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FACT SHEET

A DISCUSSION OF THE APOGEE CORREQT™ SYSTEM: ITS ORIGIN, PHILOSOPHY, AND NOTES ON DEVELOPMENT

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OVERVIEW

The Apogee CORREQT™ system is at once a philosophy of sound system design and practice, a technique, and an evolving compilation of hardware and software.

(CORREQT™ is an acronym standing for **C**omputer **O**ptimized **R**oom **R**esonant **E**Qualisation **T**echnique)

The mission of CORREQT is to enlarge, develop, and extend the performance results of the loudspeakers, in particular the Apogee product line; particularly and especially when the loudspeakers are utilized in reverberant/reflective acoustic environments.

Such utilization of the loudspeakers, (in reverberant/reflective environments), accounts for the majority of the typical deployment of the Apogee product base presently in distribution.

ORIGIN

CORREQT was originated during an approximately three year period to answer the needs of sound designers and consultants who were using Apogee speaker products in demanding applications. These users were experiencing differing perceptions of response characteristics as they changed venues; certainly not a unique experience in the sound reinforcement field. Upon systematic investigation, a common factor that emerged among the various users was their intent to find a way to minimize the contribution of the room's acoustic characteristics; thereby improving the consistency of the sonic results from location to location. The Apogee loudspeaker line had been engineered to minimize non-linearities at all normal operating levels. Because of the linear nature of the loudspeakers, it became relatively easy to identify the secondary characteristics of the acoustic environment as the greatest variable and the largest contribution to overall system non-linearity. (Note: system is defined here as the microphone(s), mixer, processor(s), amplifier(s), loudspeaker(s), and acoustic response characteristics of the room).

A method was indicated to solve for these non-linearities: a) the use of the transfer function measurement response of a two port FFT based spectrum analyzer to accurately identify the non-linearities; and b) the deployment of a precision parametric equalizer to correct the response thereby linearizing the system.

The dual channel FFT, had become an essential piece of instrumentation and quite commonplace at Apogee for research and development work along with routine production testing.

Study revealed that the dual channel FFT used as a two-port network analyzer could effectively form the basis of a response correction system by measuring the transfer function of the system (transfer function is the result of dividing the spectrum that is present at the input of channel B of the analyzer by the spectrum that is present at the input of channel A).

By using precision parametric equalizers, one could effectively 'null' the acoustical environment's non-linear response characteristics along with much of the resultant sonic coloration, exhibiting a flatter response and a greater measure of system control than previously available.

DEVELOPMENT

To develop the system, first the acoustic characteristics of many size and shape of rooms were analyzed. Typically, the most significant deviations from flat, linear response that are attributable to the room character (rather than speaker placement or multiple speaker interaction), were discovered to be in the low to mid frequency range; approximately 40 Hz to 800Hz.

Using parametric filters to alter the electrical response by the precise inverses of the acoustic response anomalies, proved to yield remarkable results. A typical subjective reaction to the corrected response was that of far less perceived reverberant field. Rooms were said to be "drier" and intelligibility factors, especially of podium and lavalier microphones (both of which are highly susceptible to re-entrant energy problems), increased dramatically. Rt60 measurements of corrected systems showed significantly less reverberation times when compared to flat systems, without incurring the loss of full range system response from the usual method of attenuating the whole bottom portion of the spectrum, thereby severely restricting system bandwidth.

Variations of this basic premise were explored. Measurements, subsequent equalization, and follow up measurements were tried repeatedly under a wide range of conditions. Much empirical data was collected and collated. A basis was formed for the validity of the technique.

The conclusion, that by attenuating the energy at those specific frequencies where the room has natural resonant modes, can significantly and perceptibly decrease the overall resonance of the room without adversely limiting the frequency response of the system, was proven many times over by two specific methods of evaluation.

The effectiveness of the technique was found to be a function of the following factors:

1) THE CHARACTERISTICS OF THE REVERBERANT FIELD.

This affects the potential for altering the perceived reverberant characteristics of the room. If the reverberant field is reasonably flat with smooth high frequency fall off characteristics, the ability to alter the room sound by equalization of the system (without adversely affecting the direct sound characteristics) is not as great as in a room where the reverberant field exhibits significant peaks and dips in response.

2) THE PRESENCE OR ABSENCE OF ROOM RESONANT MODES AND THEIR RELATIVE INTENSITY

Rooms that exhibit strong low frequency resonant modes that are narrow in Q, harmonically related and spaced apart by 2 -3 times their Q factor, can be easily flattened without creating adverse side effects. Rooms which do not exhibit easily identifiable resonant modes, or where the modes are very wide, are more difficult to equalize to advantage, as greater subtlety is required to avoid broadband attenuation, altering the system's spectral balance.

3) THE PRESENCE OR ABSENCE OF SIGNIFICANT ACOUSTICAL ANOMALIES.

Severe acoustical problems, such as pronounced flutter echoes, discreet echoes, or radical response variations from one area of the room to another, are problematic and increase the difficulty of the equalization efforts. Nevertheless, significant improvements can be realized.

4) THE CHOICE AND PLACEMENT OF LOUDSPEAKERS & THE OVERALL SYSTEM DESIGN.

Just as a poorly designed room may be problematic in nature, a poorly designed loudspeaker or loudspeaker cluster may be difficult or impossible to improve much with equalization. Additionally, the placement and directional focus of the loudspeakers will ultimately dictate the obtainable end results. The system design needs to be approached in a methodical and informed fashion to allow the CORREQT™ technique to achieve its full potential.

METHODOLOGY

The two-port analyzer, with its ability to compute transfer function, is used to measure the system response during initial system set-up. This is accomplished by stimulating the system from the internal periodic noise (or chirp) source of the analyzer and displaying the signal from a measurement microphone placed in the room. This method yields rapid measurement response for quick set up, due to the easily resolved nature of the stimulus signal.

(Note: periodic noise is white noise containing timing information, allowing the analyzer to read time domain response in addition to frequency domain response.)

As mentioned elsewhere, typical room resonant modes can be very narrow in bandwidth. Therefore the extremely narrow and specific nature of the required filter corrections indicated by the measurement results raise concern of the stability of the system. Would such specific settings of the equalizers be valid as room conditions changed; i.e. temperature, humidity, volume of air in the room displaced by patrons, etc.? Would the equalizers themselves prove stable enough?

The two-port analyzer again proved to be the answer. The system response is continually verifiable during actual performance with the house full of patrons, by utilizing the transfer function computation to compare the electrical response of the program material (music, speech, etc.) on channel A to that of the acoustic response (derived from a measurement microphone) of the loudspeaker and room characteristics on channel B. By averaging a number of datum, a stable and interpretable trace is produced even with sporadic program material as the system stimulus. Any deviation from linear (flat) response is readily shown and can be corrected for during the performance. The averaging slows the process somewhat, but is fast enough to use as a verification of the previously constructed corrective curves developed during system set-up, and allows subsequent adjustments to fine tune the system to the new room conditions.

It is important to note that the usefulness of the precise narrow band corrections that are typically employed with the CORREQT technique could prove to be invalid and possibly even detrimental if the results could not be monitored and subsequently altered throughout the show as conditions change. (At the least, if the system cannot be monitored during an actual event, then the CORREQT operator must use discretion in selecting very narrow band corrections, and artificially broaden the Q to compensate for possible changes of conditions that may occur at sometime in the future.)

That acoustical changes that occur are due to shift in the room's acoustic characteristics when it fills up with patrons, due to the absorbency and displacement of atmosphere resultant from the presence of the patrons, along with changes in temperature and humidic conditions.

For example, at the Dorothy Chandler Pavilion during the 1990 Academy Awards, a 14 dB cut with a bandcenter of 70 Hz, 1/6 octave wide (one the primary resonant modes of that room), which was identified during rehearsals, needed to be changed to a band center of 78 Hz during actual performance with a house full of patrons. Leaving this cut at 70 Hz would not have helped attenuate the room resonance mode and would have created an acoustic "hole" at 70 Hz. But by tracking the system's response and adjusting the filters accordingly, the valuable correction established during rehearsal was re-tuned to produce nearly the same result during performance.

With a multi-zone system a typical CORREQT setup may employ as many as 50 - 60 filters covering the various loudspeaker zones. This large number of filters usually dictates many equalizers crammed into a rack with no air space between them. The resultant heat can cause the filter circuits to drift significantly.

A special model of equaliser was developed to add far greater confidence to the system. The Apogee CRQ-12 employs a thermostatically controlled fan to stabilize the internal temperature rise in the unit, thereby retarding filter drift dramatically over characteristics displayed by passively cooled models.

A validity check on the significance and usefulness of the data can be made using the *coherence* function of the analyzer. This is important because the process of acquiring valid data is directly proportional to the

coherence of the information arriving at the analyzer's two input channels. The electrical reference signal must be accurately delayed to match the transmission delay of the loudspeaker's acoustic energy, relative to the microphone(s) position. Also, the general signal content must be the same at both channels. For instance, audience reaction (applause, cheering, etc) is generally NOT present at the console outputs (at least not with much intensity), but is present at most typical measurement microphone locations. By using the coherence function, one can check to see if the deviation from the flat or target response that one may view on the analyzer represents a real condition or is the result of differing signal content or timing.

DEVELOPMENT NOTES

Further development consisted of field work in which a basic philosophy to deal with multiple loudspeaker zones was indicated, so as not to detract from the progress made with the primary source(s).

Once the system's main resonant modes had been defined and corrected, the casual addition of ancillary loudspeakers was found to be prone to re-exciting and to some measure "undoing" the careful work of correcting the primary system's response. A method was described to deploy ancillary loudspeakers without the adverse effect stated above.

First the measurement microphone is located in the physical area that the ancillary loudspeaker is intended to cover. A measurement is taken with the ancillary loudspeaker switched off and the primary, or main, system on. The response is viewed. Then, the ancillary loudspeaker's response is shaped with the equalizer in such a manner as to fill in that loudspeaker's coverage zone with only the spectral data that is not present from the primary or main loudspeaker(s). Of course the ancillary speaker is carefully delayed to match the arrival time of the main source.

In this manner, the ancillary loudspeaker does not add additional energy to the room at those frequencies that are not required in its intended coverage zone (if this energy is present by virtue of carry over or "spill" from the primary loudspeaker or loudspeaker clusters). This technique minimizes the potential build up of energy at frequencies where the room has exhibited resonant modes.

EQUIPMENT

The equipment used for CORREQT consists of:

- 1) A set of calibrated measurement grade microphones; typically one for each major loudspeaker coverage zone. B&K 4007 and 4011 types can be used with good results.
- 2) A dual channel FFT based spectrum analyzer. Presently Apogee is using the Hewlett Packard 35665A.
- 3) A signal router, which accesses the inputs and outputs of the various equalizers that are patched into the system; accepts, preamplifies and selects the signal from the measurement microphones, and routes selected signals to a digital delay unit and to the two input channels of the FFT analyzer. Apogee manufactures the CORREQT Model II Switch. This unit provides for six microphones and six equalizers or equaliser chains, and is expandable to any number of devices by bussing additional units.
- 4) A high quality digital delay unit.
- 5) A computer controller which interfaces to the HP 35665 by way of the HPIB port. The computer allows indefinite storage and subsequent retrieval of response traces, along with some advantages for ease of use.

A typical approach to applying the CORREQT technique to a room follows:

[The basis of the system connectivity and usage of the equipment is shown in block diagrams #1 through #4].

- 1) The system is stimulated with either periodic noise or program material. The switch is first selected to show the un-equalized room response (ROOM MODE) at a given microphone location, usually covered predominantly by a given loudspeaker or cluster (in a design that supports CORREQTtm practice). Internally, the switch mode routes the output of the equalizer associated with this zone, (delayed by the precise increment of the loudspeaker's transmission time to the measurement microphone), to the A channel of the FFT. The measurement microphone output is routed to the B channel input. This resultant amplitude vs. frequency trace is stored and displayed from the analyzer's memory. *This trace represents the room response.*
- 2) The switch is changed to the INVERTED EQ MODE. A correction curve is then built on the equalizer and displayed in inverted form on the analyzer. Internally, the switch routes the output of the equalizer to the FFT's channel A input and the input of the equalizer to the channel B input. The equalizer curve is adjusted to exactly overlay the stored room response curve. *This trace is the inverted correction curve.*
- 3) The switch mode is now selected to display the results of the corrected system response (RESULT MODE). For this display the input to the equalizer (again delayed by the transmission time) is routed to channel A of the FFT and the measurement microphone signal is routed to channel B. *This is the corrected room response.* Additional and ongoing adjustments are usually made in this mode.
- 4) The normal, non-inverted equalizer response (EQ MODE) can then be displayed for storage purposes and later study. By repeating this process throughout a number of frequency spans and in the different loudspeaker coverage zones, a linearized system response may be obtained. Of course, this first assumes that there are no significant non-linearities in the loudspeakers or other system components themselves.

ADDITIONAL DEVELOPMENTS

Future developments will center around greater manipulation of the data and ease of storage and retrieval of instrument preset states, traces, and video display. The ideal system shall have trace-math functions; the ability to average, subtract, add, or divide multiple traces will enhance both usefulness and versatility.

Methods of averaging the response of multiple microphones located around the room are under development.

A semi-automated option is planned, in which the computer will take control of the system and set-up proposed solutions for correction of non-linearities. The operator will have the discretion to choose an appropriate point during the show to introduce the correction. He or she will be able to rapidly revert back to the previous settings if the correction is not deemed appropriate. Otherwise, the new correction can be accepted and the computer will go on to look for additional non-linear system responses.

This semi-auto enhancement will interface with future Apogee products such as the CRQ-120 programmable parametric equalizer. The digitally controlled analog equalizer will be equipped with an external controller interface and an internal recall memory bank.

SUMMARY

The power of advanced equalization techniques is only finally beginning to emerge as a star in its own right rather than a walk-on extra. Highly visible shows such as the Grammy Awards and the Academy Awards, both of which have used Apogee's CORREQT system for five years now, have proven the value in having a full time equalization operator on staff during system installation, rehearsals, and especially during the show itself.

The future development of sophisticated hardware and software will allow the process to be performed in a semi-automated manner, eliminating the requirement for a separate operator, lowering the labor costs and thereby broadening the potential user base. Additional breakthroughs in both hardware and software control systems will allow the process to be understood and mastered by operators who may not necessarily possess training of instrumentation usage in their background. Finally, the cost of the hardware itself will become affordable to all economic sizes of users, bringing the technology again to a much wider user base.

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The end goal of the Apogee CORREQT development program, in keeping with Apogee's basic corporate philosophy, is to significantly increase the quality of reinforced sound in as broad of a market as possible.