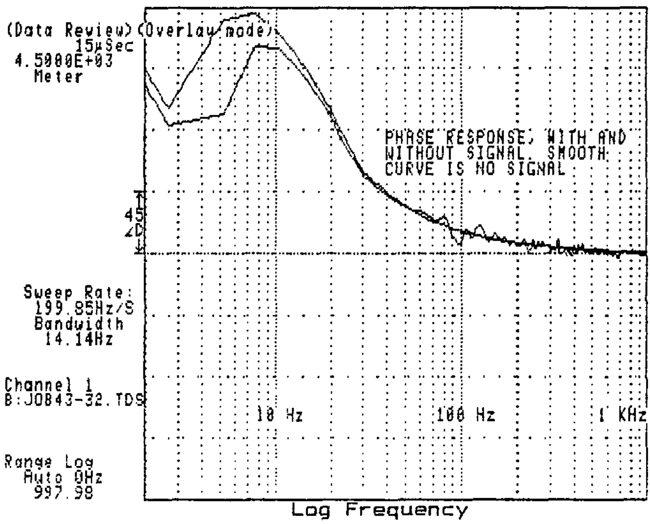
Sweep Rate & Bandwidth: 199.85Hz/Sec & 14.14Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 15A

The low frequency response of a typical compressor set for an indicated 6 dB of compression. This is a 64 sweep complex average. The sweep level was set at a level which was much more audible than for the Optimod, but was well below the average program level.

Phase vs Hz (EFC) of AUDIO & DESIGN VOCAL STRESSOR
By JIM BROWN
On 10/19/85
At S. E. A. LAB



Vertical: 45 degrees/div.
0 degrees is at the dashed horizontal line.

Horizontal: Auto 2.00Hz to 997.98Hz
Log freq axis (2.7decades)

Resolution: 70 milliseconds & 14.14Hz

Time of test: 15 microseconds, 4.5000E+03 Meter

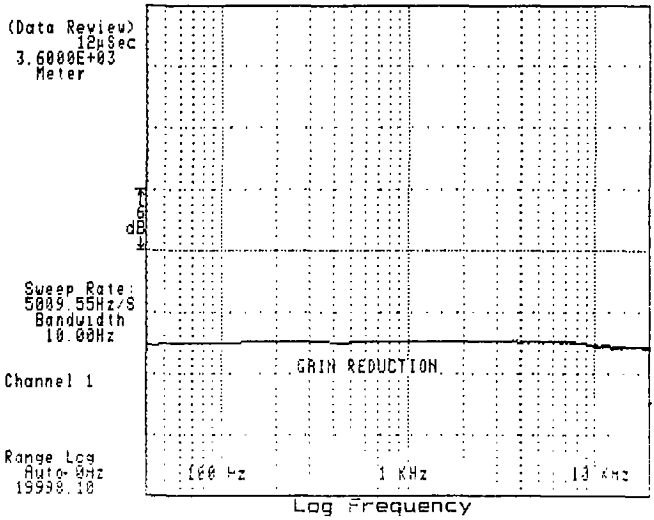
Sweep Rate & Bandwidth: 199.85Hz/Sec & 14.14Hz

Input configuration: Channel 1 Balanced
with 0dB of input gain & 12dB of IF gain.

FIGURE 15B

The phase response of the compressor of Figure 15A. The two curves show the measured phase response, with and without signal.

Mag. vs Hz (EFC) of APHEX COMPELLOR
By JIM BROWN
On 10/19/85
At S. E. A. LAB



Vertical: 6dB/div
Differenced data

Horizontal: Auto 40.00Hz to 19998.10Hz
Log freq axis (2.7decades)

Resolution: 2 milliseconds & 500.96Hz

Time of test: 12 microseconds, 3.6000E+03 Meter

Sweep Rate & Bandwidth: 5009.55Hz/Sec & 10.00Hz

Input configuration: Channel 1 Inverting
with 0dB of input gain & 12dB of IF gain.

FIGURE 16

Measured gain reduction at medium and high frequencies for an integrated compressor, limiter, and AGC. The sweep signal was audible, but well below the average program level.