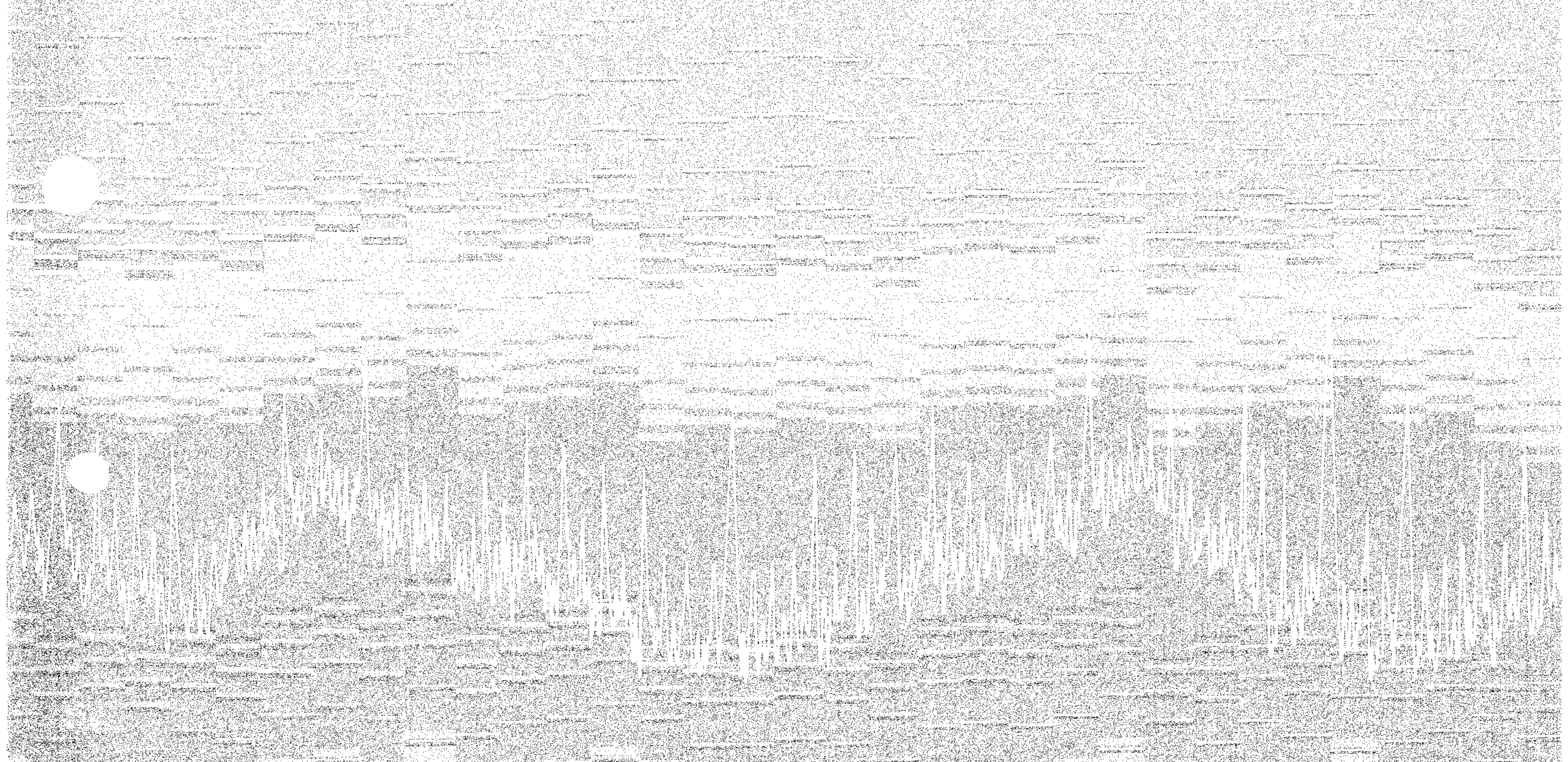
AN192

Using a Narrow Band Analyzer for Characterizing Audio Products



APPLICATION NOTE 192

HEWLETT PACKARD

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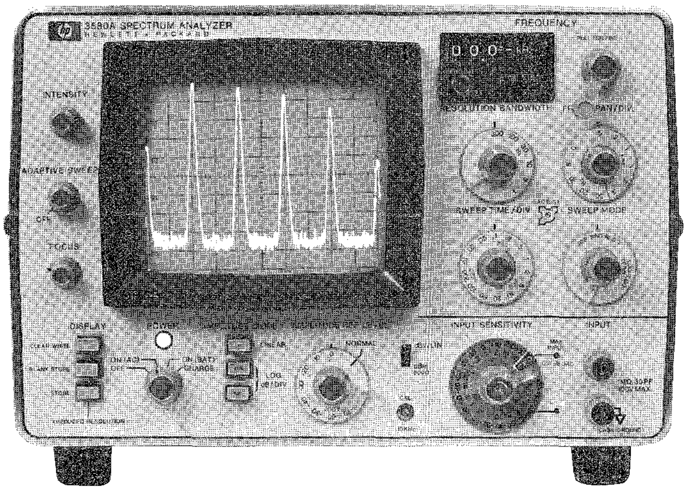
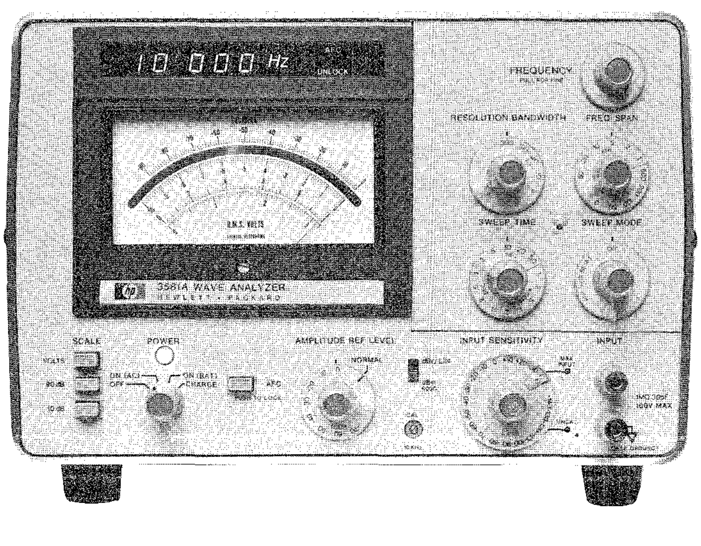
15 OCTOBER 1975

Introduction

Audio entertainment products have made tremendous advances since the days of the "Victrola". Continuous technical advances in components and circuit design have taken us closer to "concert hall" sound quality in the home. The consumer has come to expect high quality audio products and is continuously searching for sound reproduction systems with greater fidelity. The quest for fidelity will go on because for each improvement in fidelity, the ear is still capable of hearing much more. It is no longer possible to market a product which merely reproduces sound. The product must meet rigid specifications of frequency response, signal to noise ratio, channel separation, low distortion, etc., as well as be attractive and reliable. Subjective judgment can, in many cases, be an excellent means of quality classification and will always influence the final decision of the buyer. However, when questions are raised concerning development, production testing, quality assurance, and documentation of a product's capabilities, objective measurements must be made. There is a variety of methods and equipment available to aid in these measurements. A simple measurement system might consist of an oscillator, ac voltmeter, and notch filter. This system is uncomplicated and inexpensive. However, it is limited in its usefulness since numerous discrete frequency measurements must be made to insure adequate testing. Using this system is tedious, time consuming, and does not allow in-depth signal analysis. At the other extreme are the real-time analyzers. These instruments allow fast measurements and are quite useful for signal analysis due to their ability to measure entire bands of frequencies simultaneously. They are, however, somewhat difficult to use and are quite expensive.

The HP Model 3580A Spectrum Analyzer and Model 3581A Wave Analyzer combine some of the better features of both of these measurement systems. They are less expensive and easier to use than real-time analyzers while still providing detailed signal analysis for greater measurement accuracy and to aid in problem solving. Since they can make swept response measurements, they are faster than the simple oscillator, voltmeter system.





Applications

Distortion

When a device is driven beyond its linear range or through areas of discontinuity, signal distortion occurs, resulting in the appearance of additional frequencies at the output which are not present at the input. (See Figure 1, Page 3). In extreme cases, it is possible to identify distortion by merely listening. Third order distortion becomes audible at about 1.25%. Second order distortion becomes audible at 5%. In general, systems with higher frequency response need lower distortion levels to be acceptable. More commonly, however, distortion is not immediately obvious. Distortion might appear as a difference in tonal quality when comparing two systems. For these cases, a more sophisticated technique is needed to identify and measure distortion. The HP Model 3580A Spectrum Analyzer and Model 3581A Wave Analyzer are quite useful for exposing distortion due to their ability to separate frequency components of the distorted signal for analysis.

Basically, distortion is characterized in two ways, harmonic distortion and intermodulation distortion. While there is no correlation between measurement values when using these two methods, each produces quantitative results of a product's quality. The motive behind these different measurements is simplicity. Rather than specify the separate amplitudes of each harmonic, they are mathematically manipulated to give a single number which is much easier to understand.

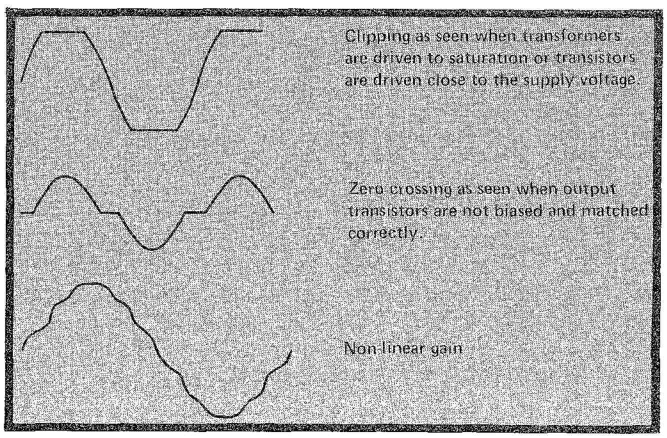


Figure 1. For sine wave inputs, there are three ways that give rise to distortion. The different distortion measurements are designed to expose these effects in different ways.

Harmonic Distortion

Harmonic distortion is a measure of the individual harmonic amplitudes, relative to the amplitude of the fundamental frequency. By this definition, the amplitudes of all harmonic frequencies should be used for the calculation. But, harmonics 20 dB below the highest harmonic can be ignored because their contribution is negligible. In actual practice, harmonics greater than the third are relatively small and can usually be ignored. THD is defined by the following equation:

$$\text{THD\%} = 100 \, \frac{\sqrt{A_1^2 + A_2^2 + A_3^2 + \ldots}}{A_f}$$

where A_h is the amplitude of the individual harmonics and A_f is the amplitude of the fundamental. To help you calculate distortion from the analyzer display, Figure 2 presents three algorithms using popular HP calculators.

The Model 3580A or Model 3581A can be tuned to measure the amplitude of individual harmonics as well as the fundamental.

To measure harmonic distortion, the device under test is usually driven at its maximum ratings to produce "worst case" distortion. The device is driven by a low distortion oscillator which should have a distortion figure better than the unit to be tested. Distortion of the device can be measured even if the oscillator has almost equal distortion but cumbersome corrections have to be made. Another alternative is to use a fixed low pass filter to remove distortion products from the oscillator. For amplifiers, the output of the unit being tested is connected to the input of the Model 3580A or Model 3581A. (See Figure 3)

Overall harmonic distortion of tape recorders can be measured by recording the test signal on tape and then measuring the output of the recorded signal as the tape is played back. On "three head" recorders, this can be done in one pass. (See Figure 4.) Record and playback distortion can be separated by using a standard test tape to measure playback distortion. Distortion of the record amplifier can then be found by finding the difference between the overall distortion and playback distortion measurements.

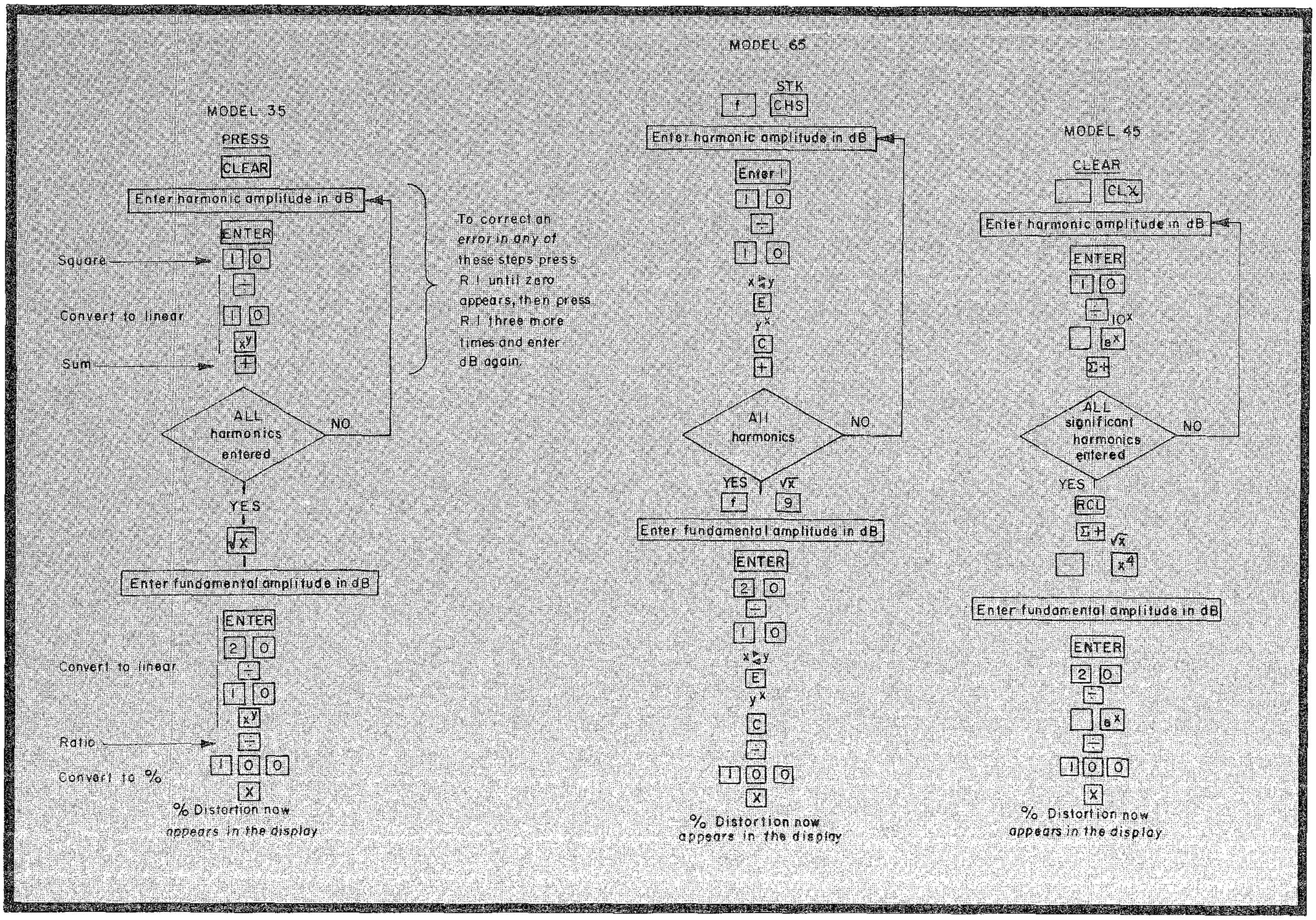


Figure 2. Three harmonic distortion algorithims using HP calculators.

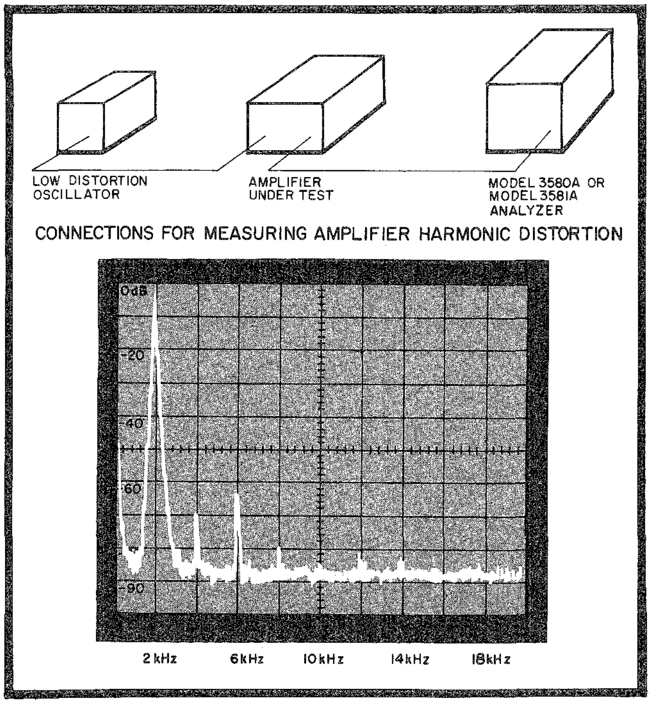


Figure 3. Measurement results show total harmonic distortion to be greater than 63 dB below the fundamental.

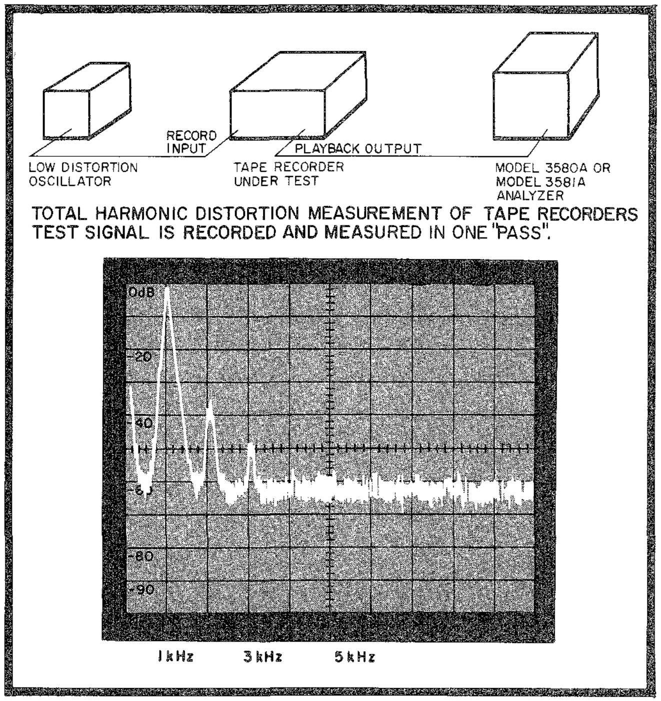


Figure 4. Total harmonic distortion of tape recorder shows THD to be approximately 2%.

Speaker distortion can be measured with the addition of a standard amplifier and microphone as shown in Figure 5

Because of the wide dynamic range of the models 3580A and 3581A, distortion as low as −80 dB (0.01%) can easily be measured. The dynamic range of the instruments is responsible for this limitation. The capability can be extended to −100 dB or 0.001% if the fundamental alone can be attenuated by at least 20 dB. This can be done for fixed frequencies with a notch or high pass filter. These filters must be built with passive components to guarantee that the dynamic range and noise of the filter are no worse than those of the instrument. The highest frequency which can be measured is limited by the frequency of the greatest harmonic of interest. While the instrument is capable of measuring up to 50 kHz, harmonics over 20 kHz are typically not of interest because they are beyond the range of hearing.

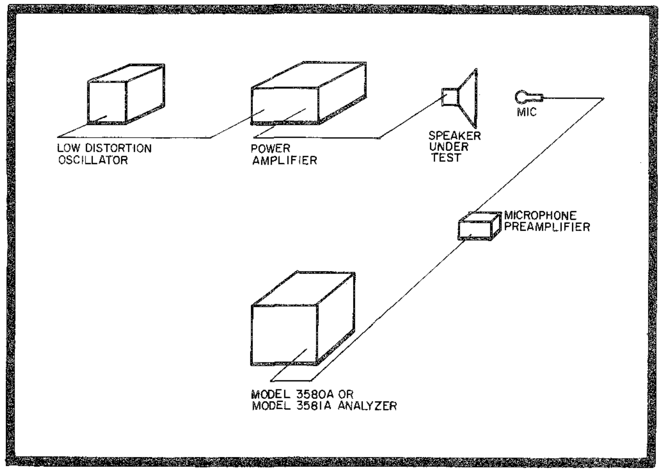


Figure 5. Measuring speaker distortion.

Intermodulation Distortion

*Intermodulation distortion is still not as widely used as an index of audio system performance as it might be. The use of the heterodyne spectrum analyzer makes it possible to conduct several different IM tests on the same test unit by simply varying the signal input frequencies and relative levels. Use of the conventional IM distortion meters limits the test performed to that for which the individual IM analyzer was designed.

There are other tests that could also be performed using the spectrum analyzer and appropriate signal sources. The recently infamous Transient Intermodulation Distortion can be investigated quickly over a range of input levels. The tracing simulation devices used for reduction of geometrically induced distortion in lacquer master discs can be checked by cutting a 1 kHz CW signal at several groove velocities and checking the harmonic content of the reproduced signal.

*Reprinted with permission from "Recording Engineer/Producer," June 1975, Vol. 6, No. 3, pages 41-51, entitled "Spectrum Analysis Applied to Audio Systems Diagnostics," by Peter Butt.

The intermodulation method of measuring distortion uses a driving signal composed of two sinusoidal signals of different frequencies. This composite signal can be thought of as a modulated carrier. The intermodulation method is useful because the distortion measurement is not affected by the distortion of the signal source. Distortion from the source does not occur at the same frequencies as device distortion so the analyzer can be used to pick out just the device distortion products. There are two generally accepted intermodulation techniques. One is the SMPTE method which uses a low frequency and a relatively high frequency signal (normally 60 Hz and 7 kHz) which are mixed at a four to one amplitude ratio (see illustration). Using this method, distortion is determined from the relative amplitude of the modulation sidebands of the higher frequency signal. For diagnostic purposes it is useful to determine even-and-odd order distortion separately. Second order distortion is determined from the ratio of the sum of the amplitudes of the two spurious frequencies, f_2-f_1 and f_2+f_1, to the amplitude of the high frequency signal, f_2.

$$\text{Second order distortion} = \left[\frac{A_{(f_2-f_1)} + A_{(f_2+f_1)}}{A_{f_2}}\right] \times 100$$

(in percent)

From Figure 6 photograph:

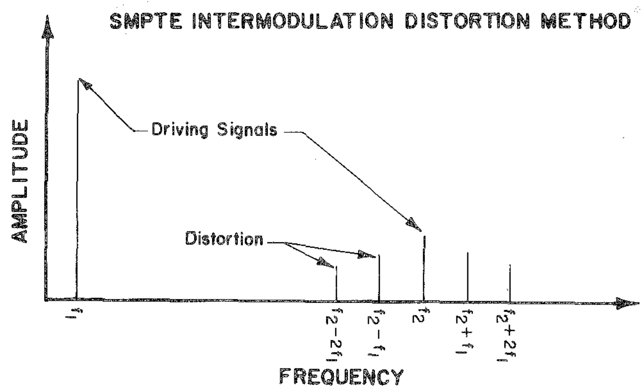
$A\ (f_2-f_1) = -50\ \text{dB} = 3.2\ \text{mv}$

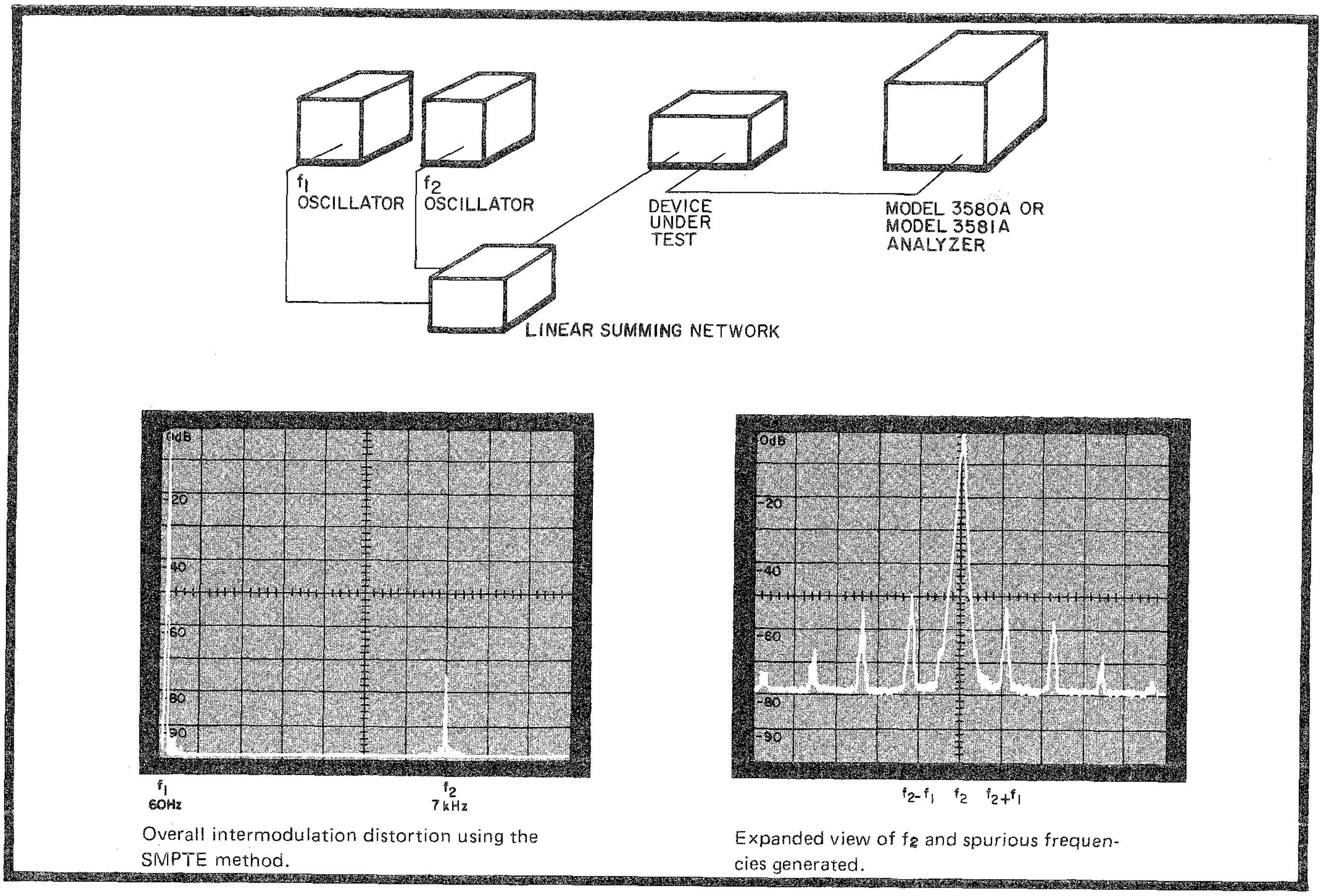
$A\ (f_2+f_1) = -54\ \text{dB} = 2.0\ \text{mv}$

$A \qquad = \quad 0\ \text{dB} = 1\ \text{V}$

$$\underset{\text{even order}}{\text{Distortion}} = \left[\frac{3.2\ \text{mv} + 2.0\ \text{mv}}{1\text{V}}\right] 100 = .52\%$$

Similarly, third order distortion is the ratio of the sum of the amplitudes of the two spurious frequencies, f_2-2f_1, and f_2+2f_1, to the amplitude of f_2





Overall intermodulation distortion using the SMPTE method.

Expanded view of f_2 and spurious frequencies generated.

Figure 6. Test connections and measurement results for SMPTE intermodulation distortion.

$$\text{Third order distortion [in percent]} = \left[\frac{A_{(f_2-2f_1)} + A_{(f_2+2f_1)}}{A_{f_2}}\right] \times 100$$

From Figure 6 photograph:

$A\ (f_2-2f_1) = -54\ \text{dB} = 2.0\ \text{mv}$
$A\ (f_2+2f_1) = -58\ \text{dB} = 1.3\ \text{mv}$
$A_{f_2} = 0\ \text{dB} = 1\ \text{V}$

$$\text{Distortion Odd Order} = \left[\frac{2.0\ \text{mv} + 1.3\ \text{mv}}{1\text{V}}\right]\ 100 = .33\%$$

For general purposes, it is more convenient to express the total distortion as a single percentage. In this case, the total distortion is equal to the square root of the sum of the squares of the second and third order distortion terms.

For this case: $\sqrt{(.52)^2 + (.33)^2} = \underline{.62\%}$

Another method, known as the CCIF method, uses a combination of two relatively high frequency sinusoidal signals of equal amplitude as the driving signal. For audio-frequency distortion measurements, their frequencies are typically within a few hundred hertz of each other. Some of the spurious frequencies generated are low in frequency while some are gathered around the driving frequencies themselves (see illustration). Even-order distortion is expressed as the ratio of the amplitude of the lowest frequency component (f_2-f_1) to the sum of the amplitudes of the driving frequencies.

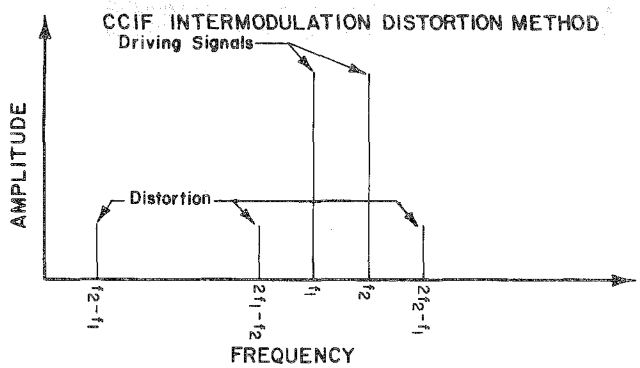


Figure 7 illustrates distortion measurement using the CCIF method.

$$\text{Even-order distortion [in percent]} = \frac{A_{(f_2-f_1)}}{A_{f_1} + A_{f_2}} \times 100$$

For this illustration $A_{(f_2-f_1)} = -62\ \text{dB} = .79\ \text{mv}$
$A_{f_1} = 0\ \text{dB} = 1\ \text{V}$
$A_{f_2} = 0\ \text{dB} = 1\ \text{V}$

$$\frac{.794\ \text{mv}}{1\text{v}+1\text{v}} \times 100 = .04\%$$

The odd-order distortion is determined by finding the ratio of the sum of the amplitudes of the two third-order products, $2f_1-f_2$ and $2f_2-f_1$, to the sum of the amplitudes of the two driving frequencies.

$$\text{Odd-order distortion} = \frac{A_{(2f_1-f_2)} + A_{(2f_2-f_1)}}{A_{f_1} + A_{f_2}} \times 100$$

For this illustration $A_{(2f_1-f_2)} = -62\ \text{dB} = .79\ \text{mv}$
$A_{(2f_2-f_1)} = -67\ \text{dB} = .45\ \text{mv}$

$$\frac{.79\ \text{mv} + .45\ \text{mv}}{1\text{V} + 1\text{V}}\ 100 = .06\%$$

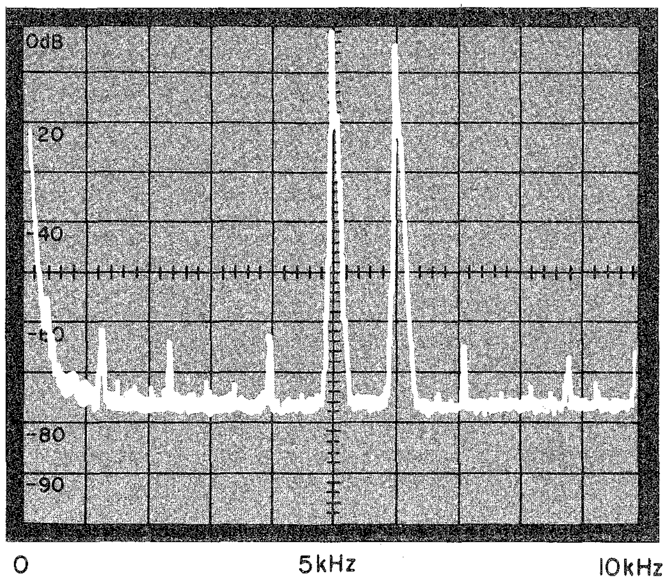


Figure 7. Distortion measurement using the CCIF method.

Tape Recorders

Flutter and Wow are terms used to describe a special type of intermodulation distortion normally associated with tape recorders. This type of distortion is due to variations in tape velocity across the recording or reproducing head caused by mechanical problems in the tape drive system. These variations result in frequency modulation of the recorded signal. The resultant frequency spectrum is similar to that obtained in the SMPTE method of measuring intermodulation distortion. In this case, however, the low frequency signal is generated by the fluctuations in tape speed and is not at any set amplitude. As with the SMPTE method, the spurious frequencies generated appear as upper and lower sidebands of the recorded signal. To measure flutter and wow, a test tape with a prerecorded tone of approximately 3 kHz is used. Flutter and wow analysis requires that the modulation sidebands be adequately separated for measurement. This is made possible with the Model 3580A because of the 1 Hz filter bandwidth. Figure 8 llustrates flutter and wow measurement using the Model 3580A.

Amplitude of 1st sideband = $1/2\,\Delta Q = -13$ dB = .224

$\Delta Q = .448$

$f_s = 3{,}000$ Hz

$f_m = 2$ Hz

$\frac{2\,\Delta Q\, f_m}{f_s} \times 100 =$ peak to peak flutter and wow [in percent]

$\frac{2\,[.448]\,2\text{ Hz}}{3000} = .06\%$

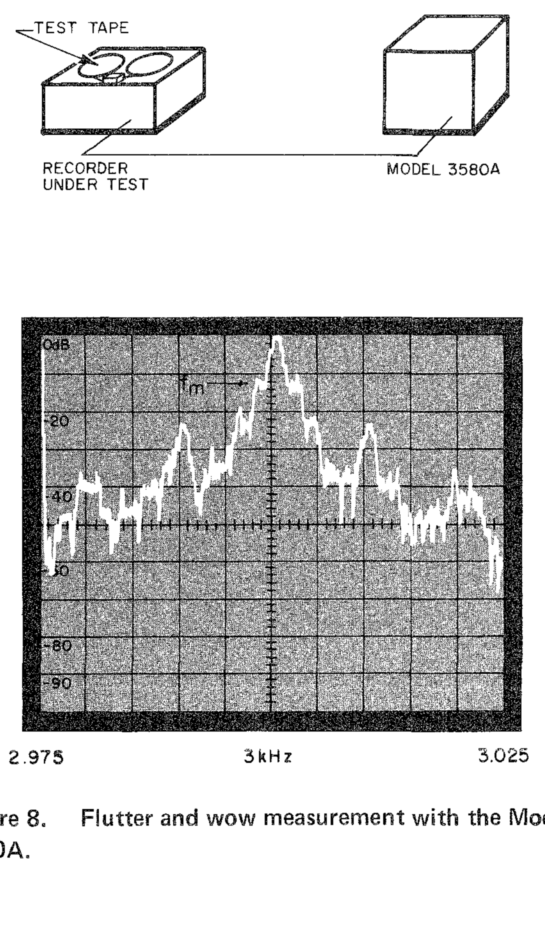


Figure 8. Flutter and wow measurement with the Model 3580A.

Musical Instruments

This is one case where the harmonic content is not called distortion. The presence of the right harmonics gives a note a full rich sound. The harmonics also account for the different voices of instruments.

Now a trumpet is always going to sound different than an oboe because of their basic structures. Each structure alters the amplitude of harmonics in a different way. Good and bad trumpet sounds will depend on the musician and small differences in construction.

A spectrum analyzer can be used to analyze a sustained note to see what amplitude relationships are associated with different instruments. Such an analysis will reveal large differences between various instruments. This suggests that one application of a spectrum analyzer would be for design of electronic organs.

The designer would like to do a better job of synthesizing the voices of various instruments. To do this, he can use a spectrum analyzer to discover which harmonics are present in the real instruments and their amplitudes. Then he tries to duplicate that performance electronically. When the two spectral responses are compared, the degree of match will be an indication of how much the organ sounds like the voice of another instrument.

To illustrate how a spectrum analyzer might be used to analyze the organ voices, Figure 9 shows the fundamental tones middle C and C sharp produce. The middle C has a frequency of 261 Hz and the sharp occurs at 271 Hz. There are spurious tones around 120 Hz and 370 Hz. Because these unwanted tones are at very low amplitude, they usually can't be heard in the presence of the wanted tones. We can now look at a wider scan and see the harmonics present in the normal organ voicing. Figure 10 shows a 5 kHz scan which exposes 15 harmonics. Notice that the first tone which is the fundamental has about the same amplitude as many of the harmonics. Given this as a starting point, the harmonics can be modified to provide flute, oboe and violin voicing. Using the spectrum analyzer, the designer doesn't have to dwell on the question of why a flute doesn't have high frequency harmonics. His job is to match the organ spectrum to a real flute's spectrum. The spectrum analyzer tells him how close he came to that goal.

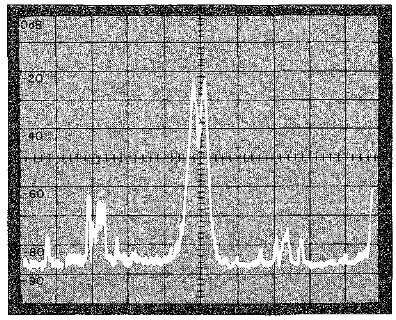


Figure 9. C and C sharp fundamental tones.

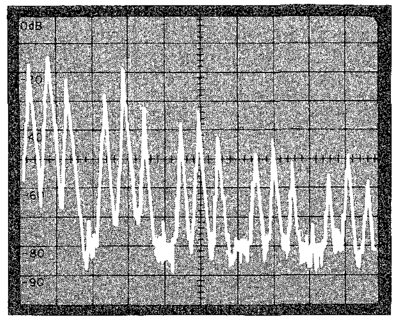


Figure 10. The harmonics of Middle C.

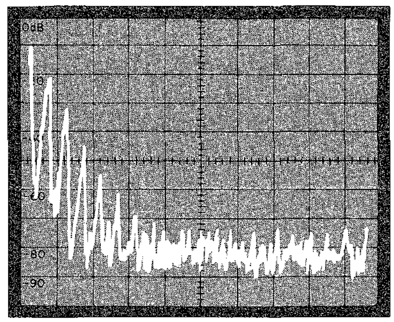


Figure 11. Middle C with a flute voice.

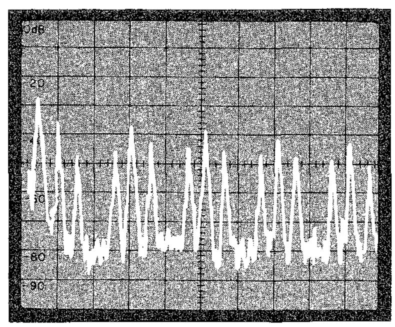


Figure 12. Middle C with a violin voice.

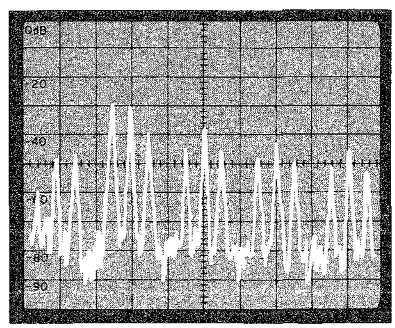


Figure 13. Middle C with an oboe voice.

Simply defined, frequency response is the ability of a device to pass or amplify, equally, all frequencies within a specified range. With respect to audio, this range is generally accepted as being from 20 Hz to 20 kHz. The importance of a device being able to reproduce signals over this entire range is not necessarily because fundamental tones are generated at all audio frequencies; actually, few musical instruments exceed fundamental frequencies greater than 4 kHz and the human voice extends only a little over 1 kHz. The importance lies in the fact that for true audio fidelity, the reproducing device must be able to handle all overtones (harmonics) which accompany the fundamental frequency. Without sufficient bandwidth, reproduction of the overtones, the sound is "flat" or "mechanical". A long distance telephone conversation is a good example of this effect because it does not produce frequencies over 4 kHz. The telephone line certainly represents something of a worst case example where the frequency response is deliberately cut off. To solve this apparent problem, a lot of money is spent to get amplifiers with very flat response to high frequencies. In the concert hall or home where the music is to be heard, there are many factors other than bandwidth that influence the characteristics of the sound. For example, the speakers and room acoustics have a tremendous effect on the sound.

The frequency response of Figure 14 shows what a room and speakers can look like even though the amplifier response was flat over this frequency range. While this picture looks like there is a lot of noise present, the many sharp peaks are caused by resonances. One conclusion is that the performance of any single component such as the amplifier does not tell the whole story. To characterize frequency response, all parts of the system have to be measured. A fast, efficient means of measuring frequency response is the "swept frequency" method. This method employs a sweep frequency generator as a signal source and makes response measurements over the entire range of interest in one sweep. The detector for these measurements may be either a wide-band device which measures the signal source and any spurious frequencies, or a tracking type which tracks the signal source and measures a narrow band of frequencies centered around the source frequency. To effectively use a wide-band detector, the test tone amplitude must be greater than any spurious tones or noise or harmonics. Because this may not be true at the frequency limits of the device under test, it is safer to use a narrow band detector. The narrow band detector is a better guarantee that the amplitude measured is that of the tone generator and is not influenced by some unknown.

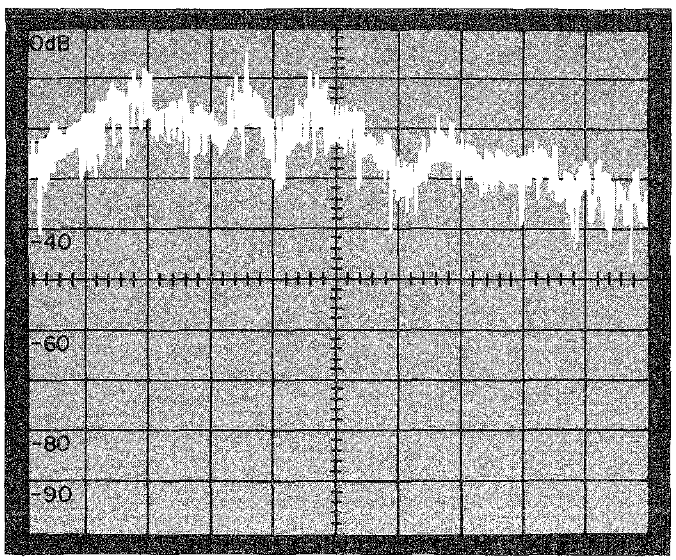


Figure 14. The detailed frequency response of a residential room as the frequency range from 0 to 10KHz is swept.

The Models 3580A and 3581A are good instruments for measuring a narrow band of frequencies as they are swept. Spurious frequencies caused by source distortion do not fall within the IF bandwidth of the analyzer and are, therefore, rejected. When using the Model 3580A or Model 3581A, the signal source for frequency response measurements is provided by the tracking oscillator output of the analyzer. This output is a sinusoidal signal, variable in amplitude from 0 to 2 volts rms, which tracks the internal tuned or swept frequency of the analyzer. With either of these simple instruments, a good frequency response measurement can be made.

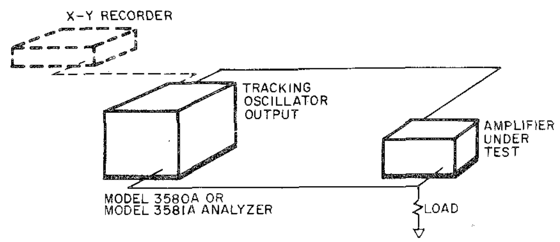


Figure 15. Amplifier frequency response measurement using the Model 3580A or Model 3581A.

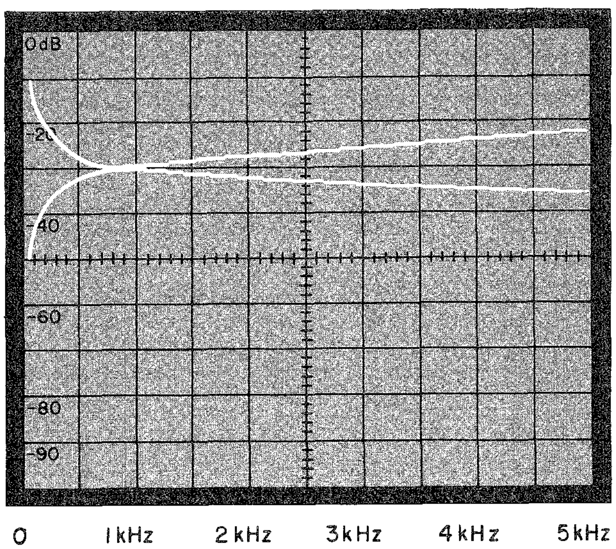


Figure 16. Bass and treble controls effect on the frequency response of an amplifier.

Figure 15 shows a typical setup for measuring amplifier frequency response. The signal source used to drive the amplifier under test is the tracking oscillator (BFO) output of the analyzer. The amplifier output is terminated with an appropriate load and connected to the analyzer input. Measurement of the frequency response is made by automatically or manually sweeping the analyzer across the frequency range of interest. It is possible to measure signal amplitudes of up to +30 dB with the Models 3580A or 3581A. This is equivalent to a power of 250 watts rms into a four ohm load. For documentary purposes, an X-Y Recorder can be connected to the recorder outputs provided, to automatically record the measurement or, when using the Model 3580A, a scope camera can be used to record the CRT display. Figure 16 shows how the bass and treble controls effect the frequency response of an amplifier. Using the storage feature, amplitude range and frequency characteristics of these controls become obvious.

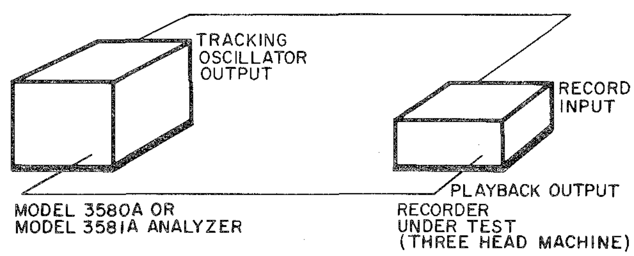
BANDWIDTH	SWEEP RATE				
	5 kHz/sec	500 Hz/sec	50 Hz/sec	5 Hz/sec	.5 Hz/sec
300Hz	7.5 ms	75 ms	.75 sec	7.5 sec	75 sec
100 Hz	2.5 ms	25 ms	.25 sec	2.5 sec	25 sec
30 Hz		7.5 ms	75 ms	.75 sec	7.5 sec
10 Hz		2.5 ms	25 ms	.25 sec	2.5 sec
3 Hz			7.5 ms	75 ms	.75 sec
1 Hz			2.5 ms	25 ms	.25 sec

Figure 17. Maximum time delay between BFO output and input to analyzer as a function of analyzer bandwidth and sweep rate.

To measure overall frequency response of multiple head tape recorders, the tracking oscillator signal is connected to the record amplifier input and the output is taken from the playback amplifier. This allows the recording and measurement of frequency response to be made in one "pass" and eliminates the problem of synchronizing the analyzer sweep to the recorded signal. See Figure 18 A problem when using this method occurs because of the physical displacement between the record and playback heads. This causes a time delay between the input and output signals which results in a significant frequency difference. As an example, at a tape speed of 7-1/2 inches per second and a head displacement of 1-1/2 inches, the time delay would be 200 milliseconds. (1.5 inches/ 7.5 inches per second = .2 seconds). At a sweep rate of 500 Hz per second, this would mean a frequency difference of 100 Hz between the input and output signal. If we used the 100 Hz filter bandwidth that is available in the 3581A or 3580A, the signal would fall outside the bandpass limits and the results would be of little value. This can be corrected simply by using a much slower sweep speed or narrower bandwidth.

This method also applies to other swept measurements where time delays are encountered, such as speaker response and room acoustic measurements.

Figure 17 shows the maximum delays for various sweep rates and bandwidths. For the case of a 200 ms delay, a 50 Hz/sec sweep rate on the 100 Hz bandwidth would give correct answers for this delay. Other bandwidths and sweep rates would also give the same results. On the analyzer, the 50 Hz/sec sweep rate can be achieved through a number of different settings of span and sweep time. For instance, covering a span of 10 kHz at a 50 sec/ cm rate yields the required 50 Hz/sec sweep rate.



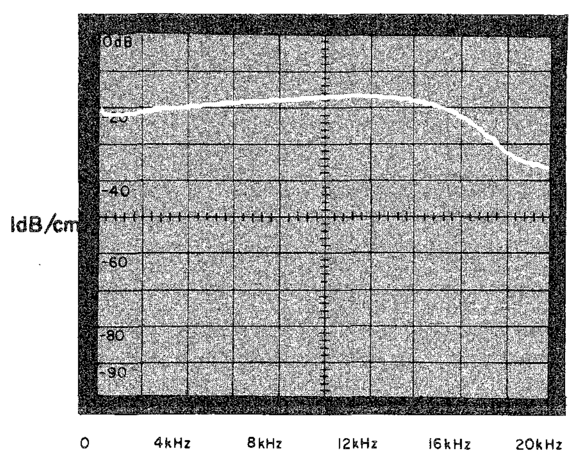


Figure 18. Tape recorder frequency response test connections. Test signal is recorded and measured in one "pass."

Frequency Response of Equalizers

A larger problem is this characterization of equalizers. The complete job is very time consuming unless a spectrum analyzer's characteristics are used.

* Shown on Figure 19 are the 12 dB shelving characteristics or two different equalizers of the same model. The left set of shelves are almost perfectly symmetrical about the reference line. Its companion does not quite match it in this respect. Note the 50 Hz cut curve as compared with the 50 Hz boost. Note also, the low frequency set of curves, shown as maxima and minima, exceed the ± 12 dB nominal values by almost 2 dB as judged by the ± 12 dB reference lines, in both cases. The high frequency shelves exceed nominal maximum boost and cut by about 1 dB.

The 50 Hz – 15 kHz "filter" function is also shown with the shelving family. It appears to be identical for both equalizers. Note the upward curve of the center reference line. It rises about ½ dB at about 15 kHz to about 2 dB at about 40 kHz. This appears to be a common characteristic of the brand "x" equalizers tested. The HP 3580A tracking generator output checked out flat across the band within .2 dB of reference at 5 kHz. All amplifiers in the console were tested with a 604 Ohm load termination. The cause of this rise was not determined nor was it established that this rise was at all troublesome.

Figure 20 shows the same two equalizers in the same order as Figure 19. Comparison of the maximum and minimum extremes shows that all agree closely with the ± 12 dB nominal.

On the left set of curves, locate the fifth notch from the left. This is the 400 Hz curve that is duplicated in the LF and MF sections of the equalizer. This curve appears slightly heavier than the others because the LF and MF 400 Hz center frequencies are not exactly identical. They do overlay each other very closely in the left family. Comparing this in the right set of curves, we see that the LF and MF 400 Hz notches do not match as precisely as in the left set.

These are variations noted in two out of thirty-two equalizers. It would have been interesting to have checked more units, had time permitted, to see how great the range of deviations from nominal performance might be. After some practice, it was possible to generate a full set of extreme shelving, peaking and notching curves in about 11 minutes, taking time to apply the Polaroid preservative solution to each of the two photos and to make identifying notes regarding each. Both shelving and notching could have been recorded on a single photo had the HP 197 camera been adjusted to sufficiently offset the two tracks. Doing this could have trimmed several more minutes off of that 11 minute time.

The entire set of thirty-two equalizers could have been checked in about six hours, excluding interruptions of biological or other necessity, and the test results inspected at leisure.

Figure 21 shows the incremental peaking response of a brand "x" equalizer at 100 Hz and 15 kHz. The +2, +4, +6, and +9 increments show up nicely and exhibit remarkable precision, but the +12 dB setting is off the screen. Marker traces are shown at reference level and at 2 dB increments.

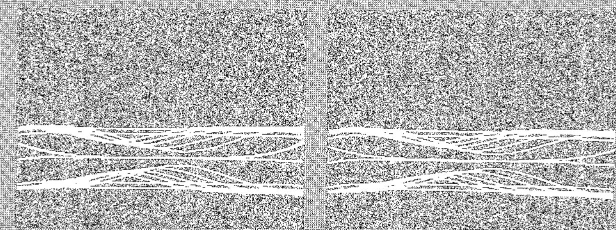


Figure 19. Shelving equalizer family of curves for two separate equalizers. 10 dB/vertical cm with horizontal reference times at ±10dB and ±12 dB, log frequency sweep, 20 Hz to 43 kHz.

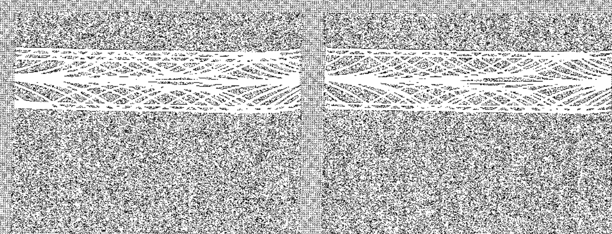


Figure 20. Peaking and notching curves for two separate equalizers. 10 dB/vertical cm with horizontal reference lines at ±10 and ±12 dB, log sweep, 20 Hz to 43 kHz.

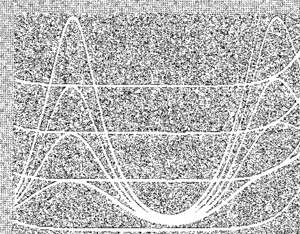


Figure 21. Equalizer peaking curve boost increments, at 100 Hz and 15 Hz center frequencies. Expanded scale 1 dB/vertical cm with calibration lines at 2 dB intervals log sweep, 20 Hz to 43 kHz.

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Acoustic Response

To get a more complete idea of the frequency response as perceived by the user, it is not enough to measure just the electronics. The speaker and room characteristics have a large effect on the sound. The effects of the speaker and room can be separated by first doing a frequency response of the speaker output in an echo free environment. An anechoic chamber is the obvious answer but it is also the most expensive and does have low frequency limitations. Another solution is to do the frequency response measurement outdoors where there are no buildings to cause reflections. Directing the speaker up also eliminates the reflections from the ground. Figure 22 shows the frequency response of such a measurement.

The next step is to characterize the room response. There are several methods of measuring room response. The source could be either a swept tone or white noise. The analyzer could be either a 1/3 octave or narrow band analyzer. The choice of source and analyzer depends to a large degree on the particular compensation philosophy used.

Looking at sources, noise sources come closest to duplicating the music or voice signals that will be used. It, therefore, should give a realistic picture of the frequency response. A tone tends to excite the resonant room modes more than the noise source. The swept tone is useful for situations where the resonant modes are causing problems and need to be identified.

Another choice is in the analyzers used to measure room response. A 1/3 octave analyzer tends to average the sharp resonances and give an overall picture. The narrow band analyzers show more detail. The following discussion describes what can be done with a swept source and a narrow band analyzer.

Figure 14 showed the tremendous effect the room had on the response. Figure 23 illustrates the effect of the many reflections and modes of driving a room. From this figure, it seems that the room has a greater effect on the sound than any other factor. Fortunately, the many nodes and antinodes are very sharp. They, therefore, effect only a very small amount of the frequency range and are not always fully excited, so the apparent effect to the listener is small. The biggest problem these resonances present occurs when there is feedback. These nodes limit the acoustic power that can be used without causing feedback.

Another measure of room acoustics is reverberation time. This is a phenomena that is important for large auditoriums. The goal is not to eliminate all reverberation because they would destroy the live qualities. On the other hand, too much reverberation will make speech unintelligible. With these limits in mind, a lot of research has been done to find the optimum reverberation time based on room volume, type of material and frequency. By surveying a room before a sound reinforcement system is attempted, the sound engineer can make estimates of the effectiveness of such a system. This has proven to be important because there is no electronic solution for bad reverberation characteristics. To make the measurement, a tone is used to excite the rooms various modes. When the tone is removed, the decay time is measured. Figure 24 shows such a measurement. This process is repeated for other tones resulting in a plot of reverberation time vrs. frequency which can then be compared to the desired characteristics.

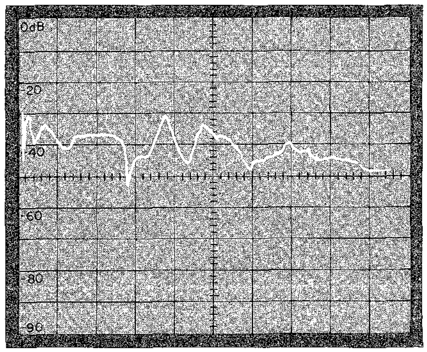

Figure 22. Outdoors.

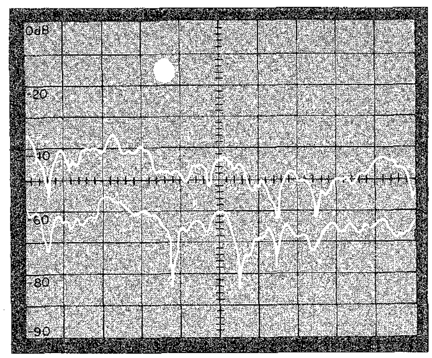

Figure 23. The two response curves (offset by 10dB) show the changes from simply opening a door in the room.

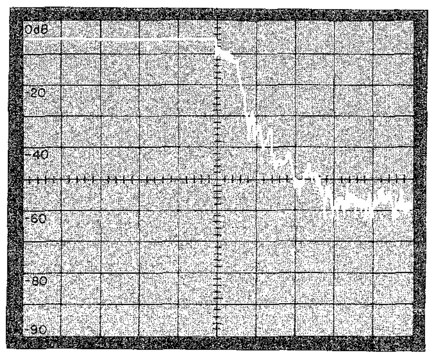

Figure 24. Decay time when the tone is removed.

Signal-to-Noise Ratio

Signal-to-noise measurements can be made in conjunction with frequency response. A response measurement is made at the desired signal reference level and recorded. When using the Model 3580A, this measurement may be stored in "memory". The drive signal is then removed from the device under test and the input is shorted or terminated with an appropriate load. The analyzer is again swept across the frequency range covered in the response measurement. With the 3580A, the noise measurement is now displayed on the CRT along with the signal measurement which was stored in memory. See Figure 25.This method makes possible comparison of signal-to-noise over the entire frequency range of interest.

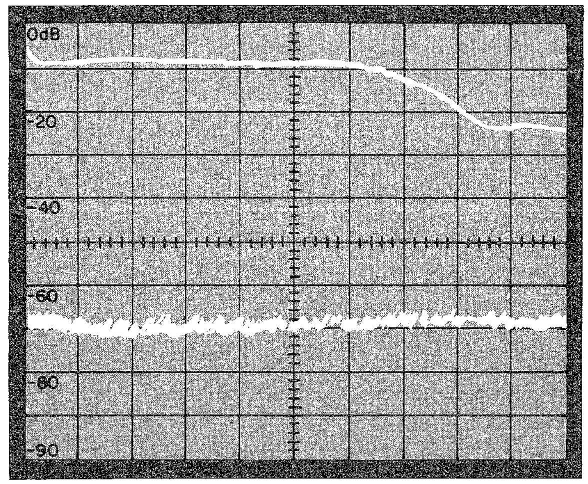

Figure 25. *Signal measurement stored in memory.*

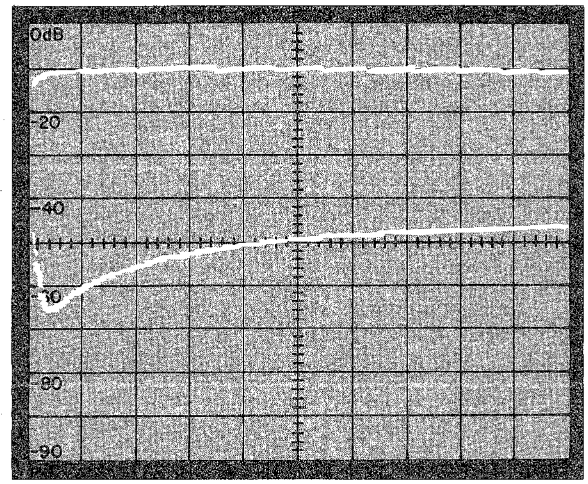

Figure 26. Cross-talk over entire frequency.

Cross-talk

Cross-talk is measured in a manner similar to that of signal-to-noise ratio. Again, a frequency response measurement is made and recorded at the desired reference level through one channel. The analyzer input is then removed from this channel and connected to the second channel. The analyzer is then swept over the frequency range while driving the input of the first channel and measuring the output of the second. This measurement together with the reference measurement documents cross-talk over the entire frequency range of interest. Figure 26 illustrates measurement results.

*Cross-talk measurements have always been a hassle using conventional methods. Even with a real, live wave analyzer, getting a good idea of how much unwanted signal is leaking into where at what frequency takes the patience of a monk. Plotting the curves from manually recorded data is always time consuming. Figure 27shows an adjacent channel cross-talk sweep from 0 Hz to 50 kHz. One input channel was driven with a −50 dBV sweep from the 3580A tracking generator while an adjacent channel output was observed on the display screen.

Although the numbers aren't apparent due to our lack of graticule, the leakage lies below −70 dBV up to about 10 kHz, rising to about −50 dBv at around 45 kHz. The driven channel gain was adjusted to yield +4 dBm at 1 kHz. The lower trace shows the program channel noise, with the equalizer in the circuit set for flat, over the 50 kHz band. The estimated average, unweighted noise over the bottom 20 kHz band is approximately −77 dBV. A 2 kHz per division horizontal scale factor would have yielded somewhat more accurate data. The cross-talk curve took about 20 seconds to trace while the noise curve took about 100 seconds.

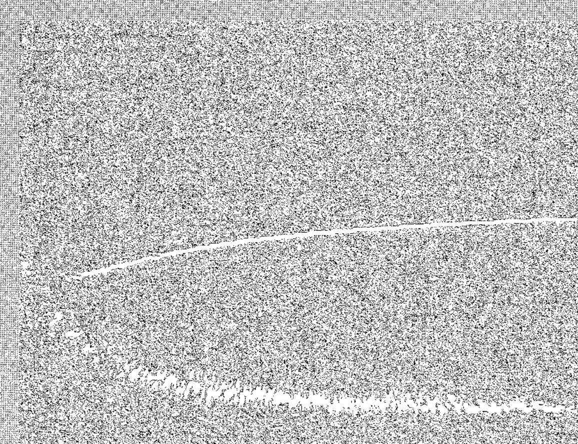

Figure 27. Adjacent channel cross talk (upper trace), and channel residual noise (lower trace). 10 dB/vertical cm. 5 kHz/horizontal cm (linear sweep).

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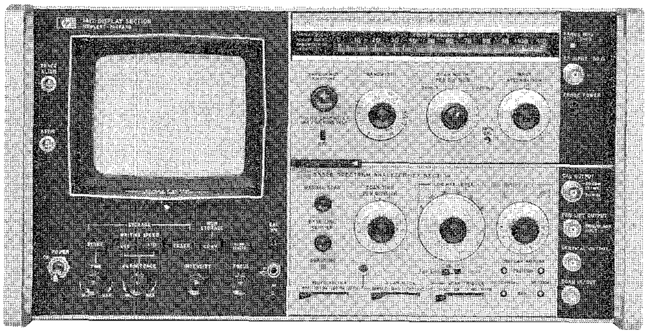

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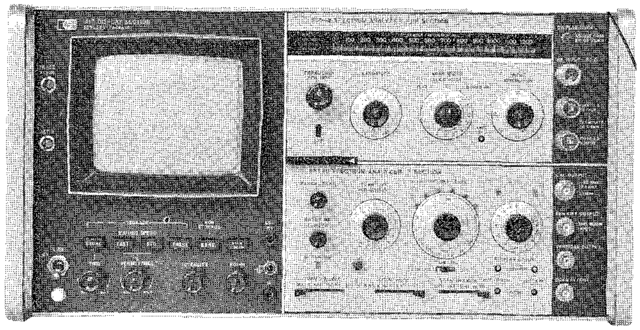

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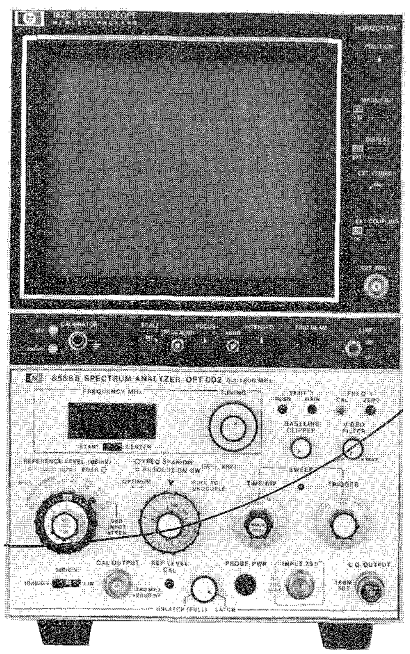

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