A NEW AUTOMATIC SPECTRUM ANALYZER BASED ON FUZZY LOGIC

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Abstract:

A new spectrum analyzer incorporating limited learning and decision making capacity for use in automated sound reinforcement system equalization is discussed. The analyzer can determine the systems' frequency response, thermal, electrical and polar pattern behavior with reference to the location of a measurement microphone. The processes for implementing equalization within the learned system limits, intelligent system protection, maintenance of target responses, and dynamic compensation for ambient noise are described.

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0. Introduction

In a previous paper, it was postulated that certain expert knowledge of a loudspeaker/room system must be acquired prior to making valid equalization decisions. The scope of the present paper is to familiarize the reader with the basics of a fuzzy process and the application thereof in a "solution generating" analyzer.

1.0 Classic Analysis methods

In general, system analysis systems require the following:

a) a reference signal generation section

b) a receiving and conditioning section

c) a processing algorithm or hardware section

d) a data display storage and/or communication section, and

e) a human interface section (buttons, knobs, mouse/icon systems etc.)

Ergonomic factors are functions of the environment in which the product is to be used and by whom it is to be used. This paper deals with methods of extracting useful conclusions from frequency response data gathered by somewhat conventional means and how this information is used in an automatic equalization process. Virtually all classical methods are admirably executed in commercially produced products. Most of these demonstrate excellent dynamic range and accuracy. In fact, all of these devices performance may far exceed the necessary range and accuracy that the end user needs to make reasonable, real world judgments on device and system performance for equalization. This is not to say that accuracy is undesirable, but a reasonable trade off of precision for speedy, expert decision making capabilities seems to be more appropriate for automatic corrective equalization.

2.0 "Fuzzy" Processes

In any automatic process based on fuzzy logic, the open loop system takes on the characteristic behavior imposed upon it by a set of rules that are followed. There is good reason for using rules as opposed to limits. Fuzzy sets allow the open loop system to operate way out of its normal performance limits and function as reasonably as possible, and then smoothly return to pre-characterized behavior when external conditions permit. A good example would be calling an audio compressor a fuzzy limiter. The compressor doesn't do a very good job of confining the signal within absolute limits, but there is more to life than just levels, limits, and absolutes; the compressor does keep the signal within reasonable boundaries with a wide variety of input signal levels. The compressor operates with a pre-determined, reasonable rule (ratio) that trades off distortion and level boundaries and an open ended membership function (threshold) that defines which data should be processed. Both ratio and threshold are programmable to suit the needs of the operator.

Establishing and maintaining spectral balance in an arbitrary or semi captive sound reinforcement system is an order of magnitude more complex. Strings of phase and amplitude vs. frequency data can be easily measured and stored, but they are not rules; they are merely information subject to interpretation. By themselves, they are insufficient for carrying out "intelligent" processes. A special logical structure is needed before workable rules may be established and executed to achieve performance within specified boundaries.

The following is a simple review of fuzzy architecture. The main elements of any fuzzy system consist of the following:

a) the Fuzzifier

b) the Controller

c) the De-fuzzifier

The fuzzifier converts input data into fuzzy data. The controller, in conjunction with the fuzzifier, evaluates fuzzy data by a user-designed set of rules that describe how the system is to be controlled. After the rules have been executed, the de-fuzzifier assigns an action values to the appropriate outputs.

Membership functions are used to effectively divide the range over which an input can vary into sections. In operation of the system, membership functions are compared with input data to see how the data compares to them. The membership functions may have names selected by the designer, such as High pass, Band Pass and Low Pass, that classify the input data. We are concerned with six types of membership functions, as follows:

1) Left Inclusive
2) Symmetrical Inclusive
3) Right Inclusive
4) Left Exclusive
5) Symmetrical Exclusive
6) Right Exclusive.

In application, the primitive membership shape is specified with numerical values for relative levels, center frequencies, and band widths. More complex membership "shapes" and slopes may be formed by using multiple primitives. There are cases where membership centers and widths move around as a function of previously acquired data or conditionals. These "Floating memberships", as they are called, improve the process by allowing moderate "learning" capacity. In fuzzification, an input is subtracted from the center of a membership function and the absolute value of the result is inverted to measure how closely it matches the center value. Fuzzy membership functions allow the use of a fuzzy variable that directly measures the difference between two inputs. Floating membership functions can be combined with a floating action output value to obtain the derivative of an input value. This type of usage saves memory because it uses fewer rules and fuzzy variables.

A fuzzy variable is a linguistic expression representing the association of an input value vs. the membership functions covering an axis. Fuzzy variables reference a membership function and an input variable. The result is a fuzzy data value that represents the degree to which the data matches the membership function. A rule consists of one or more fuzzy variables and an action output value. Rules are used to tell the controller how to respond to changes in input data.

In evaluating rules, all the values for fuzzy variables in the rule are compared, and the lowest value represents the rule. Next, the values for the rules are compared and the rule with the highest value wins.

The two most used modes of defuzzification are absolute (or direct) outputs and ramped (accumulated) outputs. The absolute mode is useful for quick, immediate control, while the ramped mode is more useful for subtle "tweaking".

The application of this technology in an analyzer/equalizer loop is simple and direct. Memberships are generally a function of frequency partitioning (i.e., low-passing is a Left Inclusive Membership, band-passing is a symmetrical inclusive membership, etc.). Fuzzy variables change as a function of where the input frequencies are in reference to the stop-band. Rules are prioritized by behavioral filtering (not to be confused with frequency domain filtering) and winning values are output via direct mode for measurement and ramped (cumulative) mode for correction. The following behavioral filters exclude frequency response data as a function of "fixability" in the frequency domain. This is not to say that the these anomalies are unfixable, rather they are not repairable in the frequency domain. For the sake of practicality, the example presented is characterized specifically for use with a 1/3-octave graphic equalizer.

3.0 Low Frequency and High Frequency Limits

The analyzer determines equalizable bandwidth by subtracting the root of the squared average of the band pass energy from the root of the squared average of low pass energy from a sweeping second order state variable filter (SVF). When the result is less than one, a flag is set indicating the lowest equalizable frequency of the transducer or transducer/room system under measurement. High frequency limits are determined by subtracting the root of the squared average of the band pass energy from the root of the squared average of the high pass energy of the same filter. When the result is less than one a similar flag is set indicating the highest equalizable frequency. The analyzer effectively determines where the system's spectral limits approximate second order bandpass behavior and flags those frequencies as being the "equalizable limits" of the system. The methodology for accomplishing this requires the following progression for stimulus and detection: The test stimulus consists of band limited pink random noise (characterized by equal power per 1/3 octave band); this insures constant spectral power in each measurement band and "time blindness". This stimulus is buffered and fed to the transducer under test. On the detection end, the response (affected via the transducer) is buffered and routed to a second order state variable filter. The three outputs of the SVF (low pass, bandpass and high pass) are routed to peak hold amplifiers and then to hardware that accomplishes averaging routines. Ratios between band pass to low pass and band pass to high pass (for low and high frequency limit determination respectively) are then formed. When these ratios reach unity (or some other predetermined constant), the associated frequencies are marked as spectral limits. Any data outside of these limits is thrown out as unequalizable.

3.1 Driver Thermal Limits

The method described herein requires the bandwidth limitation data supplied by the first procedure. Within this bandwidth, filtered random noise is routed to the driver under test. With a receiving microphone constantly taking level samples the noise is increased in level. We are interested in a point of departure from a linear relationship between the change in output level and the change in receive level. At this point we may ascertain that a breakpoint has occurred in the ratio of power in to power out. This level is flagged as the T3 and is rated in watts input to the transducer. After the absolute thermal limit is established the driver under test is allowed a cool down period, followed by a full power (as determined by the previous T3 test) interval of band-limited noise. Level data is then gathered and stored, or displayed as the "thermal signature" of level vs. time, or a constant proportion to the thermal mass of the driver under test (S40). This data is then stored as the transducer's maximum input level T3 over the time interval S40.

3.2 Array Nulls

In array analysis, data is divided into the categories of equalizable dips and unequalizable nulls. The dips are correctable in the frequency domain, but the nulls are not. A common problem in data interpretation is judging the relationships between amplitude and phase trends and inflections, to arrive at a decision as to whether or not to correct the anomaly in the frequency domain. The analyzer accomplishes this by serially analyzing the phase and frequency response at a particular frequency or a frequency band. While a frequency component (from a filtered independent source or generated test stimulus) is swept across the element of the array under test, the receiving system senses any changes in relative amplitude and phase response. Any shift in mean phase congruent with a dip in frequency response is an indicator of an element and/or reflection interaction anomaly, and hence is not correctable with conventional frequency equalization. This differs from a correctable dip in that the phase response of the correctable dip returns to the mean phase trend. It is important to note that some correction of frequency response anomalies caused by phase or time may be affected through frequency equalization in the direct field, however the amplitude anomalies will be merely displaced into the reverberant environment where they will alter the total power radiated into the room. In the case of automatic equalization in the frequency domain, any data in the frequency domain affected by wave interference is thrown out as irrelevant data.

3.3 Nulls Due to Sympathetic Vibration

The analysis system must differentiate between nulls caused by a vibratory absorber and nulls caused by interference. Since vibratory absorption is of a sympathetic and/or parasitic nature, its occurrence may be detected by exciting the transducer/environment under test with fixed frequency impulses (smoothly-windowed sine waves with at least several full periods). On the receive end, broad band level samples are taken and an average of the energy over a fixed time interval is calculated and stored. The test is performed every 1/12th octave (or a finer frequency resolution if desired, provided the user is willing to wait...) and the result is displayed as energy decay time vs. frequency. The data is indicative of the level of spurious activity vs. frequency. This test would be very useful in locating vibrating walls, door panels, glass panes or any other types of mechanico-acoustic narrow band absorbers. Any null caused by such an absorber would be designated as unequalizable and the operator would be notified of it.

4.0 Equalization Rules

After the analysis system has "learned" the real world behavior of the system and environment, it is ready for real time operation. The analysis system assumes that the sensing microphone and loudspeaker array shall remain stationary.

During operation, the system senses any changes in the spectral behavior of the array from the sensing microphone perspective and applies its rules by asking the following questions:

1) Does the measured 1/3-octave band amplitude value agree with what has been predetermined?

If yes, go to next test band. If no, store difference and go to next rule.

2) Is measured band level above or below predetermined ambient level?

If above, go to next rule. If below, go to next test band.

3) Do the spectra fall into the interference-caused null category?

If yes, go to next test band. If no, go to next rule.

4) Do the band levels fall into the sympathetic null category? If yes, go to next test band.

If no, go to next rule.

5) Do the measured band levels fall into the previously determined equalizable limits?

If yes, go to next rule. If no, go to next test band.

6) Will the prescribed change (difference) in level cause the transducer to meet or exceed its known thermal limit?

If yes, implement the maximum change that may be tolerated, signal a warning, and go to next test band.

If no, implement change (difference) and go to the next test band.

5.0 System Limitations

To facilitate optimum real-time operation, frequency test band widths are established which vary the resolution as a function of perceptual significance. Under some circumstances, the measurement resolution may exceed the capacity of corrective devices in the loop, but this should be of little consequence, as the corrective device resolution will have been proven adequate for most uses.

The analysis system examines a given spectral membership, makes a decision on what to do with the data based on previously "learned" system parameters, and implements an action in the form of ramping pre-partitioned spectral bandwidths up or down as a function of the aforementioned rules. Correction could occur in .5 dB increments over each pass. Correction increments will be software adjustable, as insufficient data exists at this time to arrive at an optimum unobtrusive change rate. It should be noted that the rate of change should not be upsetting to the mix engineer, as well as to the audience. The update speed is determined mostly by the sample time required for the low frequency memberships. This would limit the system ability to respond to short term overload, however that function is usually reserved for dedicated quick action limiters (which the analysis system could supply setup data for).

There is still room for discussion as to if and when automatic correction complicates the mix engineer's task or interferes with his or her artistic motives in a "live" situation. If a cumulative correction mode is used, along with the user option to go back to or toggle between a preset (either pre-selected by the user or previously arrived at by the analyzer), most user fears are appeased. Some well respected mix engineers also feel that a feature (manual or automatic) that allows an up datable assessment of ambient conditions would be useful. There are still many live tests to be performed before any "hard" conclusions are drawn. Research and data gathering are ongoing concerning ergonomic and subjectivity issues.

6.0 Summary

Whatever the update speed, the system described herein is still an order of magnitude quicker than its human counterpart at interpreting data and taking appropriate action, as well as always applying correction within the system's physical limits. The methodology used also lends itself to reducing "operator error" in conventional data gathering systems. By separating data into "discrete domains" (frequency, phase, time, persistence, diffusion, etc.), users will be less prone to "correction" in the wrong domain.