8050A Real Time Audio Analyzer

Features:

Designed to Highest Quality Standards

Allows Acousta-Voicing® Process to be Completed in Less than 10 Minutes

Makes Room Tests (Demonstrations) Practical with Less than 10 Minutes Tuning

Reduces Amount of Required Test Equipment

27 Contiguous 1/3-Octave Bandpass Filters

40 Hz to 16,000 Hz

20 dB Display Range

Self-Contained CRT Display Screen

Fast and Accurate — Scans All Channels in Approximately 30 ms

Simple Controls — Easy to Use

Easily Rack Mounted

Preamplifier Gain is Continuously Variable from -20 dB to +20 dB

Sold Only by Selected ALTEC Acousta-Voicing Contractors

Priced to Fit your Budget

FOR USE IN RECORDING STUDIOS, BROADCAST STATIONS, SOUND SYSTEMS, FILM STUDIOS, AND SCHOOL LABORATORIES EVERYWHERE THAT ACCURATE DYNAMIC SPECTRUM MEASUREMENTS ARE REQUIRED

The Altec 8050A Real Time Audio Analyzer is an engineering achievement in the precision instrument field. Its performance is similar or equal to instruments costing three times greater and it proudly bears the ALTEC name, the hallmark of quality. Its low price makes it available to every audio professional.

The ALTEC 8050A offers fast and accurate visual reference to obtain spectrum loudness and balance information of the acoustic system, making it a superb addition to the recording studio. Its broad application ranges from high fidelity dealers (who may use it to demonstrate frequency response of microphones and speakers as well as electronic components in a total system) to broadcast stations, auditoriums, legitimate theatres, traveling road shows and by sound stage engineers. Anyone involved in high quality sound will find the Model 8050A indispensable.

Price is no longer a barrier for not being able to enjoy the advantages of real time audio spectrum analysis. This new Altec product invites feature by feature comparison with any similar unit of its type on the market.

The ALTEC 8050A is a truly compact real-time analyzer for the audio spectrum. The instrument covers the frequency range from 40 Hz to 16 kHz with 27 parallel bandpass filters as shown in Figure 1. Detectors following the filters convert the ac filter outputs to dc levels proportional to the rms value of the ac signals. An internal scanner sequentially connects the 27 detector outputs to the CRT display screen of the instrument through a log converter and simultaneously generates a linear ramp at the "X" deflection. The gain of the preamplifier circuit in the 8050A is continuously variable from -20 dB to +20 dB, permitting shifting of the 20 dB display from the range of 70—90 to 110—130 dB. An overload lamp on the front panel of the instrument indicates preamplifier overloads as brief as 100 µs.

The real time audio analyzer is also available in the Model 01-8050A configuration. This model is identical to the 8050A, except it has a built-in power supply for using the analyzer with the accessory Model HP 15119A 1/2" Condenser Microphone.

ALTEC ® LANSING

1515 S. Manchester Ave., Anaheim, Calif. 92803

ALTEC DIVISION OF LTV LING ALTEC, INC.

SPECIFICATIONS

Type:	Real time acoustic response analyzer for full audio-frequency spectrum
Frequency Range:	From 40 Hz to 16 kHz
Measurement Range:	From 70 to 130 dB above 1 μV rms (3.16 mV rms to 3.16V rms)
Display Range:	20 dB. Can be shifted continuously over entire measurement range.
Input Signal Range:	From 3.16 mV rms to 3.16V rms. For a steady sinewave signal, the preamplifier circuitry will accept input levels up to 30 dB above full-scale indication. Maximum permissible input level is $\pm$150V dc plus 50V peak ac.
Gain:	Continuously variable from -20 dB to +20 dB
No. of Bandpass Filters:	27
Filter Characteristics:	Single pole pair. Nominal center frequencies at 1/3-octave increments in accordance with ISO preferred center frequencies (see Figure 1) $Q = \frac{fm}{B_{-3\,dB}} = 4.5$
Detection Mode:	RMS SLOW or RMS FAST. Dynamic characteristics are in accordance with IEC 179. Single sine wave tone burst, with a duration of 1/2 the time constant, indicates -4 dB $\pm$1 dB with respect to the steady sine wave indication.
Detector Accuracy —	
For tone burst signals with crest factors equal to or less than 3:	$\pm$0.5 dB with respect to the steady sine wave indication
For Gaussian random noise:	$\pm$0.2 dB with respect to the steady sine wave indication. The test signal is applied to the detector without any frequency weighting. Filter characteristics, such as response time and effective bandwidth must be considered for overall accuracy.
Y-Display Accuracy:	$\pm$1 dB over the 20 dB display range for a steady sinusoidal signal at the center frequency of each bandpass filter channel.
Scanning Time:	Internal scan covers the 27 channels sequentially in approximately 30 ms
Input Impedance:	100,000 ohms
Input Power Required:	115/230V +10%, -15%, 80VA maximum, 50 to 400 Hz
Input Connection:	Coaxial connector on bottom right side of front panel
Overload Indicator:	Lamp at upper right side of CRT screen on front panel. Illuminates if preamplifier is overloaded. Circuit responds to overload peaks having more than 100 μs duration. Minimum duration of indication is 100 ms.
Rear Panel Controls:	27 screwdriver adjustable potentiometers for setting filter output levels. Rotate each clockwise (cw) to increase respective output.
Front Panel Controls —	
LINE ON-OFF:	2-position rotary switch on bottom left of panel. Turns power on and off.
GAIN:	Continuously variable potentiometer on bottom left center of panel. Rotate cw to increase height of display on CRT screen.
RMS SLOW/RMS FAST:	2-position switch at bottom right center of panel (see Detection Mode specification)
ILLUM:	Screwdriver adjustable potentiometer at upper left side of CRT screen. Rotate cw to increase intensity of reticle illumination on CRT screen.
FOCUS:	Screwdriver adjustable potentiometer below ILLUM. control on panel. Adjusts focus of display on CRT screen.
ASTIGM.:	Screwdriver adjustable potentiometer below FOCUS control on panel. Adjusts astigmatism of display on CRT screen.
INTENSITY:	Screwdriver adjustable potentiometer below ASTIGM. control on panel. Rotate cw to increase intensity of display on CRT screen.
VERT. POS.:	Screwdriver adjustable potentiometer below OVERLOAD indicator on panel. Adjusts vertical position of display on CRT screen. Rotate cw to raise position.
HORIZ. POS.:	Screwdriver adjustable potentiometer below VERT. POS. control on panel. Adjusts horizontal position of display on CRT screen. Rotate cw to move display to right.
HORIZ. GAIN:	Screwdriver adjustable potentiometer below HORIZ. POS. control on panel. Rotate cw to increase width of display on CRT screen.
Operating Temperature:	From 0^{o}C to 50^{o}C (32^{o}F to 122^{o}F)
Storage Temperature:	From -20^{o}C to 50^{o}C (-4^{o}F to 122^{o}F)
Humidity:	Withstands up to 95% relative humidity at 40^{o}C (104^{o}F)
CRT Screen Dimensions:	1-1/2" H x 2-1/2" W (38 mm H x 64 mm W)
Overall Dimensions:	5" H x 16-3/4" W x 11" D (127 mm H x 425 mm W x 279 mm D)
Net Shipping Weight:	18.7 pounds (8.5 kilograms)
Accessories:	Hewlett Packard Model HP 15119A 1/2" Condenser Microphone (for use with Model 01-8050A Real Time Audio Analyzer)

TECHNICAL NOTES ON THE ALTEC 8050A REAL TIME AUDI(

By
Don Davis and John Eargle

Figure 1. 8050A Real Time Audio Analyzer

GENERAL

The 8050A Real Time Analyzer, built for ALTEC by Hewlett-Packard, combines in a single lightweight unit all the advantages of Altec's earlier recommended instrumentation for real time analysis. Its features include:

1. Lightweight — only 18 pounds.

2. Low cost — about one third the cost of the earlier HP Model 8054A.

3. "Phantom" powering for condenser microphones.

4. Fast or slow rms response.

5. 20 dB range read-out wide enough for all Acousta-Voicing® applications.

6. Accepts input signals over a 1000:1 voltage range.

INPUT REQUIREMENTS

Any high quality dynamic microphone can be used with the 8050A. However, if the HP Model 15119A Condenser Microphone is used, no external power supply is necessary because of the phantom powering provided by the 8050A. The input is also switchable to a non-powered receptacle for examining the outputs of electronic devices.

CRT DISPLAY

The input signal is fed to a set of 27 1/3-octave filters whose outputs are then rectified according to either a fast or slow rms characteristic. The outputs of the rectifiers are scanned once each 30 msec, and these are fed continuously to the CRT sweep display. The display area is 1-3/4" x 2-3/4". The 1-3/4" vertical scale is marked off in easy to read 5 dB increments, while the horizontal is marked off in octaves. Obviously then, there are three read-outs between each division.

At the very bottom of the plastic graticule in front of the CRT the octave frequencies are abbreviated. Because of the small area occupied by the total display, these indications are small and may require some practice to read and interpolate accurately.

The input gain control and range switch allows a 1000-to-1, or 60 dB voltage range, to be accommodated on the screen of the CRT. In practice, this gain control would be varied during the course of Acousta-Voicing to keep the entire spectrum well in the middle of the screen. A plus-or-minus 1-1/2 dB tolerance can easily be maintained visually with the vertical scale marked off in 5 dB increments.

Here are some sample displays on the CRT of the 8050A.

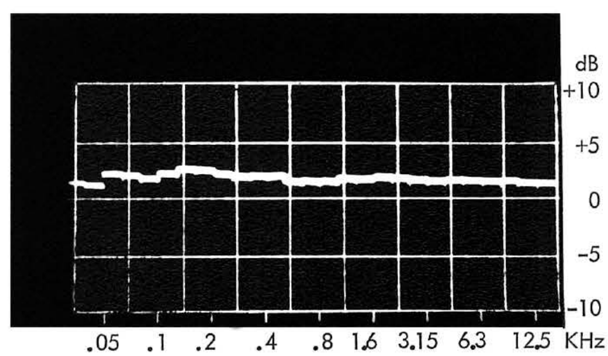

Figure 2. Response to Pink Noise

Litho in USA CP-372-2.5K

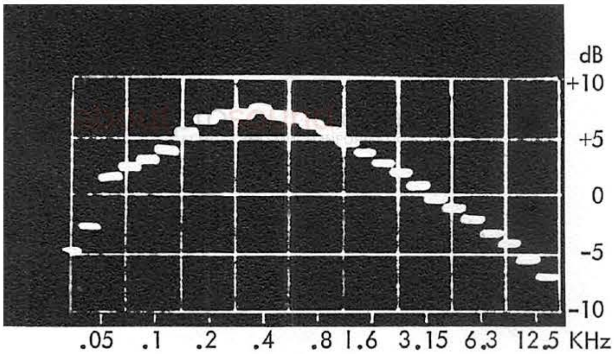

Figure 3. Response to USAS1-Weighted Noise

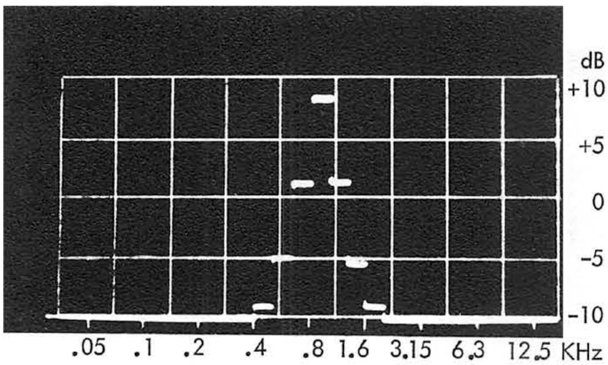

Figure 4. Response to 1 kHz Sine Wave

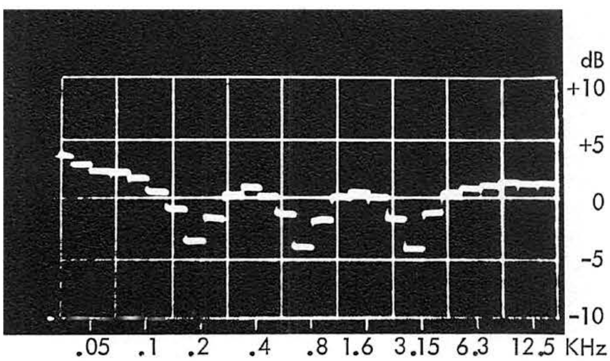

Figure 5. Response to Pink Noise Through the 9856 Active Equalizer (200, 800 and 1600 Hz) Set for Maximum Attenuation

TUNING A PLAYBACK-ONLY SOUND SYSTEM

The typical playback system Acousta-Voicing set up is shown in Figure 6.

1. Turn the GR 1328A to "pink" noise and adjust the output until the acoustic level is at least 20 dB greater than the ambient noise level at the position from which you intend to measure. Check all voltage levels in the signal path to ensure that no electronic components are overloaded.

2. Place the measuring microphone (either the HP 15119A or other appropriate microphone) at the measuring position.

3. Adjust the gain control on the front panel of the 8050A for a mid-screen placement of the curve being measured.

4. Proceed with the tuning.

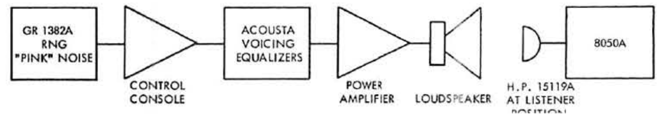

Figure 6. House Curve made using Measuring Microphone

SOME HINTS FOR EASE OF TUNING

Several hundred tunings have shown it is always best to start with the highest point on the given house curve. It is important to remember that for the first three steps on any of the narrow-band filters five bands will be affected — the band the filter is centered on plus two above and two below the center frequency. This means that when peaks are next to dips in the house curve, the entire area may need to be initially lowered by the filter centered on the peak frequency before the filter becomes narrow enough to control only one band. Normally, such lowering becomes part of the overall broadbanding needed to achieve final balance and it is for this reason that the ALTEC 9018A equalizer or the ALTEC 9017-series filters are not used as frequently as in the past.

It is best not to introduce corrections too rapidly with any one filter at the onset; just barely pull down the highest peak until it is even with the bands nearest it. This usually causes a re-adjustment of the entire area followed by the same gentle procedure at the next peak presented. It does not take long to find out that no one can hope to completely understand Acousta-Voicing without a lot of experience on the job with a real-time analyzer. Several of the contractors have set up a test system and hold informal contests among their men to see who can master a smooth, efficient tuning the fastest. Such practice can lead to very impressive on-the-job performance at times when the customer is watching.

As the tuning proceeds, opportunities to lower areas will offer themselves. Careful choice of the correct filters will result in all of these filters being at a narrow-enough setting (past 3 dB) to allow detailed finishing of the final curve. If, near the end of the tuning a bump exists in the curve, and use of a particular filter that has not been used before puts in too wide a correction, the best policy is to wipe out the whole tuning and start over again. During the second tuning, be more careful in the initial choice of the filters so as to allow the detailed correction of the desired bands near the conclusion of the tuning.

Of course, attention must be paid to avoid tuning in the null of the standing wave pattern. A short walk with the microphone of the analyzer, especially in a small control room, is fascinating, instructive and necessary.

Similar care should be observed in the handling of dips in the response of a loudspeaker and a room by diaphragmatic action of some boundary surface. This is identifiable when, after bringing down all the bands around the dips, they still fall the same number of dB below the surrounding bands. Do not chase it on down because that will only increase the insertion loss of the total equalization with but negligible improvement in the response. The correct method is to drive the loudspeaker-room combination with 1/10-octave bands of noise from the GR 1564A (see Figure 7). Use the continuously variable tuning of the 1564A and observe the effect on the 8050A to find the frequency where the absorption of the signal is greatest; then use your fingertips and feel all the surfaces of the space, including walls, doors, windows, etc. You will feel the offending surface vibrating in sympathy with the test signal. Hannon Engineering of Los Angeles, in Acousta-Voicing the Decca Records Studios, was able to find a walled-over window area unknown to the studio users that was the cause of a deep dip in the response curves. Naturally, such a scientific approach to a studio problem left a highly favorable impression of Hannon Engineering with the customer.

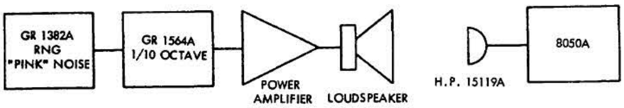

Figure 7. Finding Diaphragmatic Absorption

TUNING A REINFORCEMENT SYSTEM

Up to this point, nonregenerative sound systems have been discussed — that is, acoustic feedback has not been a factor in their tuning. When an open microphone is used, several additional conditions complicate the tuning procedure:

1. The microphone response is usually not smooth, especially in the case of cardioid microphones.

2. The multipath long-time-interval reflections, with their varying amplitude and phase, become part of a regenerative system as the overall gain is increased.

3. The listening area is spread over a wider range with a typical reinforcement system than with a typical playback system.

It is normal practice to start the tuning of a reinforcement system in the same way as a playback system — i.e., smoothing the house curve by placing the measuring microphone in a typical listener position in the audience area. It has been our practice to do this even before checking the distribution array of the loudspeaker and then having someone walk the test microphone while watching the screen on the real-time analyzer. This is a much more rigorous way to learn the fine art of loudspeaker coverage and leads to correct methods more quickly than using the 4000 Hz octave band filter (built into the GR 1565-9002) by itself. This procedure makes it very obvious why we recommend that horns be stacked and splayed versus side by side. Naturally, it is futile to attempt to Acousta-Voice a sound system that will not hold the voiced house curve over the majority of the listening area.

It is always of great interest to examine the difference between the house curve thus tuned in the audience area and the house curve observed at the location where the microphone used in the sound system is to be placed. When the sound system microphone is located in an area with a lower room constant (more reverberation) than the area where the audience is seated (e.g., where the stage house is large and live, and the audience area is highly absorptive), quite different house curves can be expected at the two locations. Because the real-time analyzer allows economical inspection of these situations, we expect to gain a great deal more insight into the behavior of sound systems in rooms during the next year as each of you report your results on literally hundreds of jobs.

Having worked out the distribution and the smoothing of the house curve, you are then left with the problems of the microphone's actual response (as used in the space) plus the advantages or disadvantages of its positioning relative to the loudspeaker, audience area, and immediate reflecting surfaces.

We have all learned that sound system microphones can vary over a wide range of useful amplitude characteristics. There is also the irrational to consider — when the performer prefers a certain shape, size, color, trade name, or personal microphone.

If the microphone is located where $D_1 < D_c$ and tuned for maximum gain, it will be sensitive to being moved. If it is placed so that $D_1 > D_c$, it will be almost completely insensitive to location changes so far as maximum acoustic gain is concerned.

We have demonstrated, on a number of occasions, the effect of mounting a microphone a foot or two away from a reflecting surface versus flush-mounting on or in the surface. The phase cancellation that occurs is easily visible on the real-time analyzer if you use a flattened loudspeaker as a sound source and move the microphone away from its stand and down to the reflecting surface. We are eager to hear from each of you about novel solutions to the flush-mounting and shock-mounting of microphones to take advantage of this effect. In Catholic churches, flush-mounting of an omnidirectional condenser microphone, such as the ALTEC M51, in the top of the altar allows substantially more speech energy to arrive at the microphone diaphragm.

If it is desired to meausre the frequency response of the sound system microphones, a high-quality loudspeaker capable of being tuned to ±1 dB uniformity should be used (the ALTEC 9844 is excellent for this purpose). The sound system loudspeaker is not normally a good choice because of its remote location. For best results, a distance of 10 feet between the loudspeaker and microphone is suggested as a good standard distance. Use of this distance will allow meaningful comparison of data taken at many locations. It would be very useful to us if you would send in copies of your microphone curves thus taken.

If the sound system microphone has rough response, and maximum acoustic gain is the main desire, then it should be introduced into the measuring chain as shown in Figure 1. If the sound system is smoothed with the sound system microphone as the measuring microphone, you should be near maximum gain. This will not be so if the acoustic environment where the microphone is to be actually used is essentially different from the environment at the loudspeaker location.

The ultimate test of whether or not you have achieved the maximum acoustic gain is to take a feedback threshold response and see if further increases in gain are possible.

In dealing with feedback, experimental use of the controls on the ALTEC 9014A can be most rewarding. You may be well into a tuning and have a feedback mode come up squarely on 1000 Hz; however, experience has shown it is wise to try both the 800 Hz and 1250 Hz controls as well. In many cases an additional 3 dB of attenuation of the 1000 Hz control will be necessary to handle the feedback and only 1 or 2 dB of the 800 or 1250 Hz control. The reason for this seems to be associated with the need to change the slope rate of the filter correction curve in the 1000 Hz region; hence the phase, rather than merely changing the amplitude. In any case, it has been proven that the most accurate matching of the inverse electrical-response curve of the filter with the raw house curve occurs when such details are applied.

From these notes on the use of the 8050A Real Time Analyzer, it can be seen that the following list of uses is just a beginning for the imaginative sound contractor:

1. House curves made with measuring microphone.

2. House curves made with sound system microphone.

3. Examination of distribution of all frequencies at the same time.

4. Examination of the house curve at the performer's location.

5. Response curves of the filter settings (see Figure 8.)

6. Detection of feedback frequencies.

7. Frequency response of microphones to be used in the sound system.

8. Examination of crosstalk between lines (see Figure 9).

9. Setting levels throughout sound system both electrically and acoustically.

10. Detection of resonating surface areas by means of observing the effect of manual damping of the vibrating surface.

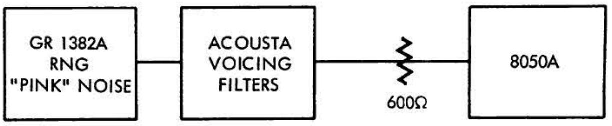

Figure 8. Finding Electrical Response of Filters

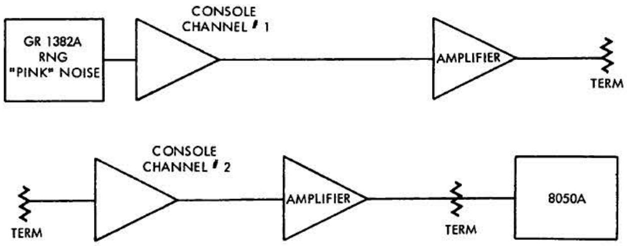

Figure 9. Measuring Crosstalk

Your competence in sound system design, installation and Acousta-Voicing will increase many fold through the use of this valuable new instrument.

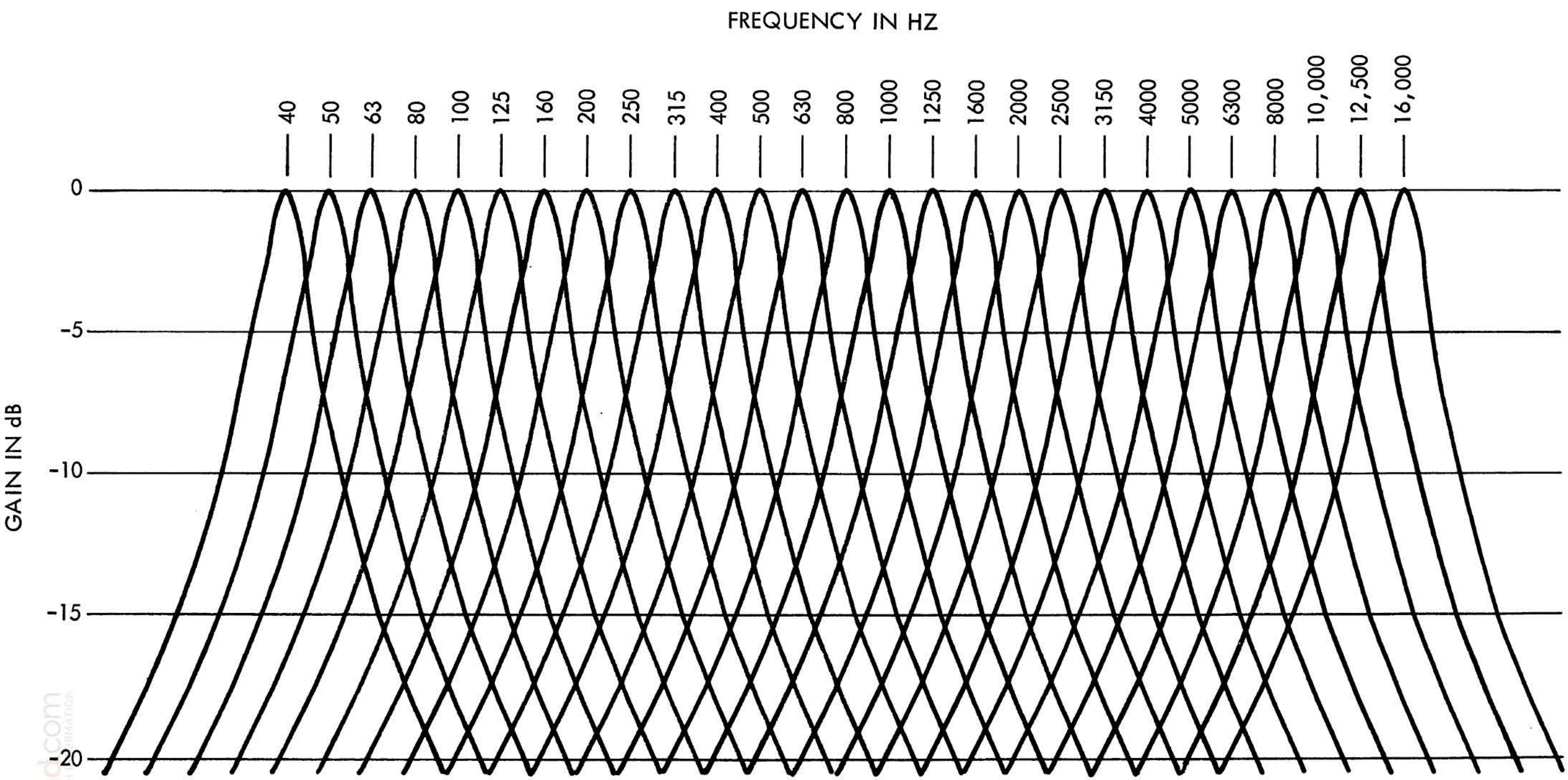

Figure 1. Bandpass Filters of 8050A – 27 Channels – 40 Hz to 16 kHz

ALTEC 8050A

ARCHITECT'S AND ENGINEER'S SPECIFICATIONS

The real time audio analyzer shall be capable of continuously displaying the characteristics of the audio spectrum from any sound source between 40 Hz and 16 kHz. It shall contain 27 adjustable bandpass filters arranged in 1/3-octave increments. The filter controls shall be located on the rear panel of the analyzer. Filter center frequencies shall be in accordance with ISO preferred center frequencies. The measurement range shall be from 70 to 130 dB above 1 μV rms. The display range shall be 20 dB and shall be capable of being shifted continuously over the entire measurement range. The input signal range shall be from 3.16 mV rms to 3.16V rms. The audio spectrum shall be displayed on a 1-1/2" H x 2-1/2" W (38 mm H x 64 mm W) CRT screen which shall be located on the front panel of the analyzer. The analyzer shall contain a preamplifier that shall be capable of accepting input levels up to 30 dB above full-scale indication. Maximum permissible input level shall be ±150V dc plus 50V peak ac. Detector accuracy shall be ±0.5 dB, with respect to the steady sine wave indication, for tone burst signals with crest factors equal to or less than 3. Detector accuracy shall be ±0.2 dB, with respect to the steady sine wave indication, for Gaussian random noise. The analyzer shall be capable of applying a test signal, without frequency weighting, to the detector circuitry. The analyzer shall have a Y-display accuracy within ±1 dB over the 20 dB display range for a steady sinusoidal signal at the center frequency of each bandpass filter channel. The analyzer shall be capable of sequentially scanning the 27 filter channels each 30 ms. The scanning circuitry shall convert the ac filter outputs to dc levels proportional to the rms value of the ac signals. The scanner shall sequentially connect the 27 detector outputs to the CRT display screen through a log converter and shall simultaneously generate a linear ramp at the "X" deflection. The analyzer shall be capable of operation in two detection modes: RMS SLOW and RMS FAST. The dynamic characteristics of the detection modes shall be in accordance with IEC 179. The analyzer shall be capable of detecting a single sine wave tone burst, with a duration of 1/2 the time constant, and indicate -4 dB ±1 dB with respect to the steady sine wave indication.

In addition to the CRT display screen, the front panel shall contain an overload indicator, a coaxial-type input connector and 10 controls. The overload indicator shall illuminate when overload peaks of more than 100 μs duration are applied to the preamplifier circuitry. Minimum duration of overload indication shall be 100 ms. The bottom of the front panel shall contain the LINE ON-OFF switch, GAIN control and the RMS SLOW/RMS FAST switch. CRT display screen controls shall be grouped on either side of the CRT screen on the top of the front panel. These controls shall be ILLUM., FOCUS, ASTIGM., INTENSITY, VERT. POS., HORIZ. POS. and HORIZ. GAIN.

The analyzer shall be capable of operation after storage in temperatures ranging from -20°C to 50°C (-4°F to 122°F). It shall be capable of operation in ambient temperatures ranging from 0°C to 50°C (32°F to 122°F). It shall withstand up to 95% relative humidity at 40°C (104°F).

Overall dimensions of the analyzer shall be not greater than 5" H x 16-3/4" W x 11" D (127 mm H x 425 mm W x 279 mm D). The net shipping weight of the analyzer shall be not greater than 18.7 pounds (8.5 kilograms). Shall be capable of being rack mounted.

Any real time audio analyzer not meeting all these requirements shall be unacceptable under these specifications.

The basic real time audio analyzer shall be the ALTEC Model 8050A Real Time Audio Analyzer. The real time audio analyzer equipped with a built-in power supply for use with an accessory condenser microphone shall be the ALTEC Model 01-8050A Real Time Audio Analyzer.

NOTICE

We recommend that you obtain your Altec products from factory trained authorized Altec Sound Contractors and Distributors. This will assure you of proper installation, a continuing source of knowledgeable advice, service, and quick warranty protection.

LITHO IN U.S.A. CP-172-5K AL-1791

Specifications and components subject to change without notice. Overall performance will be maintained or improved.