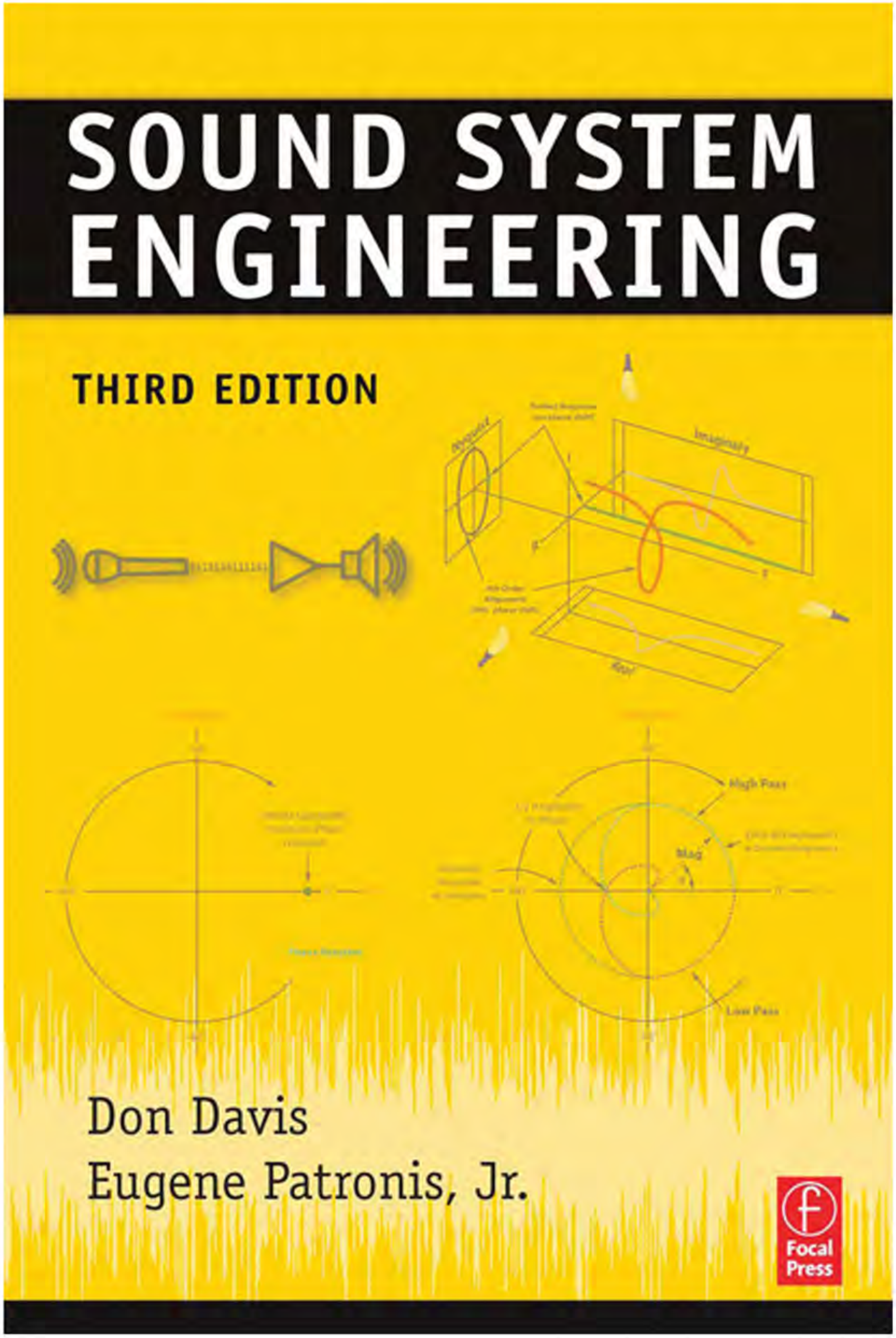

SOUND SYSTEM ENGINEERING
THIRD EDITION
Imaginary
High Pass
Mag
Low Pass
Don Davis
Eugene Patronis, Jr.
Focal Press

Chapter 6

# Audio and Acoustic Measurements


## by Don Davis

In order to better understand what we hear we often turn to measurements. As one authority in theory once remarked, "When the number of variables approached an order of magnitude, I turn in despair to my measurement apparatus."

The finest acoustical measurement apparatus available cannot duplicate what a trained human listener can achieve. If we examine an unknown signal with all extant equipment we can't tell if its music, noise, speech, or gibberish, but a $2 loudspeaker allows the trained human listener to tell which, and if speech, what language.

Instrumentation is used to measure room parameters before the design begins, to compute design factors, to install the system, and, finally, to operate and maintain the system. The greatest single division between professional work and nonprofessional work in the system business is the use and understanding of basic audio and acoustic instrumentation.

The following quote is pertinent to the intent of this chapter:

> *I often say that when you can measure what you are speaking about, and can express it in numbers, you know something about it; but when you cannot measure it, when you cannot express it in numbers, your knowledge is of a meager and unsatisfactory kind. It may be the beginning of knowledge but you have scarcely advanced to the stage of science whatever the matter may be...*
>
> Lord Kelvin 1824-1907

## 6.1 Initial Parameters

To make a measurement, choose the initial parameters by one of the three following techniques:

1. Experience with similar devices.
2. Mathematical analysis of the device and its most likely performance.
3. Cut and try experimentation.

Component designers often, justifiably, use Step 3. System designers must not. System designers need to specify proven, trusted components. Systems are complex by their very nature of combining components from many manufacturers. To increase that complexity with untried components is irresponsible.

## 6.2 Acoustic Tests of Sound Systems

Once all the electrical tests of the sound system are completed and any electrical problems are corrected meaningful acoustic tests can be performed to verify:

1. Output levels and areas of coverage for individual transducers comprising the acoustic output of the system.
2. The phase and polarity of the individual transducers.
3. The signal synchronization between different transducers sharing identical areas of coverage, i.e., overlap zones.
4. The absence of any undesired spurious energy returns from any reflective surface.
5. The measurement of the relationship of $L_D - L_R$ to confirm the $\%Al_{CONS}$ at selected audience locations.
6. The equalization of $L_D$ as required.
7. The loudspeaker impedances, see Fig. 6-1.

### *Acoustic Test Signals*

Sound engineers have available many different test sources:

**Music and speech.** Excellent if the listener is highly trained—a rarity.

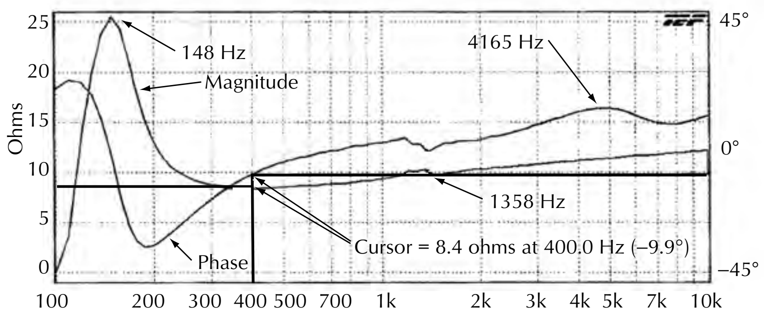
**The steady-state sinewave.** While perhaps our most useful electrical signal, it is rarely useful in acoustic tests.

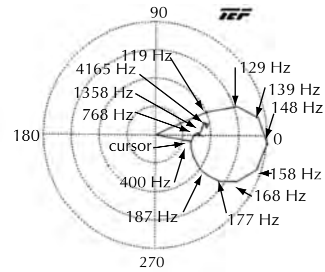
**Swept sinewave.** This source is our single most useful test signal.

**Random noise.** White, pink, USASI (or ANSI), and other special forms of noise are useful for magnitude measurements, but they pose too great a complexity in the acquisition of phase measurements.

**Impulse sources.** These sources represent the worst possible choice for acoustic measurements, especially when used in conjunction with FFT analysis. They offer the least effective use possible of the test signal's energy.

The starting point for any serious acoustic measurement system is the calibrated measurement microphone. Fig. 6-2 allows the comparison of the human ear with a quality electret microphone.




Just a glance at the Nyquist display reveals which frequencies are predominately resistive, capacitance, or inductive impedance values.

$Z = |Z| e^{j\theta}$
where,
$|Z|$ is the magnitude of the impedance
$\theta$ is the angle of the impedance

$$\text{Reactance} = |Z| \sin\theta$$
$$\text{Resistance} = |Z| \cos\theta$$
$$\text{Angle} = \arctan\left(\frac{\text{Reactance}}{\text{Resistance}}\right)$$

| | |
|---|---|
| Frequency | 400 Hz |
| Phase angle | –9.88° |
| Impedance | 8.41 Ω |
| Resistance | 8.29 Ω |
| Reactance | –1.44 Ω |

**Figure 6-1.** Loudspeaker impedances.

Notice that "intelligent design" has allowed the microphone, in some parameters, to exceed the human ear and in others to equal it for all practical purposes. One of the authors worked for a company that built a microphone capable of $L_P$ = 220 dB (for measuring the pressure wave from a hydrogen bomb explosion which produced a spike well above atmospheric pressure).

| *Specification* | *Ear* | *Mic** | *Units* |
|---|---|---|---|
| Size | 12 | 0.17 | $cm^3$ |
| Power Consumption | 50 | 25 | µW |
| Vibration Sensitivity (1 g) | 100 | 75 | dB SPL |
| Shock Resistance | 100 | 20,000 | g |
| Noise Level (A-weighted) | 20 | 20 | dB SPL |
| Overload Level (10% THD) | 100 | 140 | dB SPL |
| Dynamic Range (Pure Tone) | 100 | 140 | dB |
| Acoustic Input Z (low frequency) | 1.4 | 0.03 | $cm^3$ |
| Frequency response | 25–16,000 | 10–25,000 | Hz |

*Small electret

**Figure 6-2.** Comparison of an ear and a microphone. (*Courtesy Dr. Mead Killion*)

### *Where to Place the Microphone*

This frequent query has an easy answer: where your ears tell you there is a problem, or where you need to look at the radiation pattern for adjustment. For example, the point equally distant from two loudspeakers where you desire to "signal align" them for minimum polar response interferences, see Figs. 6-3A and 6-3B. Another example, the area where your ears tell you that intelligibility has suffered for no visually apparent reason and the measurement of the Envelope Time Curve, ETC, reveals an unexpected focused reflection damaging the direct sound level in that area.

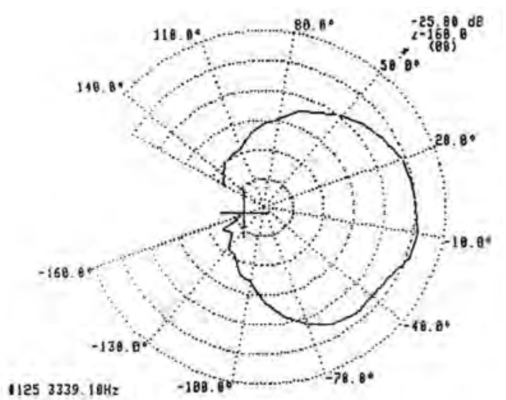
### *Measurement Analyzers*

The authors prefer Richard C. Heyser's analysis system as exemplified in a TEF instrument. This is not to deprecate other devices but is the result of the superior signal-to-noise parameters so vital to field measurements. The Heyser Integral Transform is unique, Fig. 6-4. While it has yet to realize its full potential in real instruments, its embodiment in what's currently available has radically changed how we measure. Indeed the frequency modulation function identified by Heyser has found an embodiment in HP's modulation domain analysis, Fig. 6-5A, 6-5B, and 6-5C.

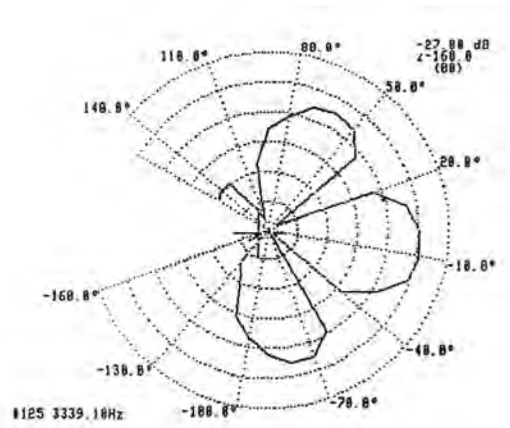
### *First, Look and Listen*

Upon arrival at a measurement site, first look and listen. We should visually inspect the site and then listen, sans sound system. Take time to walk the audience area and listen to a live talker standing where the performer will be. Listen for noise masking of the talker—echoes, focused reflections, and strange level dropouts, i.e., cancellations. Cupping your ears allows some directional discrimination with regard to reflections. This exercise helps to quickly ascertain if it's the room or the system or both that need correction. Many existing sound systems in difficulty exist in rooms where the unaided voice can be heard clearly.

A first look and listen allows identification of logical measurement points. Once a given point is

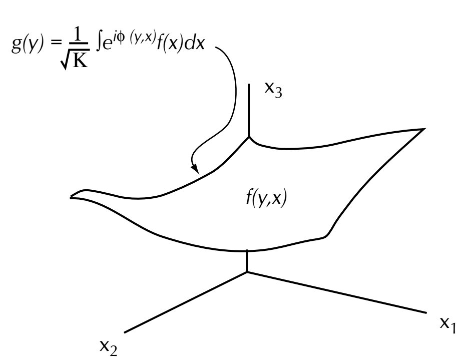



A. Horizontal polar response of two loudspeakers one stacked on top of the other and in physical alignment.




B. Horizontal polar response of two loudspeakers one stacked on top of the other and out of physical alignment by 3 inches.

**Figure 6-3.** The effect of mis-alignment on loudspeaker output.




$x = x_1, x_2, \ldots x_n$  $y = y_1, y_2, \ldots y_m$  $m \lesseqgtr n$

*f(y,x)* is a hypersurface, defined in parameters *y* and expressed over all *x*.

**Figure 6-4.** Heyser Integral Transform.

selected make a global Envelope Time Curve, ETC. The time base needs to be at least twice the room's largest dimension, or longer than the "by the ear" estimate of the room's $RT_{60}$. The source used should be one loudspeaker from the array that has the microphone in its path. If inspecting the room before the sound installation is possible, use a test loudspeaker suitable for such a space or if that is not possible, at a minimum, use one with a known $Q$. Whenever possible, mount the loudspeaker, via a portable hoist, in a logical location for a proposed system.

A global view of the time domain insures that late arriving energy is not overlooked and at the same time allows estimating the shorter time scales that will be employed to obtain more detail. A rule to remember is that you can truncate long time to shorter time but not the reverse. From the first global measurement, we looked at the Heyser Spiral. See Fig. 6-6A, followed by the ETC in Fig. 6-6B. See Fig. 6-6C for the truncated ETC of a shorter interval which revealed a missynchronized package loudspeaker that was non-minimum phase. See Fig. 6-6D for the Nyquist, where the curve encircled the origin.

These initial measurements revealed a few of the items needing correction in a minimum amount of time.

## 6.3 The ETC Plot

The Envelope Time Curve, ETC, is related to a well- established concept in communication theory known as the modulation envelope. The Envelope Time Curve is the magnitude of the analytic signal description of the impulse response.

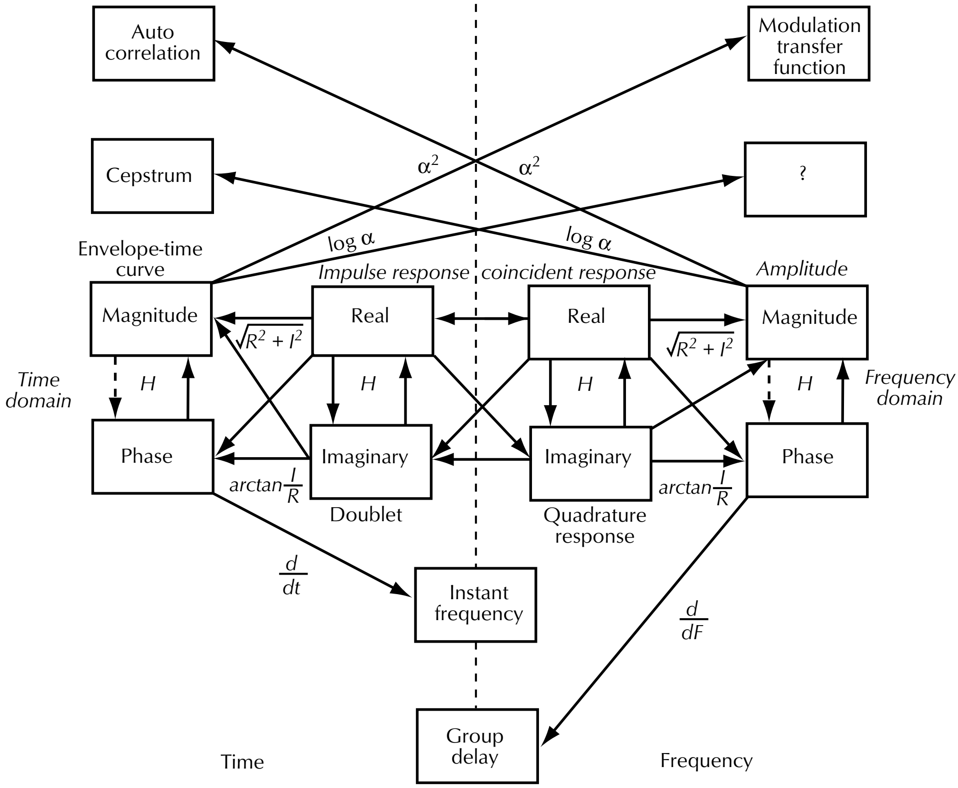
In the acoustical measurement case, let $I$ represent the impulse response which is a real function of time and let $\bar{I}$ represent the Hilbert Transform of the impulse response. Also, let $I_A$ represent the analytic impulse response. Then,

$$I_A = I + j\bar{I} \ . \qquad (6\text{-}1)$$

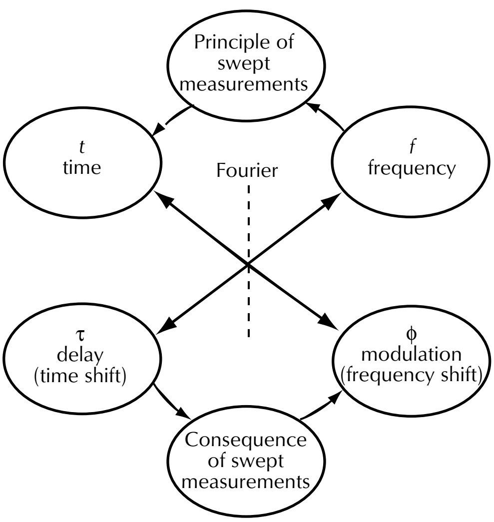
Now, consider the quantity

$$20\log\frac{\sqrt{I^2 + \bar{I}^2}}{2 \times 10^{-5}} = 20\log\frac{|I_A|}{2 \times 10^{-5}} \ . \qquad (6\text{-}2)$$

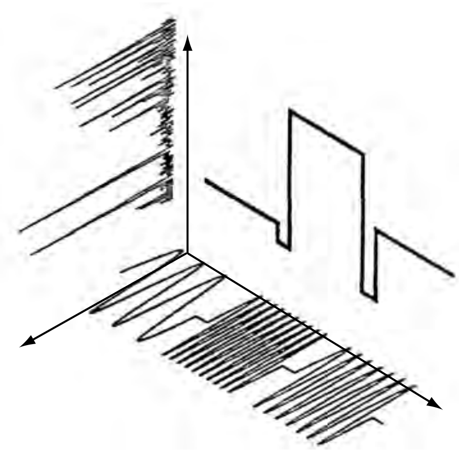
This is the quantity plotted versus time in forming the curve known as the ETC. This is similar to a smoothed version of the impulse squared response. It has proven its worth over the years in identifying




A. TEF measurements.



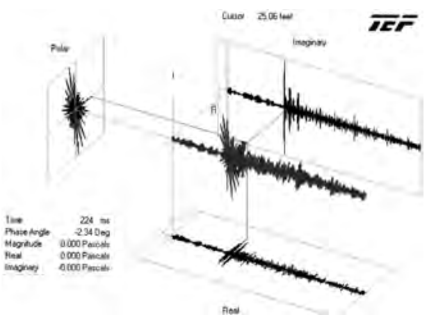

B. When the frequency response of a time invariant system is measured as a function of time the (time) delay response is converted to a (frequency) modulation function.



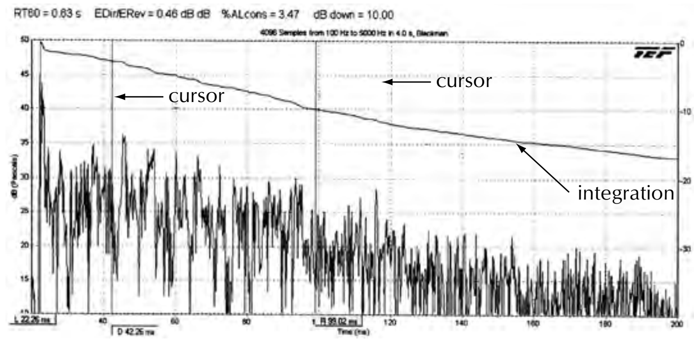
C. The same signal can be represented in the time domain on an oscilloscope (bottom) and in the frequency domain on a spectrum analyzer (left). Hewlett-Packard's 5371A frequency and time-interval analyzer shows the signal's frequency against time, inaugurating what HP calls the modulation domain. The analyzer simplifies such measurements as timing jitter, frequency drift, and modulation on communications signals.

**Figure 6-5.** Processing TEF signals.

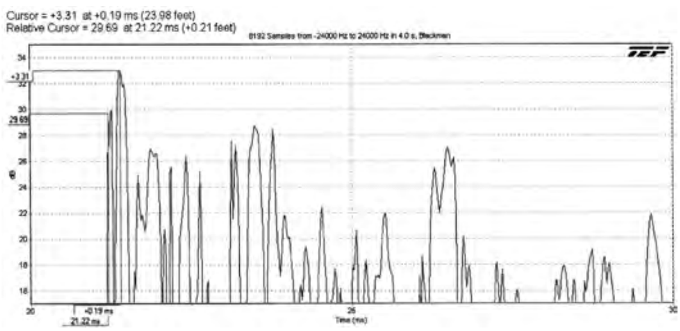
detrimental reflections, locating the desired signal delay corrections for magnitude measurements in the frequency domain, usually "fine tuned" by microsecond adjustments of the phase response, and in examining the density or lack of density of the reflected sound field at any given point in space.



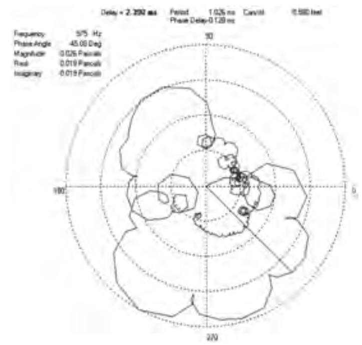
A. A Heyser spiral of the room in the time domain




B. The Envelope Time Curve (ETC). Note that the cursors (readout above plot) solve for $RT_{60}$, $\%AL_{CONS}$, and direct to reflected levels.



C. ETC with shorter time interval than Heyser spiral revealing a non synchronized loudspeaker



D. Nyquist display showing non minimum phase encirclement of origin

**Figure 6-6.** Variety of TEF displays.

### *Impulse Response*

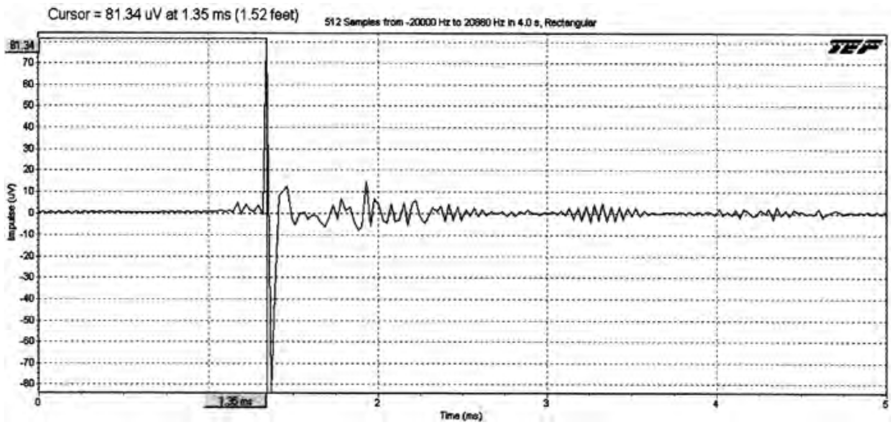
Today impulse responses are acquired in the frequency domain both to address undue stress on the loudspeaker system and to obtain an improved signal-to-noise ratio, *SNR*. It is then inverse Fourier transformed to the time domain where it can be displayed in a number of forms, see *Chapter 14 Signal Processing*, for an explanation of Fourier transform. The Fourier transform takes both the real and the imaginary parts, from the amplitude and phase measurements in the frequency domain, to compute the impulse response in the time domain, Fig. 6-7A. Additionally, the impulse response can be Hilbert transformed to produce the doublet response. The impulse response forms the real part of the complex ETC while the doublet response forms the imaginary part of the complex ETC, Fig. 6-7B. Fig. 6-7C shows the relationship between the real and imaginary parts on the Heyser Spiral.

Modern analyzers provide differing viewpoints of the same information. A good example is the comparison of the log squared amplitude of the impulse response, Fig. 6-7D, to the ETC, Fig. 6-7E. The impulse response contains all the data but the ETC clearly shows some arrivals with greater clarity due to the modulation domain aspect of the envelope.

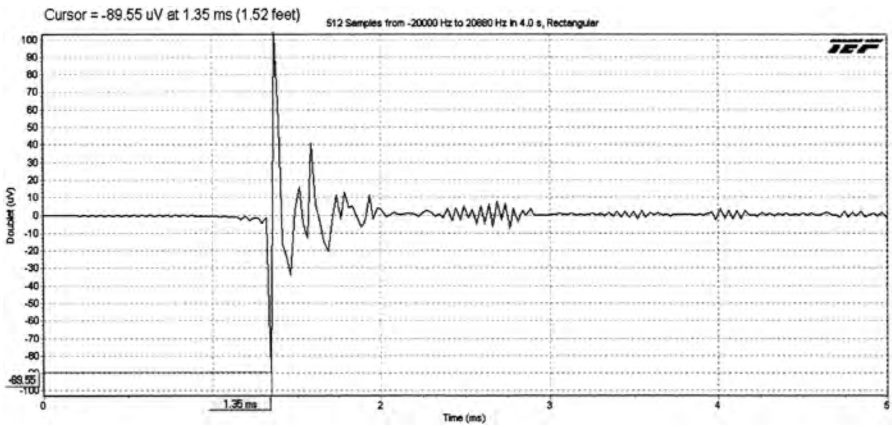
### *The Heyser Spiral*

Richard C. Heyser's remarkable insights, so often copied, so seldom acknowledged, that signal acquisition in the frequency domain via a swept sine-wave (chirp) tracked by a time offset tracking filter yields vastly superior *SNR* in both the frequency domain and the inverse Fourier transformed time domain.

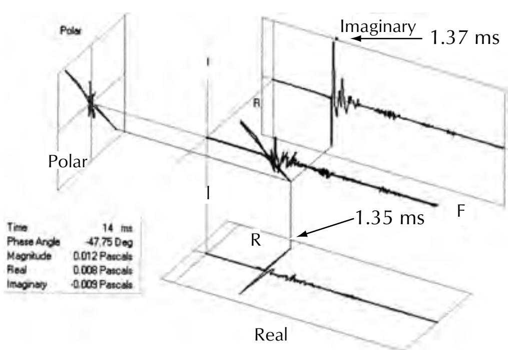
The easiest visualization of these processes is the Heyser Spiral display. Fig. 6-8A shows the frequency domain Heyser Spiral composed of the complex signal on the frequency axis, the real and imaginary parts shadowed on the appropriate planes, and the Nyquist trace of the complex signal.




A. TEF impulse.




B. TEF doublet.




C. Heyser response in the time domain

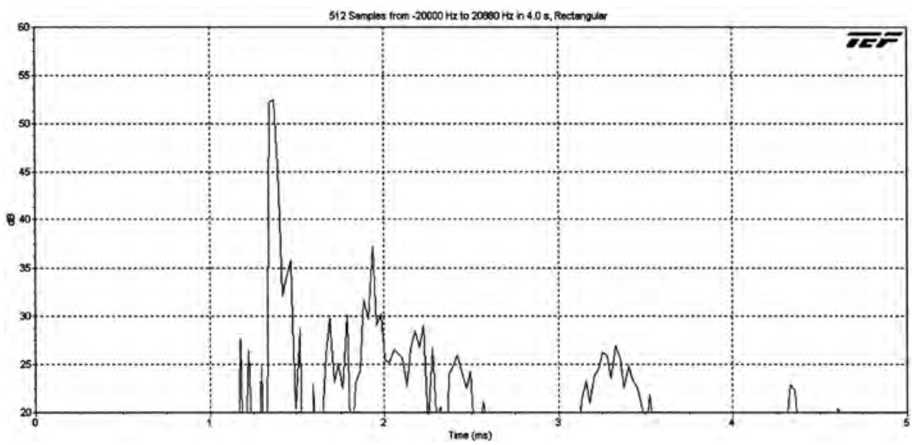
**Figure 6-7.** Time domain displays.

An inverse Fourier transform of both the real and imaginary parts in the frequency domain produces the impulse response (real) in the time domain, Fig. 6-8B.

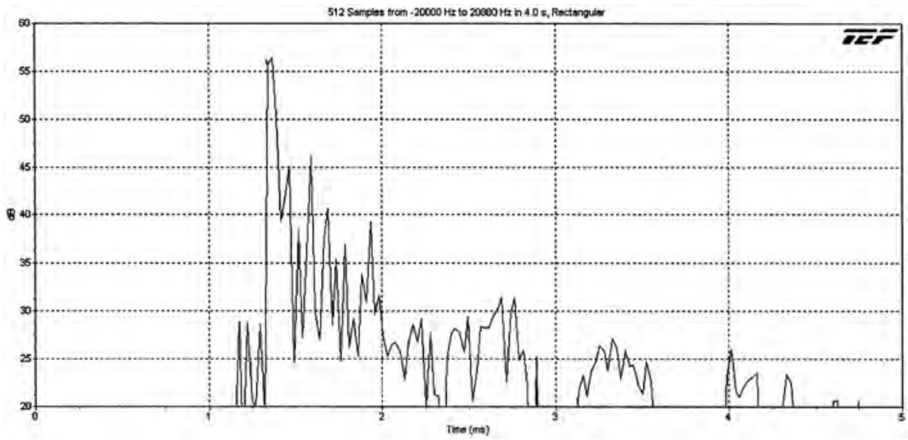
A Hilbert transform of the impulse response (real) produces the doublet response (imaginary), Fig. 6-8C. These real and imaginary parts yield the Envelope Time Curve.

### *The Magnitude and Phase Response*

Prior to Heyser, real life data from manufacturers was usually frequency vs. level responses and rarely phase response. The magnitude response is the most familiar measurement to many. It has limited value without the accompanying phase response. The phase response requires "fine tuning" via the micro-




D. Log$^2$ implulse.




E. ETC.

**Figure 6-7. (continued)** Time domain displays.

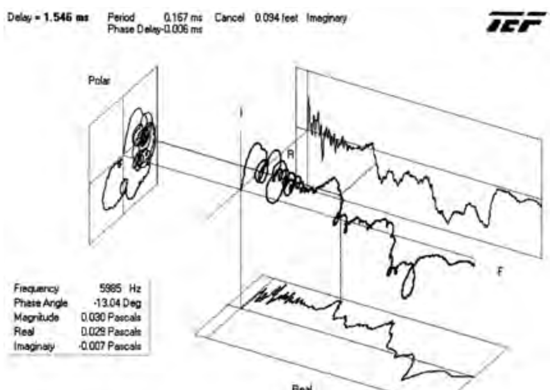
second adjustments available in modern analyzers. The adjustment is used to bring the phase response to 0° wherever the magnitude response is uniform. Once this has been done, if the device is minimum phase the peaks and dips on the phase response will be opposite the "slopes" on the magnitude response.

Why all the emphasis on minimum phase response? It is because you cannot apply conventional inverse equalization to the magnitude response unless that portion of the magnitude response is minimum phase. Non-minimum phase usually implies a significant signal delay.

Magnitude response has a vertical decibel scale and a horizontal frequency scale. Phase has a vertical scale in plus or minus degrees and a horizontal frequency scale. In all measurements it is vital to know what the frequency resolution is. Resolution that is too broad gives optimistic smoothing whereas resolution too narrow includes, in many cases, undesired reflected information.

Measuring phase instead of magnitude provides greater sensitivity and resolution. For example, finding resonant frequencies (phase passes through zero at resonance), the phase response will typically be 10 times more sensitive than the magnitude response. Acoustic delay problems jump out in phase and can be difficult, at best, with magnitude response. The pairing of magnitude and phase, Fig. 6-9 (upper two curves), allows detection of non-minimum phase frequencies—the phase inflection points don't intersect the center of the magnitude slopes. Further, a flattened phase response over a selected range reveals that the magnitude correction was properly done, Fig. 6-9 (lower two curves).

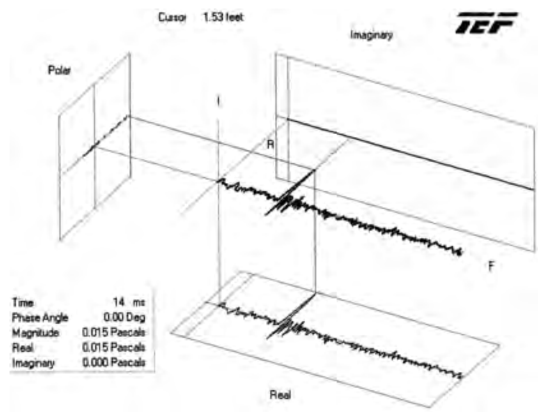
A non-minimum phase system is one that exhibits an excess delay of the signal over that termed the phase delay. Since an increasing number of audio devices include, either deliberately or accidentally, all-pass components, phase measurements are of ever increasing importance.



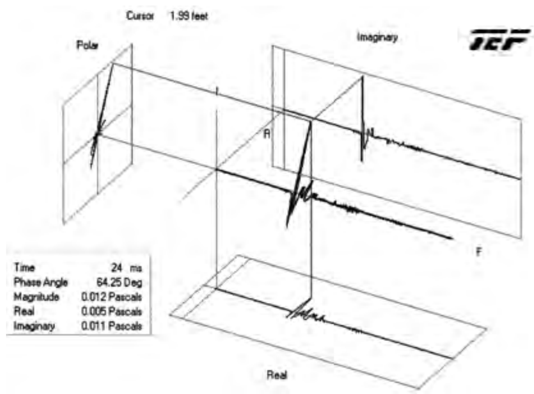

$$20\log\left(\frac{0.030\ \text{Pa}}{0.00002\ \text{Pa}}\right) = 63.5\ \text{dB}$$

$$20\log\left(\frac{\sqrt{(0.029)^2 + (0.007)^2}}{0.00002\ \text{Pa}}\right) = 63.5\ \text{dB} \qquad \text{atan}\left(\frac{-0.007}{0.029}\right) = -13°$$

A. Heyser Spiral in the frequency domain.




B. Fourier transform from frequency domain to time domain of real and imaginary parts in the frequency domain yields the impulse response.




$$\text{magnitude} = 20\log\left(\frac{0.012\ \text{Pa}}{0.00002\ \text{Pa}}\right) = 55.6\ \text{dB}$$

$$\text{magnitude in Pa} = \sqrt{(0.029)^2 + (0.007)^2} = 0.030\ \text{Pa}$$

$$\text{phase angle} = \text{atan}\left(\frac{0.011\ \text{Pa}}{0.005\ \text{Pa}}\right) = 65.6\ \text{dB}$$

C. Heyser Spiral in the time domain.

**Figure 6-8.** Using the Hilbert Transform of a function of time (convolution with $1/\pi t$) yields the imaginary (doublet) in the time domain.

### *Three Parameter Measurements*

In the arsenal of analysis today are the three parameter measurements where the resolution of two parameters is "smeared" to allow a conceptual view of what is occurring. We can choose to compromise the frequency magnitude resolution for higher time resolution, or we can compromise the time resolution for higher frequency magnitude resolution. The typical choice, because we have both ETC and EFC, energy frequency curve, for detailed individual views, is to smear both frequency and time resolutions for a compromise view of what some decaying frequency areas do over time. Typical display parameters are frequency on the horizontal scale, magnitude on the vertical scale, and time on the diagonal scale, Fig. 6-10.

While it always remains true that the reciprocal of the frequency bandwidth determines the time resolution and the reciprocal of the time window determines the frequency resolution, it is possible by "smearing" each parameter to gain an insight into the frequency vs. time behavior of a system, especially when some frequencies are longer in decaying than other frequencies.

$$\Delta f \times \Delta T \geq 1 \tag{6-3}$$

where,
$\Delta f$ is frequency resolution,
$\Delta T$ is time resolution.

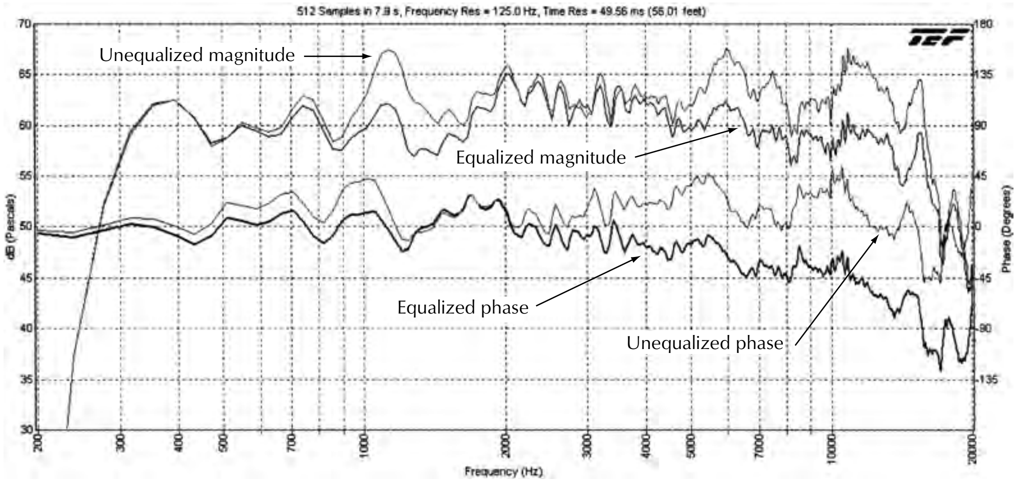



**Figure 6-9.** Equalized and unequalized transfer functions.

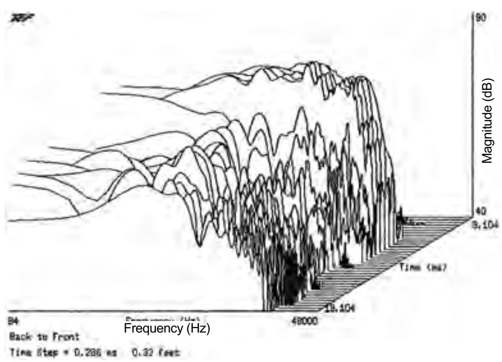
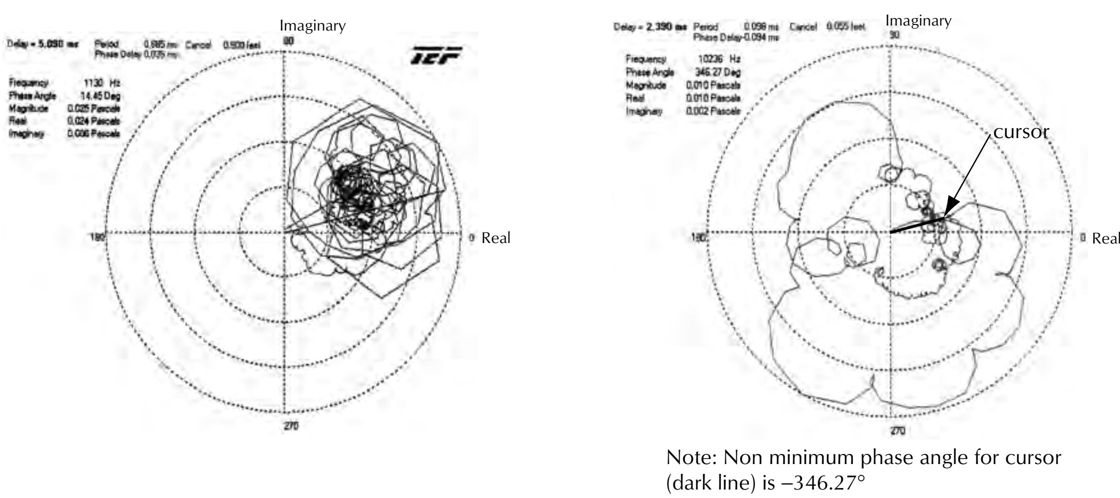



**Figure 6-10.** Three parameter measurement.

## *The Nyquist Plot*

A Nyquist plot provides simultaneously the real part, the imaginary part, the magnitude, the phase, the frequency and the identification of minimum and non-minimum phase. It is easily one of the most useful frequency domain measurements either electrically or acoustically. Modern analyzers provide cursor read out of the entire plot in addition to identifying non-minimum phase frequencies when they encircle the origin.




A. Minimum phase Nyquist plot.

B. Non-minimum phase Nyquist plot.

**Figure 6-11.** Minimum and non-minimum Nyquist plots.

The zero axis is the real component, the 90° axis the imaginary component, and the length from the origin to any chosen frequency on the plot is the magnitude component. Any range of frequency for which the Nyquist plot completely encircles the origin is a range of non-minimum phase behavior. The angle between the zero axis and the cursor set on a given frequency is the phase angle, Figs. 6-11A and 6-11B.

### *The Polar Envelope Time (PET) Plots*

The polar envelope time plot allows for any given point of measurement, instant values of:

1. The direction from which the reflection came.
2. The time of travel and distance.
3. The level.

### *History of Polar Time Measurements*

During WWII, Dr. Sidney Bertram developed a Sonar system for submarines named by its users as "Hells Bells." It consisted of a rotating hydrophone connected to an oscilloscope display through a bank of bandpass filters and associated electronics that displayed direction to target as an angle on the oscilloscope screen and the range to target as variable frequency sound. Close range—low bell-like tones, long range—higher bell-like tones. This system was used to put five U.S. submarines through a dense minefield into the Sea of Japan where they played an effective part in intercepting shipments from the Asian continent.

Today's system uses the input from six directional microphone measurements—forward, right, rear, left, and up and down. Farrel Becker developed PET for use with TEF analysis. It was after Farrel had programmed the software that a report from +30 years after WW II that we read in an IEEE journal gave recognition of Bertram's work.

The PET measurement is easily one of the most usable measurements ever devised for mapping reflections in architectural spaces. The use of a calibrated cursor gives precise distance, time, and bearing. One sound designer with deep experience in difficult acoustic environments, Deward Timothy of Poll Sound in Salt Lake City, uses the measurement in the orientation of arrays to minimize the detrimental reflections.

Each direction measured produces an individual ETC measurement. These are combined to produce the PET. Samples of each type of display are shown in Figs. 6-12A through 6-12F. To read a PET measurement, identify on the circumference the parameters for that plot, i.e., up, down, etc., or forward, back, left, right, etc.

The cursor can be placed on any dot on the screen and its length is the distance, its angle is the bearing, and its magnitude is read on the cursor printout. $L_D$ arrives first and its source is apparent as the shortest distance.

Figs. 6-12A and 6-12B, with the title "Flutter," are of a severe left-to-right flutter echo. The conventional ETC is one of the four used to make the Polar Time Plot, in this case the left ETC (microphone facing left).

Fig. 6-12C, "Room Acoustics—Front" is from the Intelligibility Workshop. There are three Polar Time Plots. The one with the floor values, first two text lines below the graph, set at 20.0 dB is in the horizontal plane and shows just the strongest reflections. The cluster of reflections in the forward direction, just above the origin, is from the rear wall of the orchestra shell.

Figs. 6-12D and 6-12E have the floor set at 24.0 dB and show more reflections. One is in the horizontal plane and the other is in the median plane. The reflections from the orchestra shell and rear wall of the auditorium are clearly seen in both plots.

Fig. 6-12F is a conventional ETC taken omnidirectionally and from the same location as the polar plots.

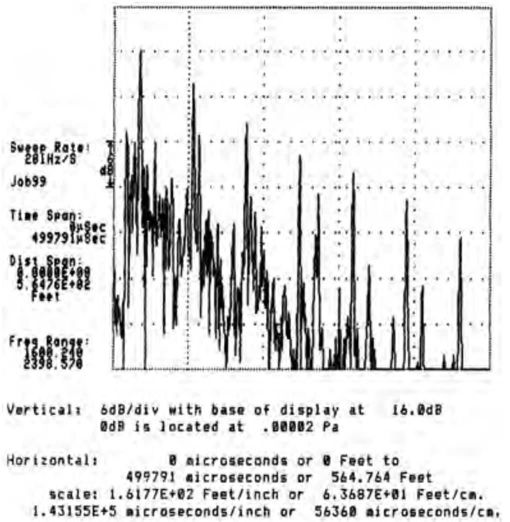
## 6.4 Site Surveys and Noise Criteria Curves

An important test that needs to be made at the site prior to building anything is a noise survey. This can be from a few minutes up to 24 hours. It consists of noise level analysis measurements, NLA, weighted consistent with the existing facts and the expected use of the building. Fig. 6-13A illustrates the variety of data that can be gathered in a one minute example NLA.

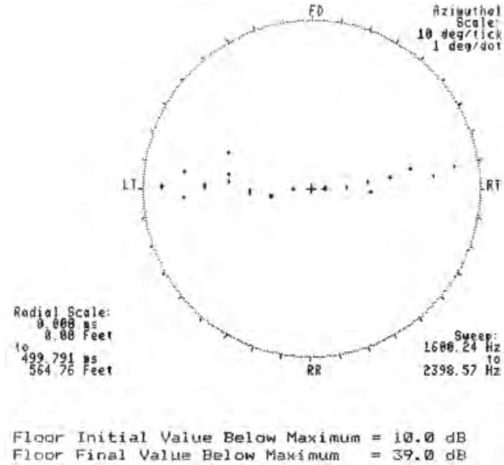
Coupled with such measurements should be the established noise criteria desired in order to estimate the required noise isolation the structure must provide, Fig. 6-13B, i.e., the difference between NLA levels and desired criteria.

Once the building is finished, the NC is measured for compliance with the chosen design criteria, Fig. 6-13C. The 2 kHz octave band is the one most often used for the *SNR* figure for $\%AL_{CONS}$ calculation.

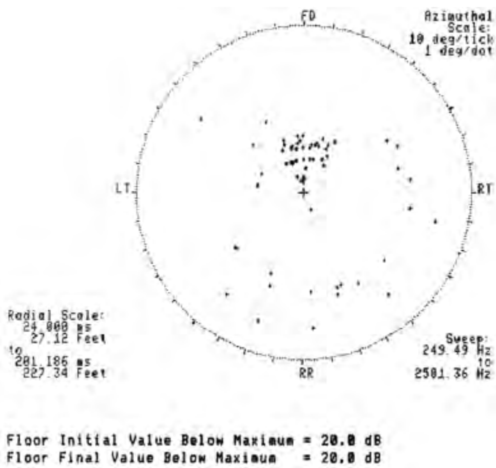
A very useful estimator of listener response is shown in Fig. 6-13D where the many variables that help shape human responses are considered and tabulated.



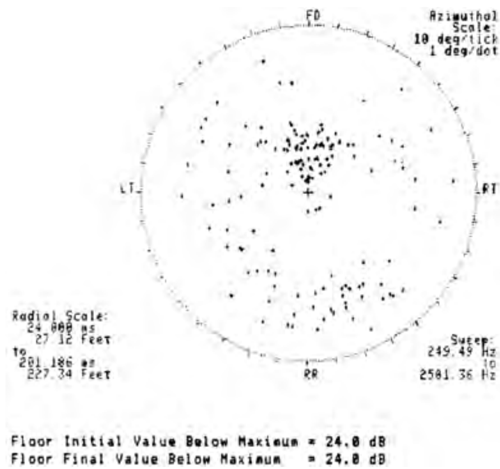

A. Flutter



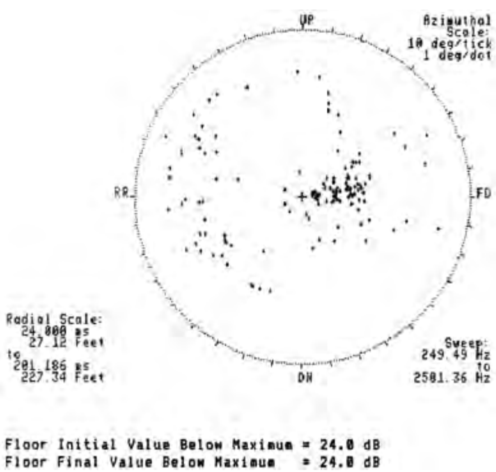

B. Flutter



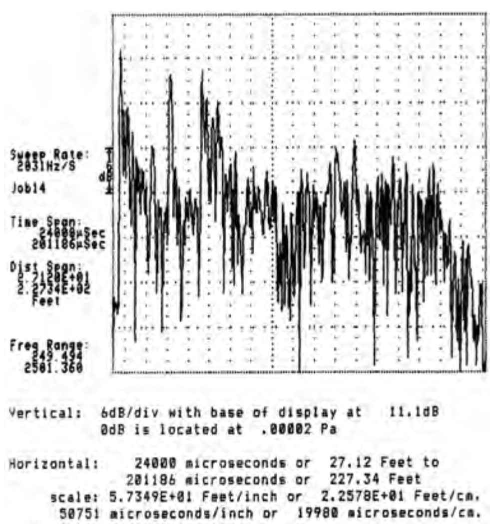

C. Room acoustics—front.




D. Room acoustics—front.

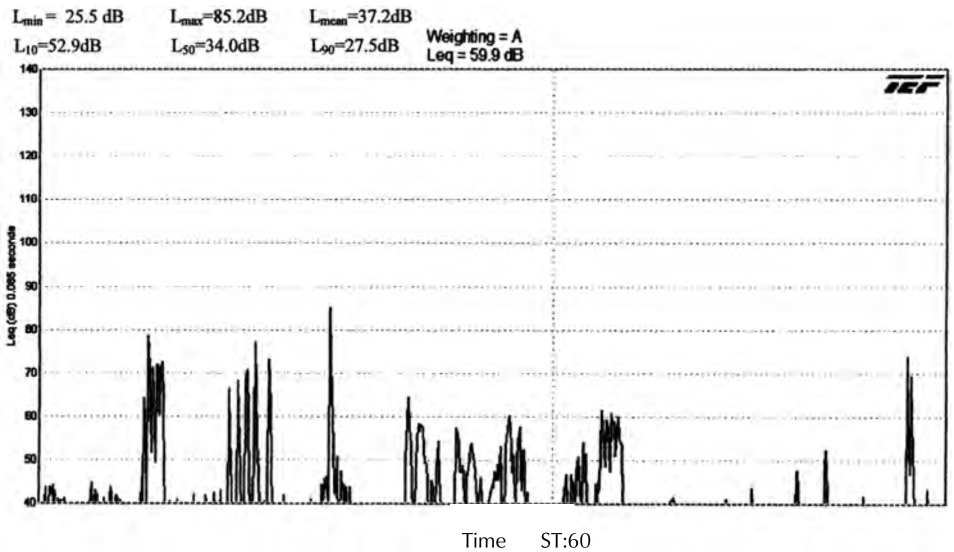



E. Room acoustics—front.




F. Conventional ETC.

**Figure 6-12.** PET measurements.

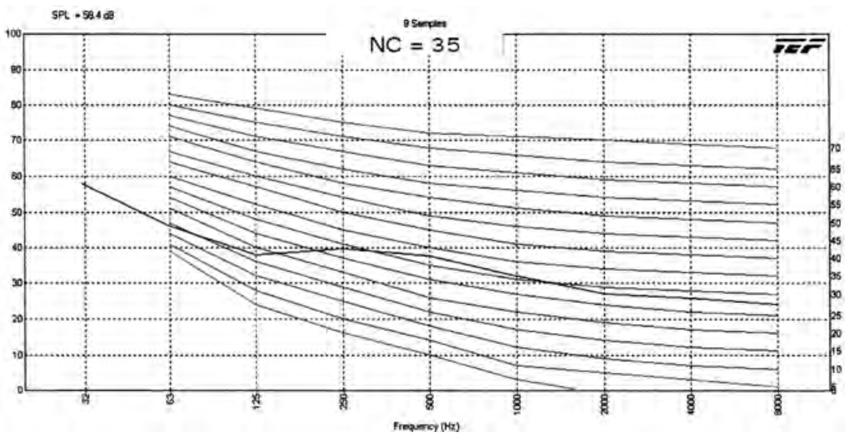



A. NLA sample.

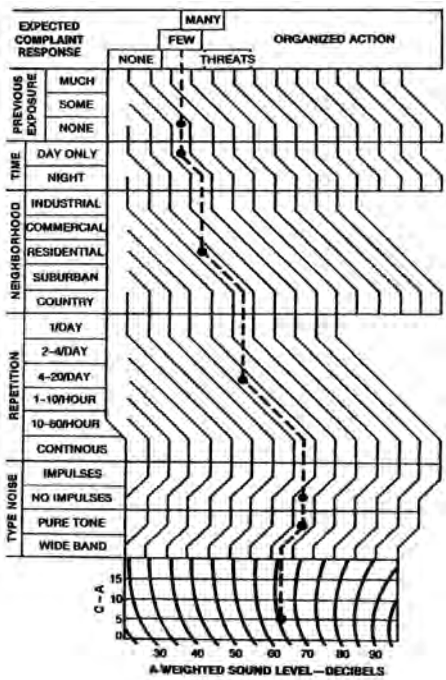
| | A Weighted | NC |
|---|---|---|
| **Residences** | | |
| Private homes (rural and suburban) | 25-35 | 20-30 |
| Private homes (urban) | 30-40 | 25-35 |
| Apartment houses, 2- and 3-family units | 35-45 | 30-40 |
| **Hotels** | | |
| Individual rooms or suites | 35-45 | 30-40 |
| Ball rooms, banquet rooms | 35-45 | 30-40 |
| Halls and corridors, lobbies | 40-50 | 35-45 |
| Garages | 45-55 | 40-50 |
| Kitchens and laundries | 45-55 | 40-50 |
| **Hospitals and Clinics** | | |
| Private rooms | 30-40 | 25-35 |
| Operating rooms, wards | 35-45 | 30-40 |
| Laboratories, halls and corridors, lobbies and waiting rooms | 40-50 | 35-45 |
| Washrooms and toilets | 45-55 | 40-50 |
| **Offices** | | |
| Board room | 25-35 | 20-30 |
| Conference rooms | 30-40 | 25-35 |
| Executive office | 35-45 | 30-40 |
| Supervisor office, reception | 35-40 | 30-45 |
| General open offices, drafting rooms | 40-55 | 35-50 |
| Halls and corridors | 40-55 | 35-55 |
| Tabulation and computation | 45-65 | 40-60 |
| **Auditoriums and Music Halls** | | |
| Concert and opera halls, studios for sound reproduction | 25-35 | 20-25 |
| Legitimate theaters, multipurpose halls | 30-40 | 25-30 |
| Movie theaters, TV audience studios, semi-outdoor amphitheaters, lecture halls, planetarium | 35-45 | 30-35 |
| Lobbies | 40-50 | 35-45 |
| **Churches and Schools** | | |
| Sanctuaries | 25-35 | 20-30 |
| Schools and classrooms | 35-45 | 30-40 |
| Laboratories | 40-50 | 35-45 |
| Recreation halls | 40-55 | 35-50 |
| Corridors and halls | 40-55 | 35-50 |
| Kitchens | 45-55 | 40-50 |
| **Public Buildings** | | |
| Public libraries, museums, court rooms | 35-45 | 30-40 |
| Post offices, general banking areas, lobbies | 40-50 | 35-45 |
| Washrooms and toilets | 45-55 | 40-50 |
| **Restaurants, cafeterias, lounges** | | |
| Restaurants | 40-50 | 35-45 |
| Cocktail lounges | 40-55 | 35-40 |
| Night clubs | 40-50 | 35-45 |
| Cafeterias | 45-55 | 40-50 |
| **Stores retail** | | |
| Clothing stores, department stores (upper floors) | 40-50 | 35-45 |
| Department stores (main floor), small retail stores | 45-55 | 40-50 |
| Supermarkets | 45-55 | 40-50 |
| **Sports activities—Indoor** | | |
| Coliseums | 35-45 | 30-40 |
| Bowling alleys, gymnasiums | 40-50 | 35-45 |
| Swimming pools | 45-60 | 40-55 |
| **Transportation (rail, bus, plane)** | | |
| Ticket sales offices | 35-45 | 30-40 |
| Lounges and waiting rooms | 40-55 | 35-50 |

B. Ranges of indoor design goals for air-conditioning system sound control

**Figure 6-13.** Noise criteria.




C. NC in residence.




D. Annoyance of neighborhood sound levels.

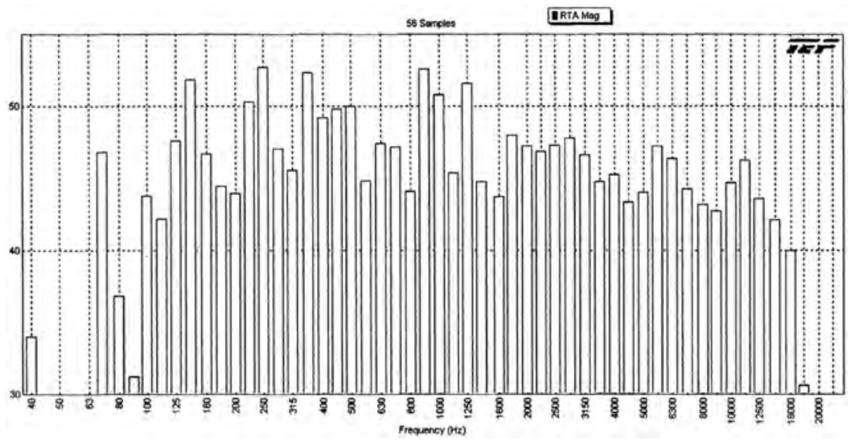
**Figure 6-13. (continued)** Noise criteria.

The most commonly encountered violation measured is a too-noisy HVAC system. Often balancing of the HVAC can provide the difference between "fail" and "pass." Because speech intelligibility is directly dependent upon *SNR* failure to specify correct noise criteria, and further failure to measure the violation can doom an otherwise successful sound system installation.

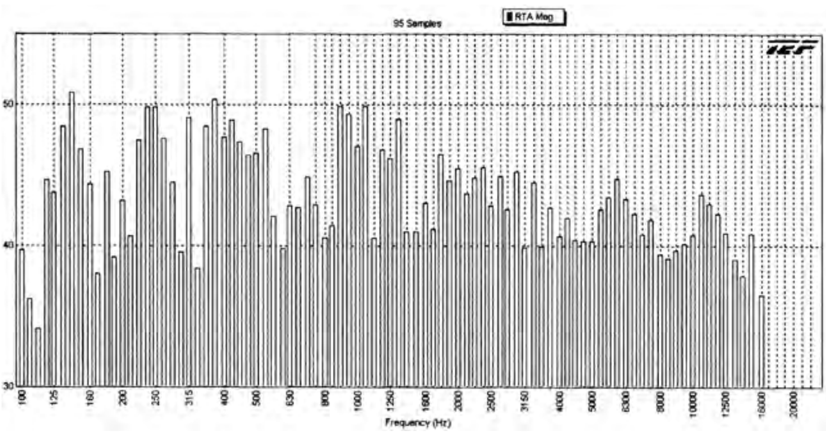
NC curves are plotted in 1/1 octave bands. They allow, at a glance, a comparison of the acoustic response at a listener from the sound system to the NC value for the signal-to-noise evaluation. Criteria exist for most common applications.

### *Constant Percentage Bandwidth Analysis*

Constant percentage bandwidth analyzers are widely used. One-third of an octave, 23% of center frequency, one-sixth of an octave, 11.5% of center frequency, or one-twelfth of an octave, 5.76% of center frequency filters allow essentially "real time" analysis of non-stationary signals. (See *Section 6.7 Fractional Bandwidth Filter Analyzers.*) One of the authors was instrumental in seeing the first 1/3 octave filter analyzers come to the market in 1968. The term "third octave" is often employed for 1/3 of an octave but is incorrect as it describes a filter for




A. 1/6 octave RTA display.




B. 1/12 octave RTA display.

**Figure 6-14.** RTA displays.

every third octave. See Fig. 6-14A for a ⅙ octave display and Fig. 6-14B for a 1⁄12 octave RTA display.

## 6.5 An Improper Use of Real Time Analysis

Constant percentage bandwidth filters have absolute widths that increase in direct proportion to the center frequency of the filter. When performing spectrum analysis with instruments based on such filters it is necessary to employ a random noise source whose spectrum has constant energy per octave, i.e., pink noise as opposed to a noise source that has constant energy per unit bandwidth, i.e., white noise.

A system possessing a uniform or flat response on a per unit bandwidth basis that is excited with pink noise will produce a flat display on a constant percentage bandwidth analyzer. Such a system excited with white noise would produce a response that rises at 3 dB/octave on a constant percentage bandwidth analyzer. Therefore, when constant percentage bandwidth analyzers are employed to study the spectra of program material where it is desired to determine the response displayed on a per unit bandwidth basis, it is necessary to precede such an analyzer by a filter that has a response that falls at the rate of 3 dB/octave. Any evaluation of program material without such a device is invalid.

It is the authors' belief that this uncorrected error is why so many professional mixing engineers still use meters and indicators in place of the much more useful real time analyzer. Trained ears didn't agree with the uncorrected visual display. The noise control people made their *criteria* constant percentage bandwidth based, thereby judging relative results. The recording engineers, home hi-fi enthusiasts, and other researchers did not realize the need and therefore failed to compensate for it.

## 6.6 Evaluation of Listener Response

Measurements that correlate well with listener response are those with a frequency resolution approximating "critical bandwidths." This falls between ⅓ and ⅙ of an octave, Fig. 6-15. One of the authors once walked into a church auditorium to be told by the organist that it was the "deadest" church acoustically he had ever played in, followed a minute later when he met the minister who stated it was "too live" to preach in. Measurements of the reverberation time vs. frequency revealed that something was passing low frequencies from the room but the mid-band frequencies were excessively reverberant. The low frequencies were being diaphragmatically absorbed by the paneling in the ceiling which, when braced, allowed the bass to remain in the room to the satisfaction of the organist. Absorptive treatment on the rear walls and seat cushions controlled the mid-frequency excess to the satisfaction of the minister.

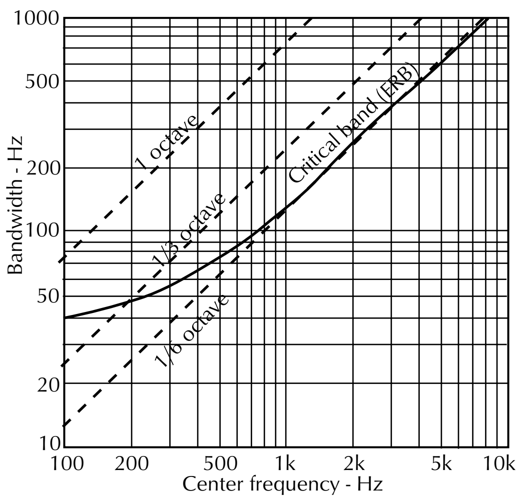



**Figure 6-15.** A plot of critical bandwidths of the human auditory system compared to constant percentage bandwidths of filter sets commonly used in acoustical measurements.

Listener response can vary drastically due to the pinnae response of the individual. For a uniform (flat) frequency response to the ear, these pinnae responses are measured at the individual's eardrum. The combined ear canal resonance and the "comb filtering" of the folds in the pinnae give the resulting responses. Listener response is important because in existing structures, the complaints become microphone positions for measurements that need to be made, Fig. 6-16.

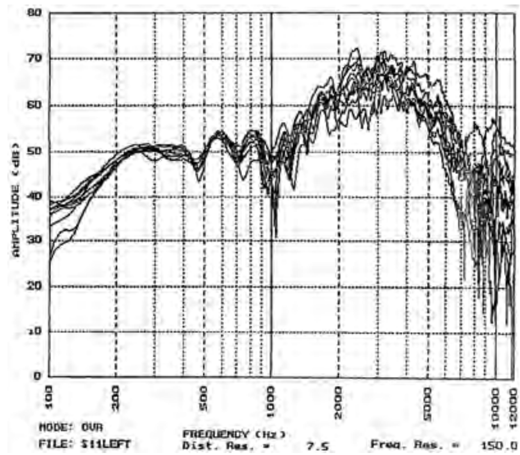



**Figure 6-16.** Pinnae responses.

The science and art of measurement begin in the brain of the measurer and the apparatus either confirms the hypothesis or provides the opportunity for serendipity to lead thought in a new direction.

## 6.7 Fractional Bandwidth Filter Analyzers

Fractional bandwidth filter analyzers are the most commonly used; therefore, we have included their mathematical structure for evaluation of frequency spacing, frequency resolution, and frequency labels from ⅟₁ to ⅟₂₄ octave equivalents and $10^{0.3}$ to $10^{0.0125}$ decades. For example, series 80 increments yield series 40 upper and lower crossover frequencies between center frequencies in the 40 series.

The coincidence of ⅓ octaves and ⅒ decades in value has led to wide use of decade filters in both equalizers and analyzers.

1. $2^{1/3}$(1.259921050)
2. $10^{0.1}$(1.258925412)

difference = 0.000995638

$10^{0.1}$ was named the "10" Series because

$$\frac{1}{0.1} = 10 \tag{6-4}$$

There are 10 bands per decade.

Band numbers were $N = 0$ and up ($10^0 = 1.0$). This allowed a very simple equation to be written that would serve for any series and any $N$ in decade intervals. From this equation, any center frequency $f_C$ could then be easily calculated.

$$f_C = 10^{0.1^N} \tag{6-5}$$

simplified to:

$$10^{\frac{N}{series}} \tag{6-6}$$

### *Series*

If the reciprocal of 0.1 is 10 Series, then the ⅟1 octave equivalent in decades would be 10/3 = $3^{1/3}$ Series (because there are three ⅒ decade $f_C$s in the ⅟1 octave decade equivalent). That makes the decade $10^{0.3}$ for the ⅟1 octave equivalent. A ⅔ octave decade equivalent would become ⅔(0.3) = $100^{0.2}$. We can now write out for the most used equivalents.

| Octave Equiv. | Decade | Series | *N* = 1.0 |
|---|---|---|---|
| ⅟1 | $10^{0.3}$ | 3⅓ | 1.995262 |
| ⅔ | ⅔ × 0.3 = $10^{0.2}$ | ⅟0.2 = 5 | 1.584893 |
| ½ | ½ × 0.3 = $10^{0.15}$ | ⅟0.15 = 6⅔ | 1.412538 |
| ⅓ | ⅓ × 0.3 = $10^{0.1}$ | ⅟0.1 = 10 | 1.258925 |
| ⅙ | ⅙ × 0.3 = $10^{0.05}$ | ⅟0.05 = 20 | 1.220180 |
| ⅟12 | ⅟12 × 0.3 = $10^{0.025}$ | ⅟0.025 = 40 | 1.059254 |

The series tells how many *N*s are in one decade (i.e., *N* = 1 to *N* for the first decade). The *N*s then repeat with only the decimal point moved to the right for each higher decade.

### *Other Uses*

The pass band for filters can be found (for the −3 dB power points) by:

$$f_{xU} = 10^{\frac{N+0.5}{series}} \tag{6-7}$$

and

$$f_{xL} = 10^{\frac{N-0.5}{series}} \tag{6-8}$$

$$f_{xU} - f_{xL} = \text{Bandwidth in Hz}$$

where,
$f_{xU}$ is the upper −3 dB point,
$f_{xL}$ is the lower −3 dB point.

The *Q* of the filter is found by:

$$Q = \frac{f_C}{BW} \tag{6-9}$$

where,
$f_C$ is the center frequency of the filter.

The percent bandwidth is found by calculating the bandwidth for 100 Hz. For example:

$$N = series \times \log f_C \tag{6-10}$$

$$N = 10\log 100 = 20$$

$$\left(10^{\frac{20+0.5}{10}}\right) - \left(10^{\frac{20-0.5}{10}}\right) = 23\%$$

$$Q = \frac{100}{23.076752} = 4.33\ .$$

### *Useful Tools*

To rapidly calculate for any series all $f_{xU}$, $f_C$, and $f_{xL}$ you can repeatedly multiply:

$$f_{xU}, f_C, f_{xL} = 10^{\frac{N-0.5}{series}} \tag{6-11}$$

The value for *N* = ½ is the first $f_{xL}$. The next number is *N* = 1$f_C$. The next number is $f_{xU}$ for N = 1 and $f_{xL}$ for *N* = 2. Again, the next number is $f_C$ for *N* = 2, etc. Having thus calculated all the $f_x$s you can extract all the bandwidths.

### *Using the Decade Exponents*

Writing out again the decade exponents we find that by using the ⅒ decade bandwidths we can multiply that bandwidth by 10 times the exponent to obtain the other bandwidths and percentages.

| | |
|---|---|
| $10^{0.3}$ | 3 × 23.029 = 69.087 |
| $10^{0.2}$ | 2 × 23.029 = 46.085 |
| $10^{0.15}$ | 1.5 × 23.029 = 35.544 |
| $10^{0.1}$ | 1.0 × 23.029 = 23.029 |
| $10^{0.05}$ | 0.5 × 23.029 = 11.515 |
| $10^{0.025}$ | 0.25 × 23.029 = 5.757 |

For values of *Q* we can take the 100 Hz/*BW* for the series 10 and again multiplying the exponents by 10 we can divide them into the series 10 value to obtain all the other *Q* values.

The difference in power bandwidths expressed in decibels is

| | |
|---|---|
| $10^{0.3}$ | 4.342/3.0 = 1.447 |
| $10^{0.2}$ | 4.342/2.0 = 2.171 |
| $10^{0.15}$ | 4.342/1.5 = 2.895 |
| $10^{0.1}$ | 4.342/1.0 = 4.342 |
| $10^{0.05}$ | 4.342/0.5 = 8.685 |
| $10^{0.025}$ | 4.342/0.25 = 17.369 |

$$dB = 10\log\frac{Highest\ frequency\ bandwidth}{Lowest\ frequency\ bandwidth} \quad (6\text{-}12)$$

Also band 43–13 is 30 bands or 30 dB in the case of series 10.

### *Decibels*

Once again, using series 10 every step in the repeated multiplication of

$$10^{\frac{N-0.5}{10}} \times 10^{\frac{N-0.5}{10}} \times \ldots$$

is a 1.0 dB step on the 20log scale. Every other step (i.e., the $f_C$s) is 1.0 dB on the 10log scale.

### *Label Frequencies*

The equations discussed in this article are for the exact frequencies you would use as center frequencies and crossover frequencies for equalizers and analyzers. In actual practice the center frequencies are labeled in a simplified manner. Table 6-1 outlines these labels for the ¹⁄₁, ½, ⅔, ⅓, ⅙, and ¹⁄₁₂ devices in use today.

**Table 6-1.** Frequency Labels for Audio Components

| ¹⁄₁₂ Octave 40 Series | ⅙ Octave 20 Series | ⅓ Octave 10 Series | ½ Octave 6⅔ Series | ⅔ Octave 5 Series | ¹⁄₁ Octave 3⅓ Series | Exact Value |
|---|---|---|---|---|---|---|
| 1.0 | 1.0 | 1.0 | 1.0 | 1.0 | 1.0 | 1.000000000 |
| 1.06 | | | | | | 1.059253725 |
| 1.12 | 1.12 | | | | | 1.122018454 |
| 1.18 | | | | | | 1.188502227 |
| 1.25 | 1.25 | 1.25 | | | | 1.258925411 |
| 1.32 | | | | | | 1.333521431 |
| 1.4 | 1.4 | | 1.4 | | | 1.412537543 |
| 1.5 | | | | | | 1.496235654 |
| 1.6 | 1.6 | 1.6 | | 1.6 | | 1.584893190 |
| 1.7 | | | | | | 1.678804015 |
| 1.8 | 1.8 | | | | | 1.778279406 |
| 1.9 | | | | | | 1.883649085 |
| 2.0 | 2.0 | 2.0 | 2.0 | | 2.0 | 1.995262310 |
| 2.12 | | | | | | 2.113489034 |
| 2.24 | 2.24 | | | | | 2.238721132 |
| 2.36 | | | | | | 2.371373698 |
| 2.5 | 2.5 | 2.5 | | 2.5 | | 2.511886423 |
| 2.65 | | | | | | 2.660725050 |
| 2.8 | 2.8 | | 2.8 | | | 2.818382920 |
| 3.0 | | | | | | 2.985382606 |
| 3.15 | 3.15 | 3.15 | | | | 3.162277646 |
| 3.35 | | | | | | 3.349654376 |
| 3.55 | 3.55 | | | | | 3.548133875 |
| 3.75 | | | | | | 3.758374024 |
| 4.0 | 4.0 | 4.0 | 4.0 | 4.0 | 4.0 | 3.981071685 |
| 4.25 | | | | | | 4.216965012 |
| 4.5 | 4.5 | | | | | 4.466835897 |
| 4.75 | | | | | | 4.731512563 |

**Table 6-1.** (continued) Frequency Labels for Audio Components

| ⅟12 Octave 40 Series | ⅙ Octave 20 Series | ⅓ Octave 10 Series | ½ Octave 6⅔ Series | ⅔ Octave 5 Series | ⅟1 Octave 3⅓ Series | Exact Value |
|---|---|---|---|---|---|---|
| 5.0 | 5.0 | 5.0 | | | | 5.011872307 |
| 5.3 | | | | | | 5.308844410 |
| 5.6 | 5.6 | | 5.6 | | | 5.623413217 |
| 6.0 | | | | | | 5.956621397 |
| 6.3 | 6.3 | 6.3 | | 6.3 | | 6.309573403 |
| 6.7 | | | | | | 6.683439130 |
| 7.1 | 7.1 | | | | | 7.079457794 |
| 7.5 | | | | | | 7.498942039 |
| 8.0 | 8.0 | 8.0 | 8.0 | | 8.0 | 7.943282288 |
| 8.5 | | | | | | 8.413951352 |
| 9.0 | 9.0 | | | | | 8.912509312 |
| 9.5 | | | | | | 9.440608688 |

**Table 6-2.** 80 Series $f_C$s

| $10^{\left(\frac{0.5}{40}\right)^{N = 1\text{–}20}}$ | $10^{\left(\frac{0.5}{40}\right)^{N = 21\text{–}40}}$ | $10^{\left(\frac{0.5}{40}\right)^{N = 41\text{–}60}}$ | $10^{\left(\frac{0.5}{40}\right)^{N = 61\text{–}80}}$ |
|---|---|---|---|
| 1.0292005272 | 1.8302061063 | 3.2546178350 | 5.7876198835 |
| 1.0592537252 | 1.8836490895 | 3.3496543916 | 5.9566214353 |
| 1.0901844924 | 1.9386526360 | 3.4474660657 | 6.1305579215 |
| 1.1220184543 | 1.9952623150 | 3.5481338923 | 6.3095734448 |
| 1.1547819847 | 2.0535250265 | 3.6517412725 | 6.4938163158 |
| 1.1885022274 | 2.1134890398 | 3.7583740429 | 6.6834391757 |
| 1.2232071190 | 2.1752040340 | 3.8681205463 | 6.8785991231 |
| 1.2589254118 | 2.2387211386 | 3.9810717055 | 7.0794578438 |
| 1.2956866975 | 2.3040929761 | 4.0973210981 | 7.2861817451 |
| 1.3335214322 | 2.3713737057 | 4.2169650343 | 7.4989420933 |
| 1.3724609610 | 2.4406190680 | 4.3401026364 | 7.7179151559 |
| 1.4125375446 | 2.5118864315 | 4.4668359215 | 7.9432823472 |
| 1.4537843856 | 2.5852348396 | 4.5972698853 | 8.1752303794 |
| 1.4962356561 | 2.6607250598 | 4.7315125896 | 8.4139514165 |
| 1.5399265261 | 2.7384196343 | 4.8696752517 | 8.6596432336 |
| 1.5848931925 | 2.8183829313 | 5.0118723363 | 8.9125093813 |
| 1.6311729092 | 2.9006811987 | 5.1582216507 | 9.1727593539 |
| 1.6788040181 | 2.9853826189 | 5.3088444423 | 9.4406087629 |
| 1.7278259805 | 3.0725573653 | 5.4638654988 | 9.7162795158 |
| 1.7782794100 | 3.1622776602 | 5.6234132519 | 10.0000000000 |

Table 6-2 is the first decade for all the filters discussed and includes all center frequencies and cross-over frequencies. As a final bonus, from the one exponent, each step is ⅛ dB on the 10log scale and ¼ dB the 20log scale.

1. All even numbers are the ⅟12 octave center frequencies.
2. Every 4th number is a ⅙ octave center frequency.
3. Every 8th number is ⅓ octave center frequency.
4. Every 12th number is a ½ octave center frequency.
5. Every 16th number is a ⅔ octave center frequency.
6. Every 24th number is a ⅟1 octave center frequency.

7. All odd numbers are 1⁄12 octave crossover frequencies.
8. Every other 1⁄12 octave center frequency is a 1⁄6 octave crossover frequency.
9. Every other 1⁄6 octave center frequency is a 1⁄3 octave crossover frequency.
10. Every other 1⁄3 octave center frequency is a 1⁄2 octave crossover frequency.
11. Every other 1⁄2 octave center frequency is a 1⁄1 octave crossover frequency.
12. 2⁄3 octave crossover frequencies start at the sixth line and every 12th line thereafter.

## 6.8 Conclusion

Acoustic tests are a mental exercise assisted by measured hints. It's when measurements don't agree with the trained ear that a chance for serendipity is at hand.

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