

S E C O N D E D I T I O N

SOUND REINFORCEMENT HANDBOOK

Written For Yamaha By
Gary Davis & Ralph Jones

YAMAHA®

THE SOUND REINFORCEMENT Handbook

SECOND EDITION

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Published by

HL *Hal Leonard Publishing Corporation*

7777 W. Bluemound Road, P.O. Box 13819
Milwaukee, WI 53213

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ISBN: 0-88188-900-8

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First Edition	First Printing:	December, 1987
	Second Printing:	February, 1989
	Third Printing:	April, 1989
Second Edition	First Printing:	June, 1989
	Second Printing:	February, 1990

Printed in the U.S.A.

PREFACE

In 1974, the people at Yamaha asked me to write some spec sheets for a dozen or so new guitar amplifiers and small mixers. As soon as that job was done, they showed me a new product. It was a console, one intended to be a major departure – a leap into the heart of the professional sound reinforcement market – for a company then known primarily as a musical instrument manufacturer (or a motorcycle manufacturer, if you were not a musician). Yamaha said they wanted to firmly establish themselves as a leader, and they felt that a comprehensive instruction manual for the new console would help do the job. We had only a prototype board and some sketchy specs, but no detailed drawings or data, so I contacted engineer John Windt, and we measured the performance. Working with pencil and portable typewriter, I created the first draft of the PM-1000 Operating Manual.

The PM-1000 console did put Yamaha on the map, insofar as the professional sound reinforcement business is concerned. The manual was so popular that it had to be reprinted many times (far exceeding the number of consoles built). It became a standard text in several college courses due to its detailed discussions of the basics of sound reinforcement.

During the ensuing decade, Yamaha asked me to write manuals for a variety of amps, signal processors, mixers and so forth, and to maintain the same standard of excellence we began with the PM-1000 manual. Unfortunately, it is very expensive to produce and print 30 to 60 pages of instructions for every product, and it is difficult to justify – especially when only 8 to 16 pages contain the “hard core” operational data. For this reason, both Yamaha and I had pondered the concept of a generalized guide for sound reinforcement, apart from any individual operating manuals.

Finally, 10 years after the PM-1000 manual was published, Yamaha decided to move forward with this project. We anticipated a book of 96 to 160 pages in length, and expected it would take a year to complete. The first edition of *The Sound Reinforcement Handbook* was comprised of 384 pages, some 256 illustrations, and took three years to complete. The first printing of 10,000 copies sold out in a year, and while we were completing this second edition, another two runs totalling nearly 5,000 more copies had to be printed. Clearly, the book has been well received, and we're grateful.

The Sound Reinforcement Handbook is the largest project I have ever undertaken. It would not have been possible without the support and patience of the people at Yamaha Corporation of America, and at their parent company, Yamaha Corporation of Japan. All of us were able to endure the strain of gathering data, editing it, and producing this book for one reason: we had an important goal to achieve.

That goal was to create a useful reference for anyone who is interested in learning the basics of sound reinforcement. In this second edition, we have improved many of the illustrations, and added entirely new topics, corrected a number of minor errors and typos (perhaps created new ones). We have made the book much easier to use by completely reorganizing the chapters and creating an index.

We had originally planned the first edition for looseleaf binding, with the intent to publish updates periodically. The cost, however, would have been prohibitive, and so this soft cover format was adopted, with the prospect of a new edition when the need arises. We all hope you enjoy and learn from this one.

Gary D. Davis

Santa Monica, California

June, 1989

THIS HANDBOOK IS DEDICATED TO THE SOUND REINFORCEMENT INDUSTRY,
AND TO ALL THOSE PEOPLE WHO HAVE WORKED SO HARD TO BRING
BETTER SOUND AND MUSIC TO THE WORLD. WE PARTICULARLY WISH TO
ACKNOWLEDGE THE LATE DEANE JENSEN FOR HIS ADVANCEMENT OF SOUND.

SECTION 2. FREQUENCY RESPONSE

2.1 A DEFINITION

One of the most commonly-used terms in audio is *frequency response*. In Section 2, we define this term, examine how it applies to various types of audio equipment, and describe the relationship between frequency response and program material.

The frequency response of a device describes the relationship between the device's input and output with regard to signal frequency and amplitude. Another term for this is magnitude response. In its most common usage, the frequency response describes the usable range of signal frequencies which the device will pass from input to output.

Consider the system shown in Figure 2-1. An unknown signal processing element (or black box) is fed from a variable-frequency sine wave generator, and a device for indicating sine wave level in decibels is connected to its output.

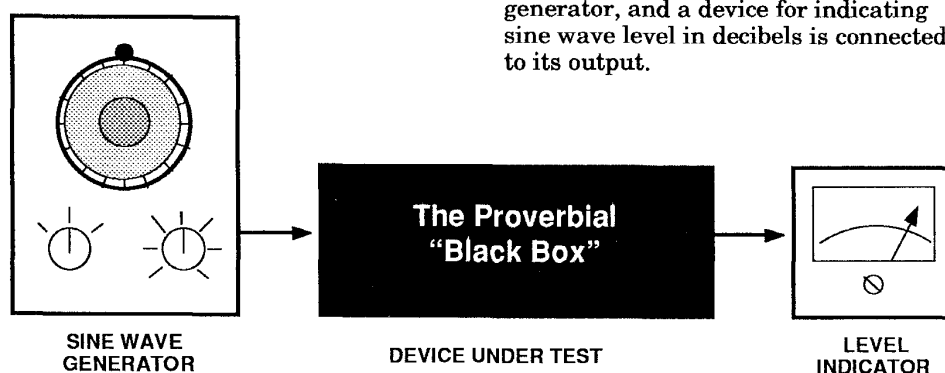


Figure 2-1. Model for measuring frequency response

The ideal oscillator produces the same output level at all frequencies (in reality the linearity is not always perfect), so the input level to the black

box is constant. As we sweep the oscillator frequency throughout the audio range, however, we may see that the output level of the black box, as registered by our meter, changes. If we make note of the level at each frequency on a graph, we can produce a chart like that shown in Figure 2-2. Here, the output level (vertical axis) is plotted against frequency (horizontal axis).

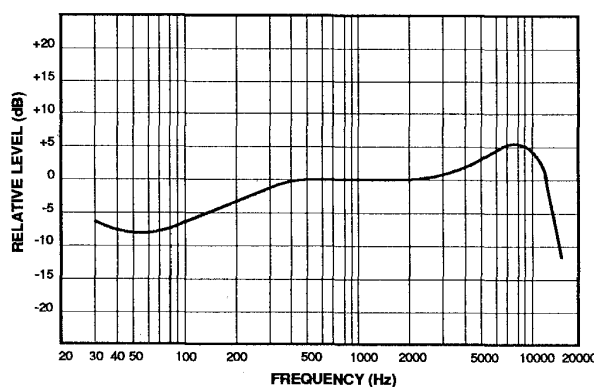


Figure 2-2. Plotted frequency response
(relative level versus frequency)
of "Black Box" in Figure 2-1

2.1.1 Basic Specification Methods

The graph of Figure 2-2 is called a frequency response plot. It shows us the range of frequencies that the black box will pass from input to output, and what fluctuations in output level (if any) occur within that range.

It is important to understand that the frequency response plot assumes a constant input level to the device under test. For precisely this reason, it gives us an indication of the fidelity with which the device transfers a signal from input to output. The less deviation there is in output level across the stated frequency band, the more faithfully the signal at the output will reflect the signal at the input.

NOTES:

1. If the input to a device under test is not constant in level at all frequencies, a correction may be made to the output plot (or the input level can be intentionally varied to make it constant). The resulting plot is said to be "normalized."
2. The term "frequency response" is used only to refer to signal processing devices and transducers — that is, any devices through which a signal passes. When referring to signal generating devices (oscillators, musical instruments, and so on), the proper term to use is "frequency range."

In its simplest form, a typical frequency response specification might read:

FREQUENCY RESPONSE:
30 Hz to 18 kHz, ± 3 dB

Note that the range (from 30 Hz to 18 kHz) is accompanied by the qualifier, "+ or - 3 dB." This is called the tolerance of the specification. The tolerance tells us the maximum deviation in output level that we can expect over the stated range if the input level remains the same at all frequencies.

Without a stated tolerance, the frequency response specification is useless, since we are left to guess the unit's effect on the signal. There might, in fact, be horrendous peaks and/or dips in the response — and these could alter the signal considerably. While a tolerance of ± 3 dB is often assumed if not specified, such an assumption is at your own risk... there is usually a reason why such an important qualifier is omitted from a specification!

Figure 2-3 shows how such a specification is derived from the frequency response plot.

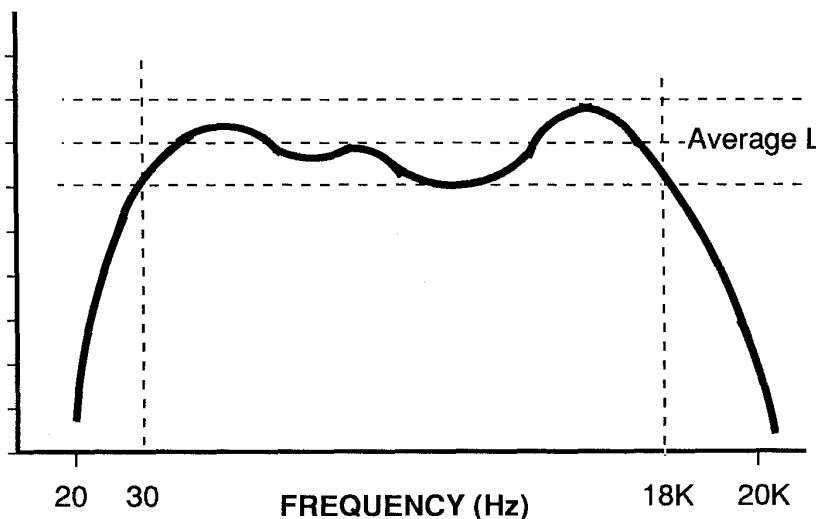


Figure 2-3. Derivation of a frequency response specification from a plot

FREQUENCY RESPONSE

Some audio devices exhibit extraordinarily flat frequency response, as shown in Figure 2-4. Such a response curve would be characteristic of a power amplifier or line amplifier, for example.

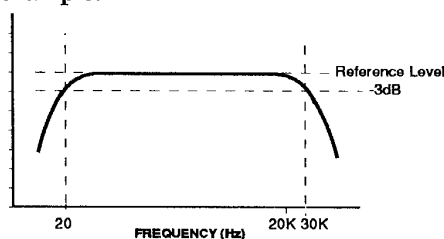


Figure 2-4. A plot of "flat" frequency response

In such cases, the frequency response limits are usually taken to be those points where the unit's output is 3 dB below the average (reference) level. The frequency response specification for the device plotted in Figure 2-4, then, would read:

FREQUENCY RESPONSE:
20 Hz to 30 kHz, +0, -3 dB.

If the frequency response of the device in question greatly exceeds the total range of human hearing (20 Hz to 20 kHz), the frequency response may be specified by simply stating the total response deviation within the audible range. The specification for the device whose response is shown in Figure 2-5, for example, might read:

FREQUENCY RESPONSE:
20 Hz to 20 kHz, +0, -1 dB.

Alternately, if one chooses the -3 dB points as the frequency limits, the specification would read:

FREQUENCY RESPONSE:
10 Hz to 40 kHz, +0, -3 dB.

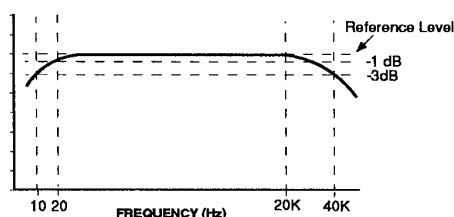


Figure 2-5. A "flat" frequency response plot of a device with a very wide bandwidth

2.1.2 Octave Relationships And Measurements

Frequency response measurements and specifications are sometimes made on an octave or $1/3$ -octave basis. Such measurements are far lower in resolution than those described above. They are widely used with loudspeakers, however, because they provide good correlation with the characteristics of human hearing (and can be adjusted by corresponding one octave or $1/3$ -octave equalizers or filter sets).

The octave is a special musical interval between two tones, formed when the ratio between the frequencies of the tones is 2:1. The ear hears an equivalence between the tones when this is the case (for this reason, pitches at successive octaves in the musical scale are named with the same letter).

With regard to frequency, the octave interval is much wider at high frequencies than at low frequencies. For example, one octave above 40 Hz is 80 Hz (a 40 cycle wide interval), but an octave above 1,000 Hz is 2,000 Hz (a 1,000 cycle wide interval). Yet we hear both of these intervals as musically similar!

This is because the ear's response to frequency is logarithmic in character. (Logarithms are discussed in the Appendix of this handbook). Accordingly, audio response plots such as those shown earlier in this section employ a logarithmic scale to indicate frequency. On this type of scale, the divisions are packed closer and closer together as the frequency increases... up to a decade (10, 100, 1000, etc.), then the spacing interval repeats. Octave and $1/3$ -octave measurements are designed to divide the linear audio frequency scale, which we do not perceive to be linear, into perceptually equal increments. In frequency response measurements and specifications, then, the average level in each incremental division is given — normally in bar graph form, as shown in Figure 2-6 (next page).

Here, the center frequencies of the bands are spaced $1/3$ octave apart. The specific frequencies given on the horizontal axis are standard center frequencies for $1/3$ -octave measurements: these same centers are also used for the filters in $1/3$ -octave graphic equalizers. The standard used is known as "ISO" (International Standards Organization), so you know if you have an equalizer with ISO

2.2 FREQUENCY RESPONSE OF PRACTICAL AUDIO DEVICES

standard octave or $1/3$ -octave spacing, it will coincide with the bands measured on an ISO standard Real Time Analyzer.

Instead of a sine wave, the excitation signal used for measurements like this is a special signal called pink noise. Pink noise is a randomly-generated signal that excites all the audio frequencies with equal energy per octave. It sounds very much like a rushing waterfall.

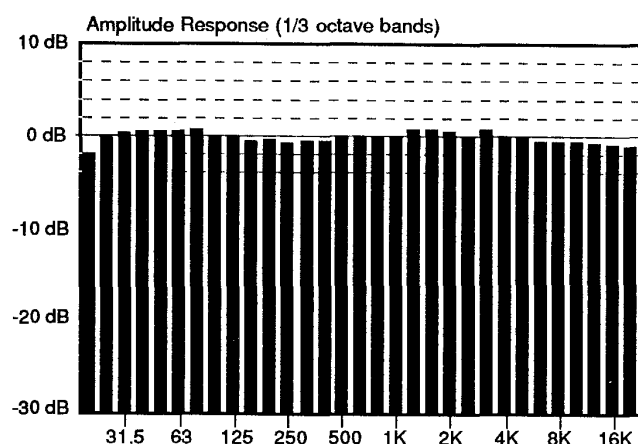


Figure 2-6. Frequency response measured in $1/3$ -octave bands

Since there are more frequencies packed into each upper octave than each lower octave, pink noise rolls off at higher frequencies. This means there is less energy at any single high frequency than at any single low frequency so that when all the noise in an octave is added up, it equals all the noise in any other octave. When pink noise is used as the excitation signal, the output of the device under test is filtered in octave or $1/3$ -octave bands, and the average energy in each band is measured separately.

It is important to note that measurements like this can conceal as much as they reveal. While they provide a good picture of the general characteristic of a device's frequency response, very narrow peaks or dips may be completely missed by the technique. In a reverberant space, the measurement may say more about the room than about the loudspeaker under test.

A practical audio system is a network of many different types of devices through which the signal must pass on its way to the listeners' ears. Each stage in the system will have specific frequency response characteristics, and will therefore modify the signal to

some degree. The overall frequency response of the system is a function of the combined responses of all the various elements in the chain.

The audio elements that exhibit the flattest frequency response characteristics are electronic circuits and cables.

2.2.1 Electronic Circuits And Cables

Cables are electrically the simplest possible elements in the audio system, and this fact accounts for their generally good response. Cables may introduce response problems at the frequency extremes, however, and the extent of such problems is determined by the design of the cable, its length, and the type of circuitry to which it is connected.

The typical audio cable consists of one or two signal conductors surrounded by insulation and a shielding conductor, as shown in Figure 2-7.

The audio signal is carried in the central conductor(s). The shielding conductor is connected to the electrical ground point in the circuit, and serves to catch most induced static or radio interference signals, shunting them off to ground. (Hum, which is often caused by electromagnetic interference, is not generally attenuated by shields unless

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they are in the form of iron conduit; instead, twisting the dual conductors in a balanced circuit is the best means to reduce hum. This is due to a characteristic known as Common Mode Rejection, which is explained in Section 11.6.)

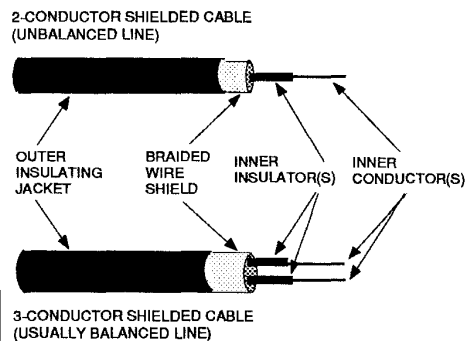


Figure 2-7. Construction of typical audio cables

NOTE: The next paragraph may be a bit complex, but it does explain why cables can affect frequency response.

Frequency response problems associated with cables are generally due to capacitance between the signal conductor(s) and shield in the cable, and capacitance between the conductors themselves. This capacitance, in conjunction with the resistance of the cable, can act like an R-C low pass filter, cutting off high frequencies and dulling the sound. The effect is proportional to the length of the cable, and is worse over very long cable runs, but is mostly dependent on the output impedance of the circuit that drives the cable. Special line driver amplifiers are used to send signals over long cables. Some cables also exhibit considerable inductance between conductors, and depending on the resistance (and impedance) of the circuit, the result can be an R-L high pass filter that cuts off low frequency response. This is why it is important to use the right type of cable for a given job.

Electronic circuits represent a highly developed and controlled technology. Unless they are specifically designed to alter the frequency spectrum (like tone controls, for example), circuits can be and generally are made very flat in frequency response.

Figure 2-8 shows a typical frequency response plot of a high quality audio power amplifier. Note that the response is absolutely flat throughout the audio range, falling off only at the frequency extremes.

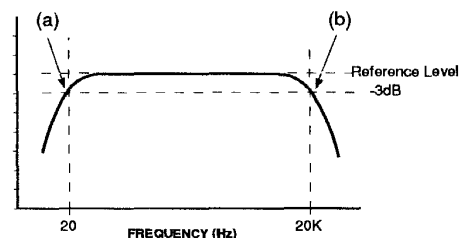


Figure 2-8. Frequency response of a typical audio power amplifier

The low-frequency rolloff indicated in Figure 2-8(a) is sometimes called a subsonic filter pole. Its effect is to reject ultra-low frequencies that could damage loudspeakers and/or modulate the audio to produce undesirable distortion.

The high-frequency rolloff (b) is sometimes called a TIM (transient intermodulation distortion) filter. This intentional limiting of high-frequency response serves to minimize TIM, and also helps to reject supersonic frequencies that can damage tweeters.

Similar characteristics can be expected of any professional audio signal processor. An exception is the digital delay or digital signal processor, both of which use very steep input and output low pass filters (so-called anti-aliasing and reconstruction filters) to avoid problems associated with the digital signal processing; in these devices, there will be a significant roll-off above the cutoff frequency of the device — as high as 150 dB per octave. Unless the design intentionally limits the response, however, analog audio signal processors should be flat throughout the audio range.

2.2.2 Microphones

Contemporary microphone technology is also highly developed, and it is possible to make microphones with very flat frequency response throughout the audio range. Because practical microphones are used in part for their musical or sonic characteristics, however, they may exhibit certain types of controlled deviations from flat response.

One such deviation is called a presence peak. This is a broad peak in response of 3 to 6 dB, centered in the region between about 2 kHz and 5 kHz. It has the effect of adding a bit of crispness to the spoken voice, and can increase the intelligibility of words.

Condenser type microphones often exhibit a peak in the neighborhood of 8 kHz to 10 kHz. This is generally due to diaphragm resonance, and can lend a slightly brittle or bright quality to the sound.

Ribbon type microphones sometimes exhibit both a presence peak and a slight low-frequency rise, usually around 200 Hz. This lends to them a warm sound quality, making them popular choices for both voice and certain instruments.

Generally speaking, the response of practical dynamic microphones used in sound work falls off rapidly in the octave above 10 kHz. Similarly, their low-frequency response may fall off gradually below about 100 Hz. Additionally, the angle of sound incidence upon the microphone may affect the frequency response so sounds coming in from the sides may sound different from those picked up on-axis; this is particularly evident with directional mics. A sound reinforcement mic that is popular for solo vocals or individual instruments may have a harsh response, with many resonances that actually brighten the apparent sound and help it to cut through a complex mix; these same characteristics may make such a mic unsuitable for pickup of multiple instruments or voices, and less-than-ideal for recording work. No one mic is ideal for every application.

The exact characteristic of any microphone is determined by a complex set of design factors, and one is well-advised both to check the data sheet and to try out the microphone. A new series of microphones introduced by Yamaha in 1986 utilizes unique

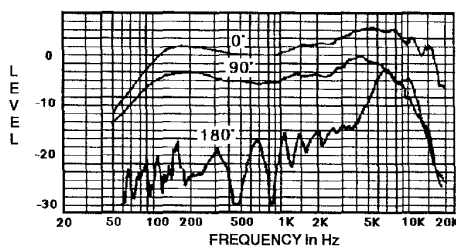


Figure 2-9. On and off axis frequency response of a typical cardioid (unidirectional) dynamic microphone

materials (beryllium laminated diaphragms) and innovative manufacturing techniques (screen-etched precision ports, for example) to achieve extended response and very smooth off-axis characteristics. In any case, some of the most important characteristics of a microphone may never be fully measured and documented. The selection of a microphone for a given application remains a highly subjective task, and the ear is the best guide.

2.2.3 Loudspeakers

Of all the elements in practical sound systems, loudspeakers are subject to the greatest variations in frequency response. It is not uncommon to find narrow peaks and dips of 10 dB or more in a loudspeaker's response. In fact, an overall response tolerance of ± 4 dB, measured on a $1/3$ -octave basis, is considered very good.

This seemingly broad tolerance is considered good partly because loudspeaker transducers are asked to handle relatively large amounts of power and produce fairly high sound pressures, so their design must allow compromises for durability and efficiency. Practical loudspeakers also must use several transducers to reproduce the full audio range; the marrying of these transducers to form a single functional unit is a complex matter, and can drastically affect the system frequency response.

Sound reinforcement loudspeakers can be divided into two basic classes: full-range loudspeakers, and special units that cover limited ranges. The

SECTION 2

latter are used to augment the former, extending their response.

Generally, full-range reinforcement loudspeakers will have a frequency response extending from about 100 Hz to between 10 kHz and 15 kHz. In most applications, this response is sufficient. Some smaller loudspeakers, such as those used for paging, may have more limited response — but for voice, it is generally true that the loudspeaker response should extend from about 300 Hz to at least 3.5 kHz.

Of the special, limited-range units, the most common are subwoofers and tweeters. Subwoofers are loudspeakers designed to operate only below 300 Hz (typically below 100 Hz), and usually perform down to about 30 Hz. They are used to extend the bass response of systems. Tweeters usually work above 5 to 8 kHz, and are used to extend the system's high-frequency response.

Practical loudspeakers are directional — that is, they concentrate the sound that they radiate in a specific direction. As you move away from their primary axis, the sound not only drops in level, but the frequency response gets rougher.

For further information on the frequency response characteristics of loudspeakers, see Section 13, "Loudspeakers."

In order to relate the frequency response of a device or system to its sonic characteristics, we need to get a sense of the frequency ranges covered by typical program sources.

2.3 VOICE AND INSTRUMENT RANGES

2.3.1 The Speaking Voice

Compared with the full range of human hearing, the speaking voice covers a relatively narrow frequency band — roughly from 100 Hz to 6 kHz. Within that range, the power distribution of speech is heavily weighted in the frequencies below about 1 kHz, with some 80 percent of the energy concentrated in the range below 500 Hz.

While speech has very little high-frequency content, however, almost all of the energy of consonants occurs above 1 kHz. Therefore, loss of high frequencies can drastically affect speech intelligibility.

As a practical minimum, the bandwidth of a speech reproduction system must extend from 300 Hz to 3.5 kHz. This is the frequency response of a typical telephone receiver, and other types of utility communications devices exhibit similarly limited response. Generally, intelligibility may be improved by introducing a presence peak in the response of 3 to 6 dB in the region between 2 kHz and 5 kHz.

Sound reinforcement applications require somewhat higher fidelity than a telephone, however. For this reason, public address systems should be reasonably flat from 100 Hz to at least 8 kHz. A presence peak can improve intelligibility in such systems, too — but it can also increase the likelihood of feedback, so it must be approached with care.

2.3.2 The Singing Voice and Musical Instruments

Figure 2-10 (next page) is a very useful graph which describes the frequency ranges of musical instruments and trained singing voices.

Musical signals are much more difficult to characterize than speech. The frequency range and power spectrum of a particular musical signal depend in large part on stylistic matters — the instrumental complement, the arrangement, the style of production, and so on. Some styles of music, for example, depend heavily on energy

between 20 Hz and 100 Hz to achieve their full intended impact, while those frequencies may be entirely absent in other styles.

The sound reinforcement professional therefore must learn to use his or her ears to carefully analyze musical material and correlate the sonic experience with system performance. The ability to do this comes only with long practice listening to a variety of equipment with different sound sources. In sound work it's very important to always listen carefully and examine what you hear, gradually building a mental catalog of experiences from which to draw.

2.3.3 Harmonics

Looking at Figure 2-10, we see that it seems to show the frequency range covered by musical instruments as limited to about 4 kHz. Yet we know that human hearing extends far beyond this limit, and that a sound system whose frequency response is

this limited will sound dull, lacking in highs. The apparent discrepancy is due to the fact that Figure 2-10 does not take harmonics into account.

Every musical sound that we hear is in fact a composite of sine waves at different frequencies and amplitudes. These sine waves combine to form the sound, and their frequency and amplitude relationships determine the quality, or timbre, of the sound. The ultimate waveform may be more of a triangle, squarewave or something not easily described in a word.

When a musical note with a complex waveform has a distinct pitch (as opposed to noise), that waveform can ultimately be created by combining a set of precisely related sine waves. These sine waves are called harmonics, and their frequencies are related in simple integer multiples. The sine wave frequency at the pitch that we hear as the note is the fundamental, and it is usually (though not always) the strongest (highest amplitude) of the set of sine waves which comprises the complex waveform.

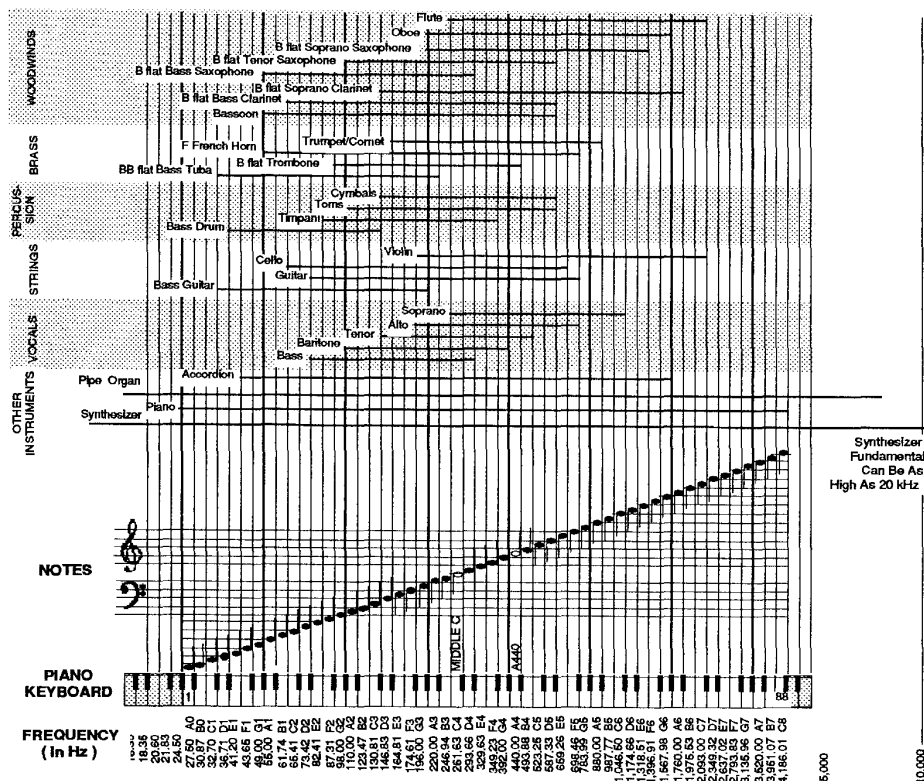


Figure 2-10. Frequency range of typical instruments and singing voices

FREQUENCY RESPONSE

Above the fundamental there are additional sine wave components whose frequencies are multiples of the fundamental frequency. If the fundamental is at 500 Hz, for example, the harmonics will occur at 1 kHz, 1.5 kHz, 2 kHz, 2.5 kHz, and so on. Figure 2-11 is a graphical illustration of the harmonic spectrum of a musical sound.

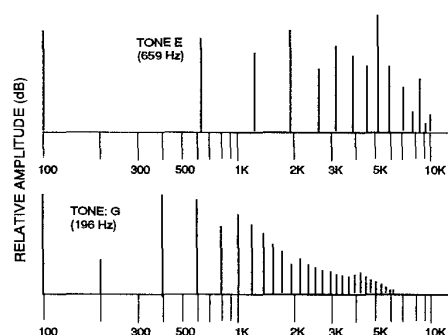


Figure 2-11. Harmonic content of open strings of the violin. The lower tones have more closely packed harmonics.

As the frequency multiplier increases, the amplitude (or strength) of the harmonic usually decreases so the upper harmonics are generally much lower in level than the fundamental. But this is not always the case. Sometimes, individual harmonics are louder than the fundamental. In such cases, the sound quality becomes more pinched or like that of a reed instrument (i.e., like an oboe, clarinet, and so forth).

If the sine wave components of a sound are not related in simple integer multiples, the sense of pitch is lost and the sound quality approaches noise. Drum sounds, for example, have very complex sets of components with non-integral frequency relationships.

Regardless of whether the sine wave components of sound occur at integral multiples, their precise amplitude relationships determine the sound quality. For this reason, flat frequency response is very important in a sound system. If the response is very rough, the quality of the sounds will be altered.

2.4 EFFECT OF ACOUSTICAL FACTORS

The frequency response of any sound system is affected by the environment in which it works.

Outdoors, the main factors affecting the system's response are wind, temperature, and air absorption. Wind tends to deflect sound very slightly as it travels, and gusty wind can appear to modulate the sound. Temperature gradients can also deflect sound, and to an even greater degree than wind gradients. Air absorption affects mainly high frequencies. As sound travels through air, the energy of the highs is lost more quickly than that of the lows. That is why sounds heard at a distance appear muffled. The degree to which the air absorbs high frequency energy is affected by the relative humidity.

Indoors, the predominant factors affecting a system's response are reflections from wall, ceiling and floor surfaces, and room resonances. Reflections not only cause reverberation effects, but also can cause cancellations at specific frequencies which show up as dips in the system response. Room resonances can cause dips or, more commonly, peaks in the response. All these effects will color the sound of the system.

The effect of environmental factors on sound systems is dealt with in detail in Section 5 ("Sound Outdoors") and Section 6 ("Sound Indoors").

13.8 Full-Range Loudspeakers

By using an appropriate crossover (as discussed in Section 13.7.1) to combine two or more drivers that reproduce different frequency bands, loudspeaker systems can be created that cover most of the audio frequency range. Such systems are referred to as full range loudspeakers.

NOTE: A full range driver is a single loudspeaker that is intended to reproduce most of the audio frequency range. One example is the typical 6" x 9" oval car stereo loudspeaker. Such units cannot do a very good job at the extremes of the audio spectrum, nor can they generate the very high sound levels required of a full range sound reinforcement system. The full-range loudspeakers we discuss here are not single driver units, but rather are systems comprised of two or more individual loudspeakers (or drivers) that each reproduce a portion of the audio spectrum.

In sound reinforcement, full range loudspeakers are usually two way or three way systems. They may employ passive crossovers, active crossovers, or a combination of the two. Systems comprised entirely of horn loaded loudspeakers or compression drivers are fairly common because of their high efficiency, although some professionals prefer the sound of direct radiator loudspeaker systems. Especially in smaller systems, the most common approach employs a combination of direct radiator low frequency drivers and horn loaded high frequency drivers.

Two basic approaches have been taken to the packaging of full range systems. These are illustrated in Figure 13-23.

One approach, Figure 13-23 (a), employs separate enclosures for each individual frequency range, these being stacked in various combinations to form the system. This philosophy was common in older sound systems, and is still used in many systems today.

Advantages that are claimed for separate enclosures include ease of handling and improved control over the

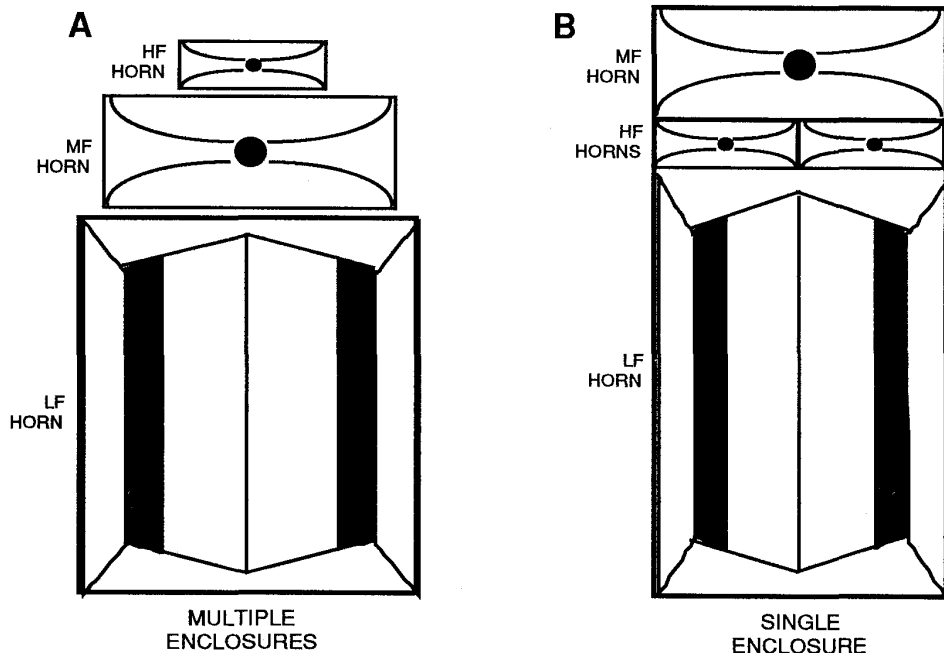


Figure 13-23. Two of the many ways to package a 3-way loudspeaker system

system. Many reinforcement professionals have come to feel that these advantages are outweighed by a number of disadvantages, such as the increased complexity of hookup (and thus greater potential for mistakes), lack of control over the phase alignment of the system, and the general physical instability of stacks (a pile of separate boxes can tumble down).

The most common contemporary approach, Figure 13-23(b), is the single full range enclosure. This approach is particularly convenient in the case of systems that use a passive crossover, since there is no necessity to daisy chain multiple connections from one cabinet to another. If an active crossover is used, a single multipin connector and multiconductor speaker cable can be used to simplify connections.

The setup and testing of full range systems is covered in Section 17 and Section 18.

13.8.1 Effect of Boundary Conditions

When a full range loudspeaker is used in an environment with boundary surfaces, its placement affects its perceived frequency response. When such effects are properly understood, they can be used as another trick for producing the desired sound quality.

Consider Figure 13-24. In (a), we see an omnidirectional radiator (a theoretical, idealized loudspeaker) in free field. We measure its sound pressure output at a distance d , and set this to be our reference pressure level, or 0 dB SPL.

If we now place a large reflective boundary (i.e., a wall, ceiling or floor) next to the omnidirectional loudspeaker, Figure 13-24(b), the sound that is radiated toward the boundary is reflected. We see that, as a result, the sound pressure at distance D increases by as much as 3 dB. The source is radiating its power into half as much space, and so this is called *half space loading*.

In Figure 13-24(c), the loudspeaker is placed at the junction of two perpendicular boundaries. Now its power is radiated into one quarter the space; this is called *quarter space loading*.

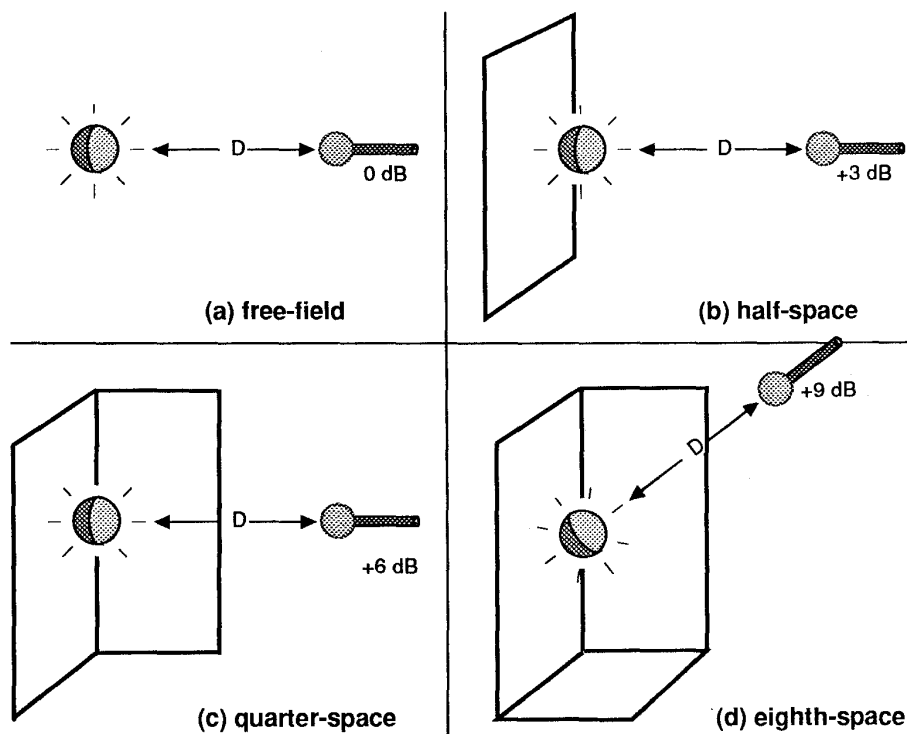


Figure 13-24. How SPL Increases Due To Boundary Effects

13.9 LOUDSPEAKER SPECIFICATIONS

Note that the SPL has increased by 6 dB relative to the full space, omnidirectional situation in (a).

Finally, in Figure 13-24 (d) we place the source in a three sided corner. This reduces the space to one eighth of the free field, and is called *eightth space loading*. As before, the SPL increases by another 3 dB, to 9 dB more than (a).

Loading affects the output of practical full range systems, as well as this theoretical system, but the effect is not the same at all frequencies.

Note that our example assumes a theoretically omnidirectional radiator. Practical loudspeakers are omnidirectional only at low frequencies (where the wavelength is large compared with the size of the unit). At high frequencies practical loudspeakers have controlled directional characteristics, and radiate sound in one direction (presumably, away from the wall).

For this reason, if we place a full-range loudspeaker at a wall-ceiling junction or in a corner, the low frequencies will be boosted while the highs remain essentially unchanged (they are aimed away from the walls anyway, and so there is little or no reflected energy to reinforce the direct-radiated energy in front of the loudspeaker). If the loudspeaker is deficient in bass, or if we want a bass-heavy sound, the boost provided by the corner placement would be welcome. Otherwise, the system may require equalization to reduce bass output. Again, this can be a benefit. If we know that large amounts of low-frequency energy will be demanded from the system, we can use half-space or quarter-space placement to boost the lows acoustically, then use equalization to cut the lows back down electronically, thus increasing the system headroom (by decreasing power demands on the amplifier and loudspeaker system).

The specifications of a driver or loudspeaker system should give us all the basic information we need to use it in an application. The main attributes that are normally given in loudspeaker specifications are frequency response, power handling, sensitivity, impedance, and directional characteristics.

13.9.1 Frequency Response

Frequency response is a comparison, by frequency, of the on-axis sound pressure output level of a loudspeaker to its input power level. A typical full range loudspeaker frequency response specification might read:

Frequency Response:
30 Hz – 15,000 Hz, ± 3 dB

The specification tells us the range of frequencies that the loudspeaker is designed to accurately reproduce (30 Hz to 15 kHz). It may be able to operate somewhat beyond this range if corrective equalization is applied, but the basic flat response region is defined by this specification.

The figure ± 3 dB (or + or -3 dB) is called the tolerance of the specification. It tells us that within the stated frequency range, if we compare the output sound level to the input power level at any frequency, the ratio between the two will fall within a 6 dB window ($+3$ dB - -3 dB) = 6 dB).

Frequency response can also be presented as a graph, as shown on the next page in Figure 13-25.

The horizontal axis is frequency, on a logarithmic scale. The vertical axis is sound pressure level, on a linear scale. The graph assumes an input signal to the loudspeaker that is equal at all frequencies. Note that this loudspeaker would barely satisfy the specification given on the previous page (30 Hz – 15 kHz, ± 3 dB).

Frequency response deviations such as those shown in Figure 13-25 are considered normal for average, practical loudspeakers. Since they are complex transducing systems, loudspeakers' behavior in response to an excitation signal is subject to a number