Harman Kardon HK 990 Stereo Integrated Amplifier with Digital Room Correction and Dual Subwoofer Bass Management – Part I

Written by Dr. David A. Rich

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Article Index

Introduction to the HK 990 – A Truly Unique Stereo Component

The HK 990 is a truly unique stereo component. It is a dual domain device capable of accepting analog or digital inputs, converting domains as required, and sending the results to the speaker. A complex room-correction system in a stereo integrated amplifier separates the HK 990 even further from comparables. For those who archive vinyl or magnetic tape to CD-R, the dual domain nature of the HK 990 tape record function provides unique functionality.

Specifications

- Design: Solid State Stereo Integrated Amplifier
- Power: 2 x 150 watts RMS into 8 ohms @ 20 Hz - 20 kHz, 2 x 300 Watts into 4 Ohms
- MFR: 10 Hz – 100 kHz
- THD: <0.07% at Full Output (8 Ohm Load)
- Analog Inputs: 7, Plus 1 Phono MC, 1 Phono MM, and 1 Balanced XLR
- Digital Inputs: 1 HRS-Link, 2 Optical Digital, 2 Coaxial Digital
- Analog Input Sensitivity/Impedance: 350mV/43k ohms for tuner/CD, 10mV/47k ohms for Phono-MM, 1mV/100k ohms for Phono-MC
- Digital Input Capability: All Standard Digital Formats
- Dimensions: 6.4" H x 17.3" W x 17.5" D
- Weight: 43.2 Pounds
- MSRP: $2,599 USA
- Harman Kardon
- SECRETS Tags: Harman Kardon, HK 990

This review is independent of Tyler Stripko’s review published in June, 2011, where Tyler comments on a different sample. This review concentrates on measurements and internal design of the HK 990 and fits nicely together. Tyler covers the listening impressions.

Part II and Part III of the current review have also been published.

What Makes the HK 990 Unique?

When I am asked to select electronics for a new stereo system, I often surprise the questioner by recommending an AV receiver. AVRs have digital-signal processing (DSP) chips which, among other things, do room correction and advanced bass management. AVRs take digital data directly in from a CD player and, after passing it through the DSP, send data directly to an internal digital-to-analog converter (DAC). The analog signal emerging from the DAC is inches from the power amplifier. No analog cables are required. The analog signal from the DAC output must pass through the volume control before it sees the power amp. In a traditional stereo preamp or integrated amp, the volume control is a mechanical device with an unreliable wiper pressing on a piece of carbon. An AVR resolves the reliability issues of the potentiometer because it is a digitally-controlled device integrated circuit (IC). The potential divider uses a tapped integrated resistor, and the desired tap is selected by a CMOS switch. The digital aspect also enables the balance function to be accomplished concurrently. Balance is feasible since each channel’s digital control value is different. At high frequencies, mechanical balance controls are often the primary source of reduced channel separation, a problem not associated solid-state devices. Tone-control functions are embedded in the DSP, obsoleting analog electronics (formerly associated with loading the main signal path), mechanical switches, and mechanical controls.

The overhead of an AVR for stereo can be costly: one is paying for stuff they are not using, including five power amplifiers, five DACs, and five digital volume controls. Obviously, cost savings are squeezed across the analog section to keep the price reasonable when seven, rather than two, channels of analog electronics reside in the box.

The video switching and up-sampling circuitry may represent half the price of an AVR, but have no utility for a stereo music system. Some functions of the DSP processing (e.g., Dolby and DTS loss-less multichannel coders, surround sound synthesis, and five additional channels of room correction filters) offer no value for stereo; so, you are paying for more DSP than is needed. Last and definitely least, a TV is required to access the AVR's setup menus, which are not accessible from the front panel. Obviously, the rear panel is simplified with no video I/O's and only two channels coming in and out.

An ideal setup for the stereo enthusiast would have the AVR manufacturer jettison the superfluous stuff, upgrade the analog section, and alter the configuration of the front so setup adjustments can at the front panel, not on a TV screen. I have asked every manufacturer who sells AVRs and traditional analog stereo products when they will adhere to this roadmap; "never" is typically the answer on the belief sales volume will not justify the costs of re-engineering.

Harman Kardon must have had an epiphany for they are introducing the HK 990. The target market is high-end analog integrated amplifiers, not traditional stereo receivers; hence, the $2500 price tag. At that price, state-of-the-art analog and digital components are de rigueur.

The HK 990 shares the EzSet/EQ II digital room-correction system with the top-of-the-line $2500 AVR 7550HD as well as its bass management system for two subwoofers. Other current Harman Kardon AVRs use a simpler system.

The Analog Electronics

In the halcyon tube days of the 1950's, Harman Kardon's top of the line Citation products were among the best and continue to sell for top dollar on Ebay. The Citation subsidiary was the first to produce state-of-the-art consumer solid-state equipment, but the pricing relative to tube equipment was severe. Harman kept refining the Citation solid-state line. Those with long memories will recall the very high-end Citation XX power amp and XXp preamp of 1980. Overseeing the designs was then VP of engineering, Matti Otala, who joined in the late 1970's.

The HK 990 amplifier is a direct descendent of the circuit topologies that Harman developed under Matti Otala. The exotic Otala design rules were intended to improve the sound quality of the unit. A search of Otala's name in the Audio Engineering Society (AES) library finds 24 papers, starting with a seminal work on transient intermediation distortion in 1974 and running through input current requirements for high-quality loudspeakers (1983). Some of these papers were controversial. Otala is perhaps best known for advocating for the use of low levels of return-loop gain that are invariant across the audio band (a so-called low feedback design).

Bob Cordell's new text, Designing the Audio Power Amplifier, examines some of the issues that engineers face with Otala's concepts. The Cordell text is a great read for those interested in the design of a modern amplifier with very different return-loop gain characteristics. The Cordell text requires some background in analog electronics. A book of similar spirit is Doug Self's Audio Power Amplifier Design Handbook.

If you are attracted to the sound of an amplifier designed with Otala rules, then he may have been on to something. I will not opine, whether an Otala design is "closer to the truth" as Robert Lawrence Kuhn would put it for matters of cosmic significance.

I do not have access to schematics of the Citation XX units, which had limited production runs and prices far above Harman's norm. Some popularly-priced Harman units, circa 1981, signaled Otala's influence, but it was not until 1987 that the Citation name was resurrected for the 21 preamp and 22 power amp (200 watts/ch into 8 ohms). The 21 and 22 had the full Otala treatment, though by then he had moved to Nokia to design cell phones. The 1993 price of the pair was $1700. The Citation name was dropped in 1993, but similar products were manufactured through 1997. The trail goes cold as Harman USA went all-in for Home Theater. Harman stereo components continued to evolve in the EU. The HK 990 is the first high-end stereo unit to make it back on this side of the Pond. The HK 990 appears to be the best implementation of the Otala design rules of anything that sold in quantity in the U.S.

The only issue I identified on the analog side is a high 330pf capacitor at the input of the moving magnet phono stage. When combined with the cable capacitance of your turntable, the total capacitance may fall outside the range recommended for the cartridge, with the exception of Grado cartridges. Most who purchase a $2600 integrated amplifier will likely be using a moving coil cartridge with most high end moving magnets out of production. That MC stage on the HK 990 has no interface problems.

There is a red glow at the top of the HK 990. This is not due to the presence of vacuum tubes inside the unit. The glow is from the phono board, which uses LEDs as part of the circuits that set bias currents.

Performance with High Resolution Files

High-resolution digital downloads are becoming more common, so any equipment you purchase going forward should have the capabilities to deal with them. From measurements provided by Harman it was clear that high resolution LPCMs streaming data from a file server made it from the SPDIF inputs to the DAC. This held true even if the room correction and bass management were active. If the high resolution files were recorded with wide bandwidth the Harman measurements show the HK 990 will pass spectral information to 45kHz.

The HK 990 lacks a USB port. Converters to SPDIF are relatively cheap for 96kHz sampling rates. 192kHz conversion is more problematic. The HK 990 does not support direct streaming of MP3, WAV, or FLAC. Almost no two channel DACs provide for this. The conversion must be done by the device storing the high resolution files. At some point, it is necessary to draw the line between the long-term standard like LPCM over SPDIF and what is constantly evolving and device specific (Apple Air Play). Replacing a $2600 unit every couple of years is not economical. Instead, use a lower cost unit to deal with all the permutations for file streaming and storage of today and next year.

Dual-Domain Tape Recorder Outputs and Tape Monitor Functionality

A dual-domain tape monitor should be a must-have for those seriously rooted in recording. With a dual- domain configuration, both analog and digital inputs appear at the analog tape output jack Analog and digital inputs also appear on the unique to the HK 990 record digital output The source on the record outputs can be independent of the program playing on the speakers.

More on the tape recorder interfaces will be found later in the review. One issue I do want to mention now is that mixing analog and digital connections may result in a recorder self loop and high-level oscillation that could damage your speakers. Properly connected this cannot occur.

Overview: A Three-Part Structure

My review has three segments. The first segment is the introduction that you just read, which highlights the special aspects of the HK 990, and this is a nice precursor to the rest of the segment, which examines the performance of the EzSet/EQ II room-correction system.

Using block diagrams, the second segment considers the challenges of adding digital inputs and digital signal processing to a stereo integrated amplifier. This segment is a good read for those interested in the internal operation of an AVR at the block level. Since the HK is a stereo unit, the blocks are less complex. Once you have a handle of the operations at a block level, then the transition from two to seven, or even eleven, channels becomes more straightforward. The second segment also comments on the quality of the data converters in the HK 990 and additional performance data when the unit processes a high resolution file.

The last segment turns to the design and construction of the HK 990 at the circuit level. The analysis starts with the analog circuits in the direct mode, which is equivalent to an analog integrated amplifier. The designs of the power amp and phono stage are in the spotlight, and we also look at the tape recorder output and monitor function in more detail.

HK 990 EzSET II Room Correction Introduction

The frequency of the correction in the HK 990 room-correction system can be limited to 1 kHz corrected. 1 kHz is slightly above the room-dominated region (500Hz). Full-band correction (to 20kHz) is also an option. The system not only makes the standard set of measurements at the listener’s seat but also require you to position the microphone two feet away from the speaker, on axis, to estimate the direct near-field response without room reflections. Not surprisingly, the accuracy of the measurements two feet away can be problematic because two feet is inadequate for the wave front of some large complex (4 way for example) speakers to converge. Harman engineers indicated that the microphone could be moved back to three feet or even half the distance between the speaker to the listening seat to handle this issue.

The near-field measurement is a unique feature of the HK 990. One assumes the room-correction system uses the near-field measurements as a guide for the EQ above 1 kHz. Above 1 kHz, a speaker’s near-field frequency response errors dominate over room related effects. Unfortunately, it is not as simple as making a direct near field measurement. The anechoic off-axis response of the speaker also affects what the listener hears. Harman may have guessed the end user has limited patience for making measurements at each speaker.

The HK 990 room-correction system supports two mono subwoofers with a separate room correction filter in each signal path. Harman Labs developed the multiple mono subwoofer technique to reduce variation of the frequency response at the listeners’ seats. The lab identified four as the optimal number:

The four subwoofer technique also widens the room area where the response of the subwoofer is well controlled. A home theater with six to nine seats is the obvious application for the technique. The full four-channel subwoofer system can be implemented with a product called BassQ. It is an external box that corrects only the subwoofers and is marketed under the JBL Synthesis brand to custom installers, though is made available on request to consumers. The HK 990 and AVR 7550HD use this technique with a two subwoofer limit. Only with two subwoofers does the HK 990 room-correction system allow multiple in-room microphone measurements. A single room measurement is made at the prime listening position if the speakers are run full range or with one subwoofer.

The HK 990 cannot be connected to a computer to display pre- and post-corrected frequency response on a computer screen. This is a feature on some competitive systems. Disappointingly, a calibrated microphone is also not provided. By “calibrated,” I mean the microphone was placed in a test device with a reference microphone known to be flat. A file is created that represents the microphone error. The microphone is marked with a number that matches the file. When you set up a room EQ with a calibrated microphone, the first thing you do is enter the file into the processor, typically, a computer which interfaces with the electronics.

Anthem offers calibrated microphones starting with the entry level $1000 AVR. Audyessy-enabled AVRs products can be attached to an optional calibrated microphone, but only a custom installer can access it, which is a major downside. Both Anthem and Audyessy, with the calibrated microphone, use a computer that displays pre- and post-frequency response charts. The Anthem and Audyessy enabled products are AVRs with all the disadvantages to stereo users outlined above.

HK 990 - Problems Identified in the Measurement of EzSet/EQ II

I made measurements of the HK 990's operation over several months in two rooms. The rooms were equipped with two brands of full-range and two brands of satellite speakers. One subwoofer was deployed. Speakers were placed in different positions to test the responsiveness of the EQ. All speaker positions were plausible for pleasure listening in the home, although the listening seat position might require some adjustment. I placed the microphone at the adjusted position. The room's acoustic response was captured with the Acoustisoft RPlusD measurement system (www.acoustisoft.com). Having used Acoustisoft products for more than ten years, I have confidence in their accuracy, especially when averaging large data sets. My microphone is from IBF-Acoustic and it is recalibrated to a reference microphone every few years.

Three problems were identified with the HK 990 room-correction system:

(1) The level of boost below the port frequency of the speaker might exceed 15dB in the case of a subwoofer and 6dB in the case of full-range speakers. While the frequency response of the speaker appeared to be extended in acoustic response curves by this boost, the signal was distorted when listening with single tone SPL at 85 dB. This was especially apparent for the subwoofer.

The bass boost was evident under most test conditions, though there were a couple of one-offs where the boost moderated or even was a cut. In several instances, there was a repeatability issue likely related to the ambient low-frequency noise level of the room. I have not observed this repeatability problem with other EQ systems. The HK 990 produces a single, very short, chirp. Competitive systems have longer chirps or MLS noise. The test signals are often repeated in competitive implementations to average-out low-frequency noise.

(2) The subwoofer level, as calculated by the HK 990, was 6 – 8 dB low and required manual correction. Also, the crossover frequency selected by the unit was far too low. Manual settings appeared to be accurate with respect to the actual crossover frequency in my measurements.

(3) The EQ2 mode, appears from the measurements to allow a max EQ of approximately 2 kHz, does not work correctly. At the approximate 2 kHz transition point, the EQ does not return to 0dB insertion loss; instead, it holds at the transition point amplitude of the uncorrected response. This behavior raises or lowers the acoustic response above approximately 2 kHz and creates an unnatural result.

I examine how the HK 990 performs as it is shipped to the customer today. The developer of the EQ system in the HK 990 reviewed my measurements, and reported the problems found are real. On a hopeful note, the developer indicated the problems can be corrected with software. It is unclear if, or when a software update may be made available.

Best-Case Measurements for the HK 990

At the listening seat, I co-located my microphone with HK's. I could make measurements to 20 Hz by setting the gate time to 750 msec. With such a large gate time, the measurements are equivalent to an RTA, but with finer resolution than one-third octave. Figure 1 shows my acoustic measurements.

Normally, I would spatially average over nine points to reduce noise in the curves, but given the problems, I found myself making so many measurements with different speakers and rooms that there was no time to complete averages for each measurement. Without spatial averaging, the curves are too noisy to allow for 0.1 octave smoothing. As a workaround, I used a wide smoothing of one-third octave for the curves.

Figure 1 is a best-case result. The low-frequency room modes are well suppressed. Above 1.5 kHz the uncorrected curve and the corrected curve in the EQ2 mode match closely. The corrected curve was made with an Infinity Classia C336 speaker in the larger of the two rooms.

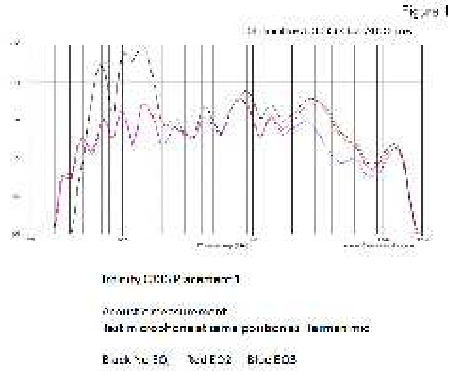
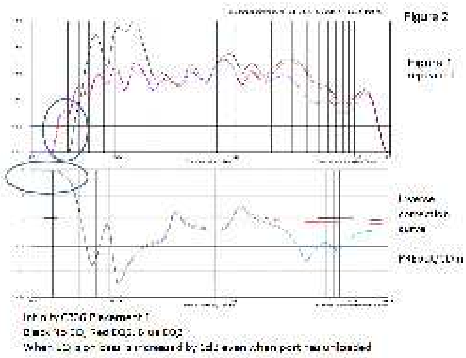
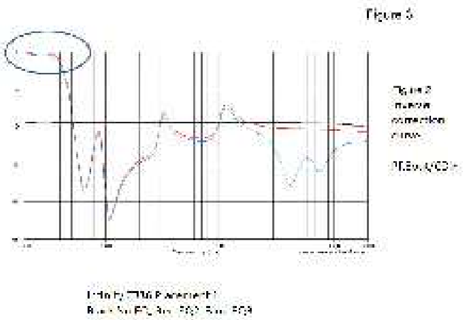


Figure 2 is the electrical correction of the HK 990 from the CD input (analog) to the preamp output. Figure 1 is repeated below the curve. At a given frequency, when the level of the preamp out increases relative to flat, the acoustic output of the loudspeaker increases by the same amount. For the HK 990, I show the corrected response for EQ2 and EQ3. EQ1 only involves the subwoofer not used here. Figure 2 identifies the 6 dB push in the electrical correction curve below 45Hz, well below the frequency that the port has unloaded for the C336. The net result is a cone flapping around when they try to reproduce a signal when the woofers are unloaded.

In Figure 2, I tried to normalize the scales of the acoustic measurements and the electrical correction curve measurements (Preamp out / CD in) to illustrate the one-to-one correspondence between the change in the acoustic response and the change in the electrical correction. Unfortunately, jamming two curves into one figure obscures the details. The electrical correction measurements alone are sized to take up the whole page in Figure 3. I show many electrical correction curves taken with different speakers in different positions. Flipping through the plots, please note the 6 -7 dB boost in most cases. Exceptions are the electrical correction curve on Figure 12, which shows a 1 dB attenuation, and Figure 15, where the boost is limited to about 1.5 dB. This boost is endemic in room correction systems unless they have special control settings (Anthem ARC for example). The room EQ has no way of establishing the frequency at which the speaker is no longer able to reproduce a clean signal at say 90dB SPL at the listening seat. The 6dB value is relatively constant across brands.

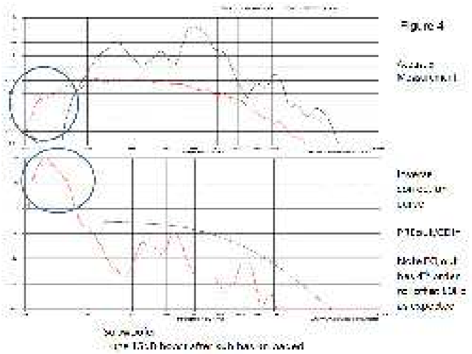




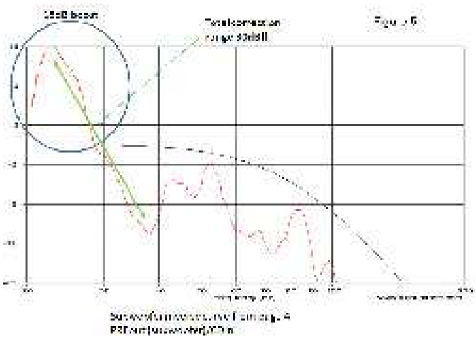
HK 990 Performance with a Subwoofer

Figure 4 follows the same approach as Figure 2, but this time for a subwoofer. Black is EQ off and red is EQ on. The room modes of the post EQ subwoofer are well suppressed but the low-frequency response (-6 dB) of the subwoofer has been shifted from 27 Hz to 21 Hz. The subwoofer has a 10 inch cone in a 12.6 inch cube. It produces high SPLs to 30Hz and then the show is over. The rolloff of the subwoofer exceeds sixth order. A 15dB boost was supplied by the HK 990 to move the low-frequency limit down to 22Hz. This is clearly seen in the electrical correction curve. The black curve is the EQ off, but 80Hz LPF bass management is in the signal path.

This boost challenged the subwoofer. The frequency response of the subwoofer changed as the test level was increased. This indicates that a sliding EQ in the woofers overload prevention scheme was being activated.(I have not shown this measurement). In Figure 4, the test tone level is low enough to keep the subwoofer operation in its linear range.



The electrical correction curve is illustrated in Figure 5. The subwoofer is also attenuated by 15 db at 38 Hz, implying a 30 dB correction range from 23 Hz (the 15 dB boost) to 38 Hz!



As this review was going to press (as we would say in the old days), I finally gained access to the designer of the EQ system. Our dialogue occurred several months after the measurements had been completed. The cause of the issue described above is a control mode found in the HK7550 video GUI that was not transferred to the HK 990. This is the subwoofer cone dimension mode, which runs from 8 to 15 inches. Subwoofer cone size is a proxy for the 6dB cutoff of the subwoofer. Subwoofer specification for cutoff frequency can be very optimistic. Subwoofer cone size is not perfectly correlated to the 6dB cutoff but, in the absence of any other reliable data, it makes sense to use subwoofer cone size to determine the -6dB frequency point. With that frequency in the processor, the electrical correction is limited to the -6dB value. It is a really smart idea since most EQ systems do not ask the question and basically try to EQ the subwoofer down to 20Hz. Like full range speakers, this limit appears to be about 6dB.

I think many of you may have guessed what happened in the HK 990. With no way to enter the critical subwoofer cone dimension, the system defaults to 15inch. It then assumes the subwoofer is good to at least 20Hz and applies electrical correction unbounded. That is 15dB in the case of the 10 inch NHT B10-d subwoofer I used. NHT gives a -6dB down spec of -27Hz. It is hard to get an exact number from the uncorrected in room curve (Figure 4) because of the room modes but 27Hz is certainly plausible. Peak output (short term) is given as 109dB at 1 meter at 30Hz. If we accept the max SPL number (something I have no way to test without a visit from the homeowners association) this is clearly a subwoofer good for full-range music reproduction provided you are not enamored with 32ft organ pipes.

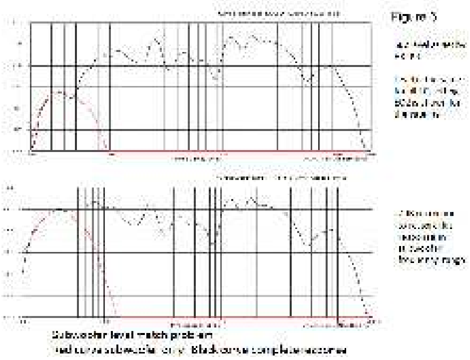
I bring this up because someone is going to question my use of this subwoofer with a state-of-the-art integrated amp. My intent was to use two, thereby allowing me to determine how well the BassQ system worked. Obviously, two subwoofers are doubly expensive as one and consume twice the space, so the NHT subwoofer appeared ideal. With something clearly amiss when I used one subwoofer, I never went on to two.

Figure 6 shows the acoustic response of the subwoofer with the level selected by the HK 990 and the level of the satellite speaker selected by the HK 990 (top curve). The subwoofer is 7 dB low. I made a correction (bottom curve) to archive a flat response across the frequency band. It is not clear how a consumer might implement this change without test equipment. I have not seen this issue with other room EQ systems when the subwoofer is properly set up, i.e., when the subwoofer level is not set too low (the HK 990 gives a warning message when this occurs). To be sure, I checked with an SPL meter to see if the level of the test tones to the main speaker matched the subwoofer. Owing to the brevity of the HK 990's tones, this test is hard to execute.

The setting of the internal crossover of the subwoofer can also create a level matching problem. Most systems like the internal crossover bypassed, but some prefer the filter to be set slightly higher than the desired crossover point. For these room EQs, the LPF of the subwoofer is cascaded with the LPF for the subwoofer in the DSP to save some DSP computations for the crossover which, in turn, can be used to correct other response errors in the subwoofer.

There is one other thing that can happen when the subwoofer crossover is bypassed: the SPL of the subwoofer is overestimated because the frequency range over which the uncorrected subwoofer is producing the test tone dwarfs the frequency range when the final LPF is incorporated. A good room correction system should compensate for this by filtering the subwoofers' spectra before calculating the SPL.

Needless to say, I tried to get the HK 990 to work properly anticipating all these potential problems, but could not make an improvement. The fact the subwoofer was set to the 15inch mode may have caused an internal SPL estimation error. The engineers at Harman could not confirm this idea, but given the 30dB correction range was introduced with the 15 inch subwoofer setting, such a problem would not be outside the realm of possibility. A revised HK 990 with a 10 inch setting should not show this problem. If it is a real problem, Harman is aware of it now and will put it on the punch list of corrections to be implemented if the software is updated.



Deep Dive to Examine the EQ 2 Setting

Figures 7 – 19 explore the problem with the EQ 2 setting. I am going to spend some time with this issue even though Harman says they can correct it with the software fix because I have seen it in other room EQs that offer the option to limit the frequency of correction.

Figure 7 is set up like Figure 2. Again, the Infinity C336 is used, but in a different position. Now the EQ 2 curve (red) does not match the uncorrected curve (black) above 2 kHz. Instead, the entire curve shifts down 2 dB. Looking at the electrical correction curve makes it clear why this happened: the red curve of the electrical correction plot has not returned to the black curve (0 dB insertion), but stays 2 dB down.

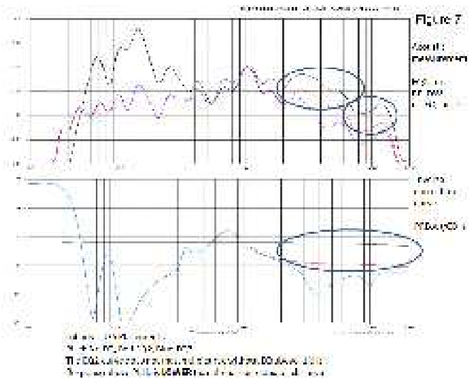


Figure 8 shows the electrical correction curve alone with more annotation on the problem.

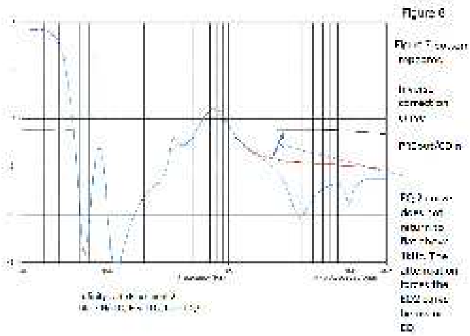


Figure 9 refers to a competitive high-quality speaker, which I call P. I am not identifying it because the single shoot measurement at one-third octave smoothing belies the speaker's true performance. It is in position 3 in the larger test room. Now the red curve in the acoustic measurement (EQ2) above 2 kHz is above the black curve by about 1 dB. The speaker brightens with this offset above 2 kHz and is clearly less flat with EQ applied. Again, the electrical correction curve shows the problem: above 2 kHz it is 1 dB above the 0 dB insertion curve (black).

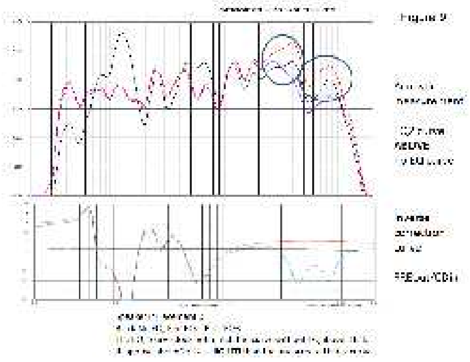


Figure 10 illustrates the electrical correction curve alone with more annotations.

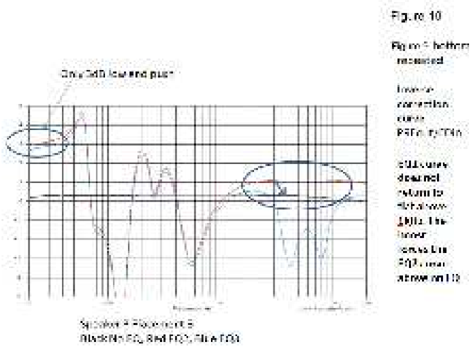
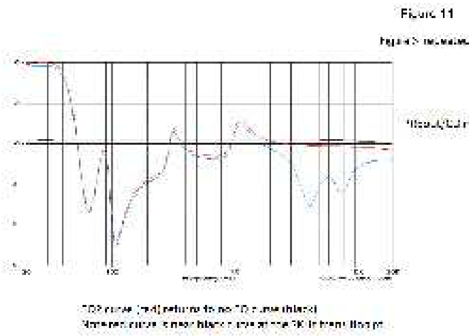


Figure 8 had signal attenuation above 2 kHz, but on Figure 10 for the different speaker in another part of the room, there is lift. Figure 2 illustrated the unit's proper operation with no loss or gain above 2 kHz. The electrical correction curve for this case is repeated in Figure 11.

Figures 11 – 15 reveal why the EQ2 curve above about 2 kHz (it slightly different for each speaker-room combination) is not always at the 0 dB reference level. These electrical correction curves involve three different speakers in two rooms with multiple placements. The EQ2 curve above about 2 kHz takes on the amplitude of the correction curve at about 2 kHz. This is about flat for the graph in Figure 11, so the red curve remains at the 0 dB line.



In Figure 12, note the electrical correction curve which is up more than 3 dB at 2 kHz and the EQ2 curve remains 3 dB hot above 2 kHz.

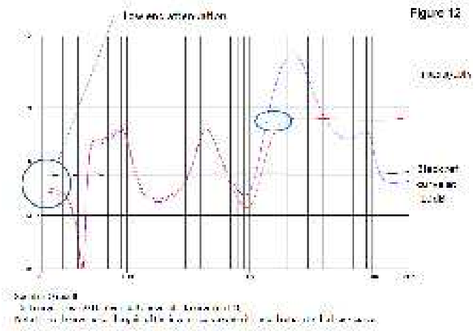
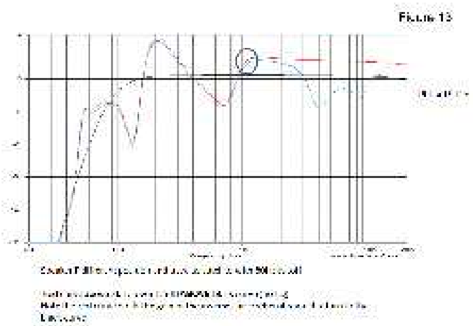
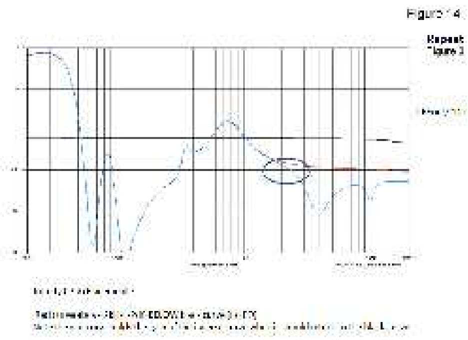
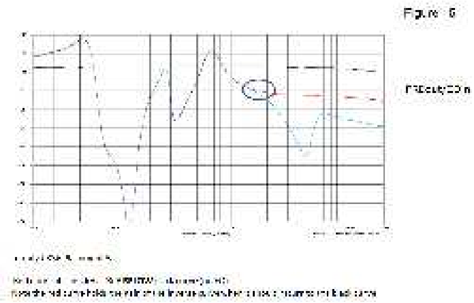


Figure 13 highlights a 1.5 dB shift above unity gain. The level of the correction curve is matched at 1.2 kHz. The respective lifts of Figures 12 and 13 brighten the speaker.



Figures 14 and 15 are electrical correction curves where the correction curve is below unity gain at 2 kHz. The EQ2 curve above 2 kHz now shows a loss of about 3 dB for both cases. The cut in gain in Figures 12 and 13 dulls the speaker.





Understanding the Problem with EQ2: An Alternate Perspective

The Anthem ARC room EQ (Figure 16; a full set of measurements will be presented for this system soon) has a variable max EQ frequency. I set it to 700 Hz, which is consistent with the peak of the uncorrected curve. The acoustic frequency-response plot from the Anthem ARC display is different than the image Acoustisoft RPlusD produced in prior Figures. The red curve is the uncorrected response (9pt average at one-sixth octave smoothing). The green curve is the corrected response. Note how the curve moves to match the uncorrected response curve (red) around 700 Hz. Above 900 Hz, the corrected curve tracks the uncorrected curve (red on top of green). For this to occur, the electrical correction curve must be at unity gain above 900 Hz (not shown).

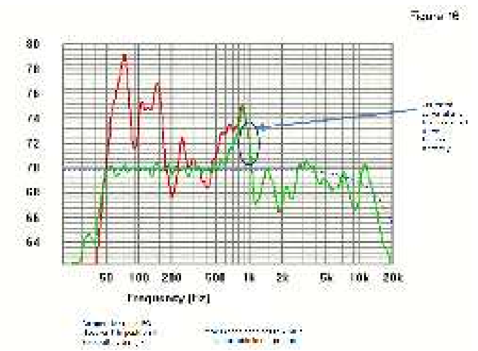
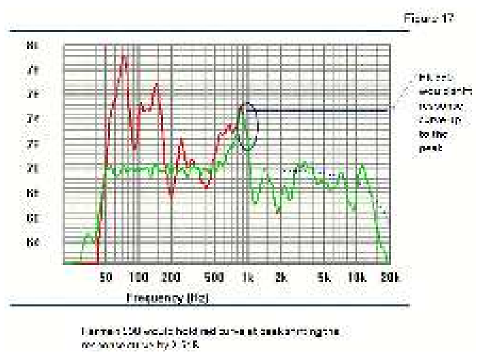


Figure 17 illustrates the response of the HK 990 to this condition if its maximum correction frequency were 700 Hz. The HK 990 would brighten the speaker’s sound by setting the level at the peak, which is about 2.5 dB.



With the same speaker and the maximum frequency of the Anthem ARC at 1.5 kHz, there is a dip (Figure 18). The Anthem ARC gravitates to the dip (red on top of green), signaling a shift of electrical correction curve to unity gain above 1.5 kHz.

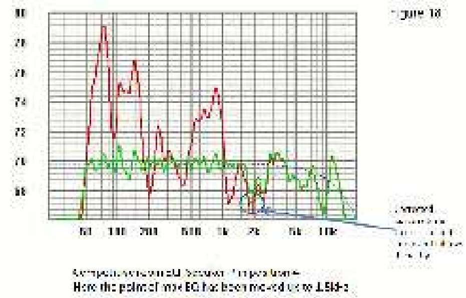
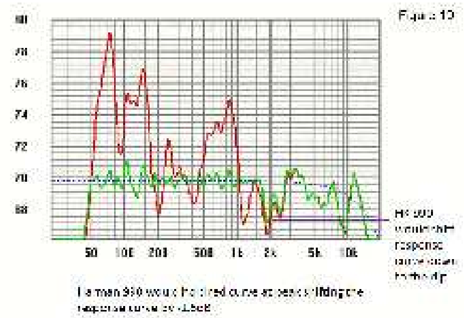


Figure 19 shows what the HK 990 would do. The curve above 1.5 kHz matches the level of the dip and the electrical correction curve is 1.5dB down above 1.5 kHz.



The EQ 3 Setting: A Detailed Interpretation

Looking at the curves in Figure 10 - 15, it can be seen EQ3 (blue curves in the measurements) is reducing the top-end of the speaker, above 1 - 3 kHz depending on the speaker and room placement. The setup for figure 12 is the exception. EQ3 made the speaker sound different than its intrinsic voicing It is a matter of preference whether EQ3 is an improvement over no correction.

Since EQ3 appeared to be working properly for full range speakers from the data above I committed time to a nine-point average for the Infinity C336 in two new positions. The results pre- and post-EQ3 are shown below in Figures 20 for the speaker in position 4.

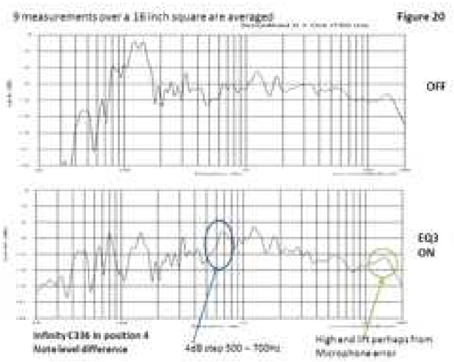
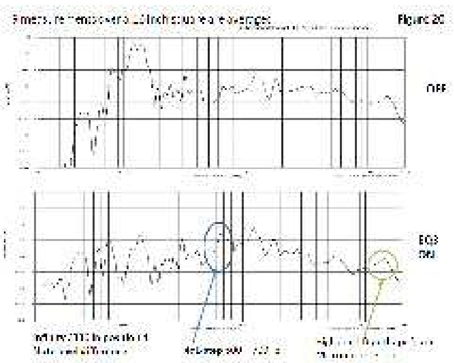


Figure 21 below is the pre- and post-equalized response for the speaker moved to position 5.



The figurers 20 and 21 illustrate the flat response of the Infinity C336 above 400Hz before EQ is confined to +/- 2 dB. Below 400Hz, my room dominates the response; however, once EQ3 is activated the response below 400Hz improves significantly. This is not the best I have seen at this price point, but it is close. Also note the push of the low end as was observed in the earlier Figures. This time the response is pushed down by about 15Hz as a result of the 6dB maximum lift in the electrical correction curve.

Harman engineers were not able to address whether EQ3 was performing properly. The step starting at 500Hz suggests a problem matching the near-field correction curve to correct variations in the speaker's response to the part of the curve corrected for errors due to the room. They also could not confirm if the downward slope above 1kHz is the desired result.

There is a lift above 8kHz at the post correction response. I expect, but cannot confirm, if this is the result of the microphone's top end response dropping. The system cannot tell a drop in level at the microphone from one occurring from the speaker. Accordingly, the software infers the speaker needs some lift in the high end. I have seen this problem in other room EQs with small un-calibrated microphones.

I am not overly concerned about the EQ3 issues. EQ3 is designed to correct a poor response curve (rough or significant octave-to-octave variation) of the speaker and it is unlikely someone would spend $2600 on the HK 990 and match it to bad speakers. Not all speakers that win the "best of the year" awards are flat on-axis. This is a result of the designers voicing of the speaker and how the designer handles the difference between the on-axis and off-axis responses. EQ3 removes the designer's voicing. I advise limiting the correction to a 500 -1kHz limit and leaving the voicing intact in stereo. Multichannel has a different set of issues and may benefit from full range correction. When working correctly, the EQ2 mode is a significant value-add for the HK 990 over a system that just does full correction.

Subjectively (and this is my sole subjective observation), the correction below 400Hz before the bass push is similar to my preferred EQ systems. The push in the bass was not as artificial sounding as I have heard with AVRs whose EQ systems also attempt to flatten the frequency response below the port frequency. The AVRs' amplifiers did not have the current sourcing ability of the HK 990. The parallel configuration of the Infinity C336's three woofers makes it difficult to drive when the gain is up 6dB. To confirm this observation, the distortion of the speaker in the area that the room EQ has pushed the bass must be measured. In my lab, these measurements are not feasible owing to my combined problem of high ambient room noise at low frequencies and close-in neighbors with little tolerance for 35Hz test tones. In any case, the C336 speaker sounded better with amplification when used with the Anthem ARC room EQ that prevents bass-push. Since the Anthem ARC is PC-based, post-corrected bass response can be panel controlled.

The EQ off is not normalized to EQ3 as can be seen be in figures 20 and 21. The off mode is about 2dB higher. You thus need to adjust the volume control to normalize the levels. This increases the time it takes hear the sonic change when EQ3 is activated.

I will leave it to [Tyler to tell you if this target curve is pleasing out of the room-dominated area.]

Conclusions About HK 990 Room EQ

With the addition of digital room-correction, the HK 990 travels into unique space for a stereo integrated amplifier. As a dual-domain device, its internal electronics are conversant with analog or digital inputs, including high resolution files. Recording from digital and analog sources is simplified. In addition, the HK 990 has no mechanical controls for improved reliability, improved gain tracking, and increased stereo separation. Unfortunately, the room EQ does not function properly in many operating modes. We can hope Harman patches the code. The engineers at Harman clearly understand what needs to be changed. It remains hazy whether these changes, if made available, will be available to products already sold, available only with a change at a service department, or something that can be done in the home. The timeline for these fixes also remains an open issue.

In Part II of the series, I examine dual-domain signal flow of the HK 990 at the block-diagram level. Those interested in AVRs may want to continue to Part 2 since the blocks are very similar. It is much easier to read a diagram in two channels instead of eight.

Tags: Amplifiers | Integrated Amplifiers | Stereo

Comments (5)

pics , pricing
written by Donnie , October 22, 2011

I'd like to see interior pics , especially at this price point. Also I don't want to have to read so far down the page to find the price.
Seems like a great piece from a company with the potential to make great gear.

Re. pics , pricing
written by David Rich , October 24, 2011

Donnie,

Tyler Stripko published pricing, specifications, setup, use, and listening impressions in June. Refer to the link at the beginning of this review. I have two upcoming parts to be posted October 27 and November 3. The next part begins with an interior picture and examines the unit's configuration at the block diagram level. More details about the contents of my review are at the bottom of page 3.

Thank you
written by Vinh , October 24, 2011

I am absolutely astounded! A technical review that offers insight into the deficiencies of a product?

Simply outstanding! I haven't read a review this good in years. This is why I visit this website every month or so, in hopes of finding the rare article that contains actual substance.

Thank you, thank you, thank you!

2-channel receiver with room EQ, almost complete...
written by Maarten , October 24, 2011

I also use a multi-channel receiver in a stereo setup. My Audessey takes care of speaker EQ, including the subwoofer and crossover.
A stereo component with room EQ is great. What I am missing however, are HDMI inputs that accept DSD streams and the latest high def surround formats. I hope more companies will make stereo receivers with room EQ. Good informative review by the way! I look forward to the sequels.

Re. Thank you
written by David Rich , October 25, 2011

Vinh

Your words are much too kind. My acoustic measurement tool (AcoustiSoft RplusD) is a primary reason I can detect problems and clearly illustrate the issues to the reader. The versatility of the low cost AcoustiSoft RplusD, written by Doug Plumb, is exemplified in the review.

Please do not stay away for a month. There is a wealth of a material on the site similar to this review. The Secrets Blu-ray Player HDMI Benchmark series is a good example. Part 2 of this review is scheduled to appear on Thursday.

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Harman Kardon HK 990 Stereo Integrated Amplifier with Digital Room Correction and Dual Subwoofer Bass Management – Part II

Written by Dr. David A. Rich

Thursday, 27 October 2011 00:00

Article Index

Introduction to the Harman Kardon HK 990 Stereo Integrated Amplifier with Digital Room Correction and Dual Subwoofer Bass Management – Part II

This second part of the HK 990 review examines the signal flows in the HK 990 dual-domain integrated amplifier. This analysis is relevant to all units, including modem AVRs, with a DSP signal processing unit in the main signal path. The signal flow is more transparent in this stereo unit than a 7.1 multichannel unit; so, if you are interested in AVR, please stay tuned in to Part 2.

Analog circuit design methods for consumer high-fidelity products date to 1950 with the emergence of reel tape, LPs, and FM. Consumer-friendly discussions of analog component design are found in early (circa 1947) issues of Audio Magazine and early (circa 1951) issues of High Fidelity. Digital recording and CD players arrived at a time that magazines were still somewhat technical. It was not until Dolby Digital (originally called AC-3) started coming off Laser Discs in the mid-1990's that units with core DSP processors appeared. By that time, the print magazines had de-emphasized technical aspects of design, preferring to focus on measurements results at the terminals of the DUT.

Specifications

- Design: Solid State Stereo Integrated Amplifier
- Power: 2 x 150 watts RMS into 8 ohms @ 20 Hz - 20 kHz, 2 x 300 Watts into 4 Ohms
- MFR: 10 Hz – 100 kHz
- THD: <0.07% at Full Output (8 Ohm Load)
- Analog Inputs: 7, Plus 1 Phono MC, 1 Phono MM, and 1 Balanced XLR
- Digital Inputs: 1 HRS-Link, 2 Optical Digital, 2 Coaxial Digital
- Analog Input Sensitivity/Impedance: 350mV/43k ohms for tuner/CD, 10mV/47k ohms for Phono-MM, 1mV/100k ohms for Phono-MC
- Digital Input Capability: All Standard Digital Formats
- Dimensions: 6.4" H x 17.3" W x 17.5" D
- Weight: 43.2 Pounds
- MSRP: $2,599 USA
- Harman Kardon
- SECRETS Tags: Harman Kardon, HK 990

Part II of this series examines the dual-domain DSP based system at a deeper level than the treatment found in most reviews, but it is not so technical as to leave an interested audiophile puzzled.

If you missed Part I......

Click Here to Add or View Comments

HK 990 Looking Inside the Unit

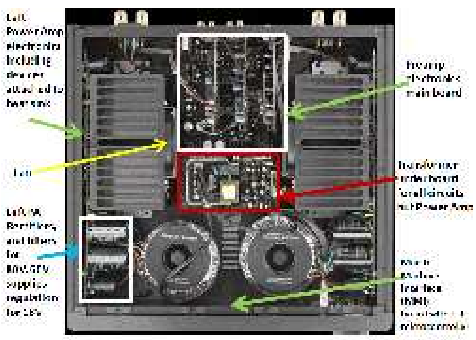


Figure 1 shown above is the HK 990 with its top removed. The key subcomponents are identified with notations along the perimeter of the photo. The power amp (to be discussed in Part III of the review) and its associated power supplies monopolize most of the amplifier's footprint.

At the front of the unit is a PC board with the control buttons, the volume control-knob (a rotation-sensing device that sends a digital signal when the knob is turned), and the display panel. The electronics on the board, including a microprocessor, are the man–machine interface (MMI). Upon sensing a change on the panel, such as a depressed button, the PC board converts the action to digital data that guide the internal electronics of the

HK 990. In turn, the internal electronics respond with digital data to control the display so you can tell what the unit is doing.

The preamp section has one large board at the base of the unit and four daughter boards (obscured in the photo for lack of resolution) positioned at right angles to the main board. This configuration shoehorns the maximum electronics in the minimum amount of space. Surprisingly, the phono board is as large as the mixed-signal boards. Almost all digital interconnection from the front panel run to the preamp section.

The vertical placement of the daughter boards allows external I/O jacks at the rear to be mounted directly on one of the boards. From the rear of the unit, shown in figure 2 below, the locations of three of the boards are apparent. The phono board has more jacks than anticipated because the analog tape recorder connections are also on the bottom of the phono board.

The main board interconnects the daughter boards. Almost all the interconnections among the boards are digital. A second on-board microprocessor, the digital volume control, and the preamp outs are also on the main preamp board. The second microprocessor coordinates the operation of the digital circuits on the daughter boards as well as interfacing to the front panel.

The on-board position of the preamp outputs puts them in very close proximity to the volume-control circuits. You can see the preamp outputs at the bottom of the rear panel. The preamp output jacks define the position of the horizontal main board. Imagine all the wires in an analog integrated amplifier running between the front and rear panels. This would manifest itself in measurements as degradation in crosstalk or a high-end rolloff.

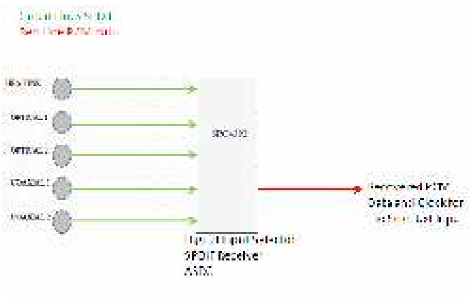
Those who find wire to make a sonic difference often forget how much of it is running inside a piece of electronics. Worse, each of these wires terminates with connectors that slide over pins on the PC board. Those concerned about the quality of an RCA plug and jack would be disappointed to see these connections. The ability to remove the wires from the PC board is of necessity because they must be removed to service the interconnections and/or allow replacement of a defective PC board. Yards of wires, including long traces on PC boards, are not needed when the MMI is up front and the analog signals as proximate to the rear panel jacks.

The main board also contains the power supply regulators. Multiple regulators isolate the different circuit elements. A few sub-regulators are resident on the daughter boards and power some components. With all the power supply regulation on the preamp boards instead of a separate power supply board, there is little wiring of the regulated supplies to the analog components. In front of the main board is its power transformer, which is independent of the power amps.

The big takeaway is the brevity of the analog and regulated power supply interconnections. In addition the long interconnects and multiple RCA jack – plug connections required if you used separate DAC, phono preamp, or power amp are eliminated. In the HK 990 no analog signal leaves from the area of the main preamp board to keep analog connections short. The sole exceptions are the connections to the power amps and headphone jack.

Click Here to Add or View Comments

HK 990 Digital Signal Selector, Clock and Data Recovery, Jitter Reduction, and Digital Reconstruction Filtering

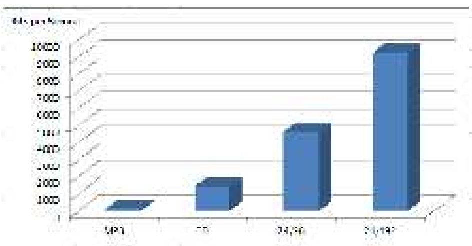


The block diagram of figure 3 above shows the digital input signal flows. The electronics are on the right-most daughter board. Its placement is defined by the optical and coax digital input jacks on the rear panel. The clock and data from the Sony/Philips Digital Interconnect Format (SPDIF) digital inputs are recovered by a TI SRC4392. The chip also functions as the input selector for the digital signals. In an AVR, digital audio data from the HDMI receivers can be selected in this system block. The HDMI receivers are usually on the video board, which is typically at the top of the audio boards in an AVR, which explains the placement of the HDMI plugs atop the rear panel in most AVRs.

The “Phase-Locked Loop” (PLL) is a basic circuit block for the clock and data recovery. Although the SPDIF signal is a serial bit stream, it is coded so the continuous clock signal and the signal that contains the LPCM data can be recovered and separated (the clock and data are shown together in the single thick red wire.). A PLL is on the die of the TI SRC4392. The mechanics of a PLL, its application in a clock and data recovery system, and its ability to reduce jitter, are best left for another discussion. Many texts have been written on Phase-Locked Loop design in both the analog and digital domains. I will try to summarize the essentials in the Primer at some point.

A two-channel (for stereo) asynchronous sample-rate converter (ASRC) resides on the TI SRC4392 to resolve an issue that the recovered clock from the SPDIF signal is not in sync with the clock that drives the DACs. The ASRC also provides digital filtering to remove signals above 20kHz (for the case of a CD) that occur during the sampling process. I discuss ASRCs in more detail below.

A. Maintaining High-resolution Data Through the Digital Signal Selector Front End



1) Bit depth

Figure 4 above shows the audiophile currently has a wealth of high resolution downloadable formats. Whether a modern unit with SPDIF inputs accepts and recovers 24bit incoming data from high-resolution material (DVD-A disc; WAV file; FLAC file) is a less than transparent issue, but one that is significant. Perhaps, more important, is the truncation of the signal as it makes its way from the SPDIF receiver to the DACs. To determine if the 24bits are preserved, the amplitude of a dithered signal is examined as it is reduced from -90dB below full scale (this is called the amplitude linearity test). The in-band signal-to-noise ratio (relative to full scale) is also monitored during the test. The test should occur when the complex DAC process options, such as room correction and bass management, are selected.

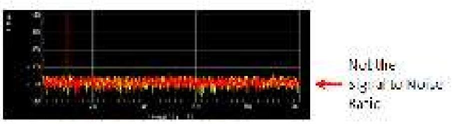
For all cases, the signal-to-noise ratios and amplitude linearity are limited by the DAC in the digital path if the 24bit word makes it to the DAC. The product sheet on the HK 990 is unclear about transmission of the 24bit words, but Harman Labs sent me data that indicating the 24bit signal was passed with the DSP in the signal path. The measurements covered the complete analog signal path to preamp out, including the volume control.

The HK 990 measurements from Harman labs showed the equivalent signal-to-noise ratio was 19 equivalent bits. A 21bit signal had a -0.7 error relative to its expected amplitude. At 22bit word (-132dB from full scale) had a -1.3dB error. These results are consistent with the performance of the parts used in the HK 990 which will be discussed below.

The signal-to-noise (SNR) ratio of a signal recorded with a 16-bit quantization (98dB) cannot be lowered by signal processing on the received data as some high end companies imply. The number of bits that represent the signal grows as the signal progresses through the digital filters and other digital signal processes. The additional bits occur as mathematical operations are performed on the data. Failure to keep these additional bits may cause distortion and frequency response errors.

Some CDs are recorded with shaped quantization noise. The noise is shaped by removing it from areas of the spectrum that the ear is most sensitive to it. Since the total noise power over the full band must be constant, more noise power is placed at areas of spectrum where the ear is less sensitive to noise. Quantization noise can be shaped only during the recording process. Noise shaping must be done before the data is pressed on the CD. If a noise shaper was not used the process cannot replicated from data coming of the CD.

The noise floor of fast Fourier transform (FFT) developed spectrums shown in audio equipment reports is not the signal-to-noise ratio. The noise floor is the noise over a small frequency band. The size of the band depends on parameters of the FFT. This is shown in Figure 5 below.



2) Sampling Rate

Both the bit depth from the SPDIF digital input and the sampling rate of the high-resolution input must be preserved. The HK 990 accepts inputs with sampling rates up to 192kHz. Like the bit depth, the issue is not whether the SPDIF receiver locks-up to the incoming signal, but the quality of the signal received by the DAC. Activating a function such as room correction may result in sub-sampling (96kHz sampling reduced to 48kHz) in some products.

To identify sub-sampling, the spectra at the preamp output are monitored as a high-resolution file is connected to the input. Sub-sampling has occurred when the spectra above 20KHz are missing.

Again the product information for the HK 990 does not provide the critical information but data from Harman Labs, showed the HK 990 passes the full spectra (45kHz) of a 96kHz input, but this did not increase for a 192kHz input. It is hard to believe anybody could discern the difference.

The SPDIF output mutes on any SACD player when an SACD is inserted. Instead, one must use the analog output on the SACD player and run it to the HK 990 analog inputs. This is not an issue if you use the HK 990 in the digital bypass mode. If you want to enable room correction or bass management the HK 990 must first convert the analog signal from the SACD player back to digital.

An AVR attached to a Universal DVD player by HDMI will transfer a SACD digital data (DSD or transcoded to LPCM) since HDMI encrypts digital data. But dealing with video setup screens on both the AVR and Universal DVD player to enable the link can be complicated, and stereo-centric audiophiles will find this task tedious beyond belief. You need a TV in the room to set things up. The TV can be removed after setup, but may be called back should either the universal DVD player and/or AVR forget its settings as a result of a glitch like a power outage.

The fact that the operability of the HK 990 is not display-dependent is what makes it so wonderful. If you are going to use the room correction or bass management of the HK 990 the best thing to do is use a good CD player and not an SACD player. The CD player will read the CD layer of the SACD and transfer that over

SPDIF. The redundant DAC –ADC conversions are eliminated. Given the inability to get high resolution data off an SACD using SPDIF it is clear that high resolution downloads are the way to go in stereo. SACD is important only to the multichannel listener.

B. The Performance and Viability of Proprietary Digital Interfaces

Harman's proprietary digital interface to its CD players, called the HRS link, provides a return path for the HK 990 system clock to the Harman HD 990 CD player (not shown in the digital input signal flows block diagram above). Tyler Stripko reviewed the HD 990. Figure 6 below shows the special HRS Link multi-wire jack.

As discussed above, digital devices with an SPDIF transmitter have clocks that are asynchronous to the main crystal oscillator on the HK 990. It is impossible to match the crystal oscillator frequency on the device transmitting the SPDIF signal to the crystal oscillator on the HK 990. The HRS link resolves the problem for the HD 990 CD player by forcing the HD 990 CD player to slave the HK 990 clock.

The crystal clock oscillator in the HD990 CD player is disabled when the HRS system is operative. Jitter on the HRS return path is not an issue because the clock that drives the DACs is not recovered from it. The DACs are clocked with the crystal oscillator on the DSP board, which is very close to the DAC.

These types of proprietary links emerged when SPDIF first appeared. Sony's first solution was an extra SPDIF cable running the clock back to the CD player. Unfortunately, comparable systems of different manufacturers are not fungible. Sony no longer supports the extra SPDIF cable for clock return. Marantz has a BNC input on some of its high-end SACD players marked External Master Clock.

The latest Sony proprietary-link system is called H.A.T.S. for its HDMI connections. The HDMI 1.3 CEC return line controls the data transmission rate to the universal player based on the clock in the AVR. You need a Sony AVR and Sony Blu-ray player that supports H.A.T.S. Pioneer has a nearly identical, but not inter-operative, system called PQLS. Both H.A.T.S. and PQLS are derived from the Audio Rate Control (ARC), which is a new function in version 1.3 of the HDMI spec. Games of deviating from the HDMI spec to create proprietary modes of operation caused big box stores to insist on absolute HDMI interoperability between all suppliers. This appears to have wiped the Audio Rate Control (ARC) off the latest generation of products.

Wedding yourself to these proprietary links can create long term problems. Disk players often require expensive repairs after warranty up to the point where unit replacement is more economical. One is unlikely to find a new unit that supports the old proprietary link. Also consider the circumstances of audiophiles with high-end universal DVD players that had proprietary links when Blu-ray emerged.

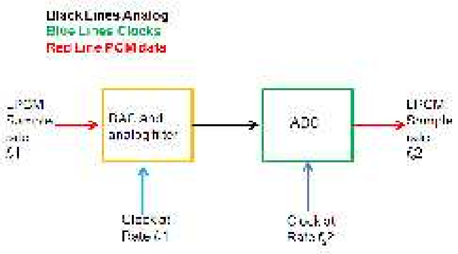
Also note that, most units that store and stream high-resolution files only have standard SPDIF outputs. The takeaway is to be certain the equipment you purchase has electronics that insurer the SNR and THD of the analog output is not degraded by impairments of the equipment sourcing the SPDIF signal or impairments from long runs of the SPDIF cable.

C. Working with Asynchronous Clocks without a Proprietary Link in Systems with DSP Processors

As discussed above the clock frequency of a unit with the SPDIF receiver differs from the clock in the unit with the SPDIF receiver. One solution to the problem is the proprietary link. A second alternative is to clock all electronics in the unit with the recovered clock from the source device. The recovered clock must have very low jitter when it clocks the DAC or SNR and THD will be degraded. Methods are available to achieve this. Note direct measurements on the clock sent to the DAC in the presence of jitter at the input can guide the designer in optimizing the system. It is possible for a reviewer to open up the box and make these measurements but I strongly dislike this. Any effect of clock jitter should be observable at the analog outputs of the unit in the form of increased noise or new distortion components.

In units like the HK 990, and most AVRs, the digital electronics in the unit (often multiple DSPs) are clocked by a local crystal oscillator. A digital circuit block, called an Asynchronous Sample Rate Converter (ASRC), is used. The recovered clock and LPCM data (one channel) from the SPDIF receiver enters the ASRC. In addition, the local clock oscillator on HK 990 or AVR, enters the ASRC.

The ASRC processes the incoming LPCM data and the LPCM data leaves the chip is not the same as what came in. To understand this, let us look at a conceptual ASRC. An ideal DAC is clocked with one oscillator. The DAC, after reconstruction filtering, drives an ideal ADC clocked by a different oscillator. This conceptual ASRC is illustrated as a block diagram in Figure 7 below.



Note that the ASRC does not produce a clock it just provides a mechanism to interface the LPCM signal between to clocks that already exist. Clearly, the LPCM data entering the ideal DAC is not the same as what is exiting the ideal ADC if the clocks are different frequencies.

The ASRC can take in a clock with jitter on the LPCM input side and a jitter-free clock on the LPCM output side. Does this not imply all the jitter has been rejected. No! The effects of the jitter (increased SNR and distortion above what is measured with jitter-free clocks) may be present on the LPCM data exiting the ASRC. The worst-case situation is the mixed-signal example shown above. The DAC rejects none of the jitter on the clock driving it All the distortion and noise resulting from the jitter is now part of the LPCM data leaving the ADC.

ASRCs, in practice, are fully digital and have no internal DAC or ADCs. A fully digital ASRC interpolates (up samples) the LPCM input data to a very high sampling rate, which is then re-sampled at the output clock's rate. The system just described is impractical to integrate. Digital designers have developed systems that can emulate the functionality in less silicon.

Artifacts from the design limitations for a practical all-digital ASRC are small frequency response variations and distortion components in the LPCM data at the output. These arise in the process of the sample-rate conversion (no jitter on the clocks). The distortion components may not be harmonically related to the incoming tone encoded in the LPCM. The distortion and frequency response deviations are called out in the ASRCs data sheet for standard products like the TI SRC4392 used in the HK 990.

Distortion and frequency response variations can be measured on LPCM coming out of the ASRC; indeed, modern analog test equipment first converts the signal to LPCM before analysis. Different graphs are supplied for common sample rate conversions in audio products.

A practical fully-digital chip ASRC will reject some jitter. A section of the digital circuitry estimates the frequency ratio of the two clocks. The rate at which this estimate can change is limited. Robert Adams, who created the first ASRC for an audio application, points out the estimate is “computed using thousands of past input and output sample clock events and is therefore immune to small perturbations in the arrival time of any clock edge.”

The ratio estimate is less affected the faster the clock edges arrival time changes. Some ASRC data sheets offer a graph that demonstrates how fast the clock edge arrival times must vary for the ratio estimator to become insensitive to it. This graph cannot replace graphs showing SNR and distortion increase with different types of jitter present on one of the clocks. I have not seen an ARSC data sheet with these measurements.

Without a SNR and distortion measurements at the output of the ARSC in the presence of clock jitter at one of the clocks, a relative evaluation of two ASRC designs is not possible, and I cannot identify from the current data sheets which ARSC is best with respect to jitter rejection. Valid comparisons between standard products are obscured even further because the designer customizes the ASRCs performance by selecting internal options.

Murkier still, there are no standard tests that simulate the statistics of jitter that might be on the recovered SPDIF clock presented to the ASRC. First the test clock must simulate the statistics of jitter of the oscillator that is part of the PLL in the SPDIF receiver. At the SDPIF receiver recovered clock jitter statistics can change dependent on the LPCM data being transmitted (recall the clock and data are encoded on the single SPDIF cable) This is called data dependent jitter. Some LPCM test patterns have been proposed that are said produce worst case data dependent jitter but no consensus exists on which patterns are worst case. Recovered clock jitter will increase as the SPDIF signal passes through longer lengths of cable. Again this jitter may have different statistics.

For those who want to know more about jitter, how it affects DACs, and how an ASRC operates, please consider an article written by Robert Adams. It is published on pages 11 -22 of Issue 19 (Spring 1994) of The Audio Critic. The complete issue is available as a free PDF download on The Audio Critic website.

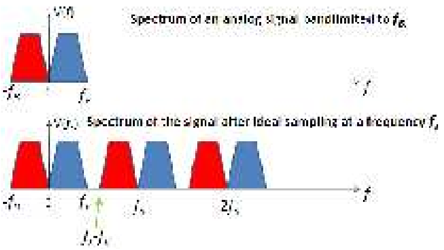
ASRCs may be found as part of a DSP chip (eight are in the Analog Devices ADSP-21469 designed for use in AVRs). An ASRC can be executed as software and implemented as an independent DSP or a DSP that performs multiple processing functions.

The service manual provides clues about the performance of the analog and data converter stages that can predict the performance when device measurements are made at the units rear panel as I will demonstrate in Part 3 of the review. In contrast, the parts used in the product provide no clue to the jitter rejection. Only special measurements at analog outputs will expose how much the jitter is attenuated. Without standardized tests I have chosen not to report on SNR and distortion impairments at the analog output as a result of jitter on the SPDIF input.

D. Digital Filtering of the LPCM Signal

An ASRC may provide the LPCM data at its output at a higher sampling rate than what came in. For example, LPCM data from a CD player arrives at 44.1kHz input, but could leave at 192kHz.

Artifacts from the sampling process, called images (shown in figure 8 below), are removed during the process of increasing the sampling rate.



For a band-limited 20kHz analog signal sampled at a rate of 44.1kHz samples/sec (fs) the spectra appear around 0Hz (desired signal) as well as from 24.1kHz to 64.1kHz and then repeat around multiples of 44.1kHz. The repeating spectra are the stop-band images from the sampling process. For a 20kHz signal (fB), the right side of the first image (often called the folded frequency) is close-in at 22.1kHz (44.1kHz – 20kHz). The suppression of the stop-band images allows the sampled signal to better represent an analog signal sampled at 44.1kHz. Information lost from band limiting of the signal before the sampling process during recording cannot be restored although some manufactures try to imply this. For high resolution files fs will be between 88.2 – 192kHz. The maximum frequency that can be recorded (fB) can also be extended since the right side of the first image will be at a higher frequency.

How well the ASRC removes the images is again dependent on its design. The process of attenuating the stop-band images of a sampled signal and increasing the sampling rate digitally often takes on interesting names from each company. For the HK 990, it is named the fourth-generation of the Real Time Linear Smoothing system (RLS IV). To repeat although the sample rate has been increased, no information about the sampled signal above 20kHz (for the case of a CD player) can be recreated. Instead, what has been accomplished is the removal of the folded tones in the digital domain to obviate removal later by complex analog filters.

In the absence of an ASRC a standard digital filter up-samples the incoming signal at an integer rate of 2, 4 or 8 and removes the stop band images. The filter may by as implemented as DSP software or it may be done in digital hardware. Like the ASRC, the filter leaves small frequency response variations and distortion products (folded tones not completely removed, for example). The design of the filter determines how well it performs in the frequency and time domains. Sometimes a standard digital filter may be placed after an ASRC to reduce the stop band images further.

Often the digital filter used is in the DAC chip itself, not an external block. The internal DAC filter is bypassed by the HK 990 in favor of the ASRC and perhaps additional filtering in the DSP block. The performance of the filter in the integrated circuit DAC may vary. Filters with poorer performance have more amplitude variation across the audible band, although this is normally dwarfed by the variation from the analog circuits. The other difference in filter performance is the attenuation of the folded spectra.

In general, the performance of the digital filter in the integrated DAC correlates with the THD and SNR performance. For superior performance in the mixed signal domain, more silicon area is made available for the digital filter because the improvement in analog noise floors can expose problems with the simpler digital filters.

Designers may design custom ASRCs and up-sampling filters to gain a performance advantage over generic products. Typically, the designer will do this in software using a standalone DSP. Some designers develop the filters to achieve performance objectives in the time and/or frequency domains. The custom DSP implementation is the most expedient way in which a designer can distinguish the performance of his product from the competition. This assumes a DAC has been selected with state of the art performance. In addition analog components connected to the DAC must be selected so they do not degrade the performance of the

DAC. The Harman HD 990 CD player used proprietary software for the ASRC and up-sampling filter. Harman calls the system RLS III. An Analog Devices Blackfin DSP (Package photo shown in Figure 9 above, ® Analog Devices) was selected to execute the custom DSP code.

Like jitter, folded tone suppression must be measured. Examining the unit provides no clues, especially if the filter has custom crafted code for a DSP. Unlike jitter standard tests exist. For example a full scale 20kHz LPCM signal on a CD.

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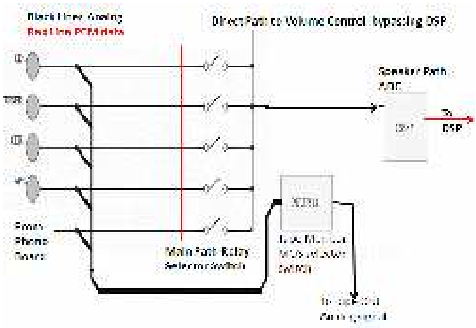
Analog Input Signal Flow of the HK 990



The electronics represented on the block diagram (Figure 10 above) are located on the left-most daughter board as dictated by the placement of the analog inputs on the rear panel.

Relays, not silicon switches common to most AVRs, drive the function selector for the analog inputs. As can be seen, the signals from the phono board are also routed to the relay bank. Each relay is a fraction of an inch from the analog input jacks on the back. The MMI at the front of the unit activates a relay based on the user's selections. In the direct mode, the output of the relay is connected to the digital volume control. Some posters in the AVS forums have been complaining about the clicking sound of relays in higher-end equipment. They should stop and think: does one just want the signal to be transmitted across a metal strip or would they rather the signal was transmit across a MOSFET transistor with significantly higher resistance that varies with the incoming voltage. Relays and the associated drivers are more costly than a silicon switch. Reliability of a relay and the associated driver circuitry is the only downside.

For DSP functionality to be enabled, the analog inputs must be converted to digital by an Analog-to-Digital Converter (ADC). In the HK 990, the ADC is a Cirrus CS5361, one notch down from top of the line Cirrus ADC. The bit-equivalent signal-to-noise ratio (SNR) is 17.5 bits worst case (A-Weighted). The bit-equivalent distortion is 16.5 bits worst case at 1kHz with an input signal 1dB below full-scale.

Even on expensive AVRs, the ADC is often a low-grade single-ended part, a kludge by AVR designers who assume the only analog signals entering emanate from cassette decks and VCRs. They completely forget about converting a phono signal, which needs a high-quality converter to prevent sonic degradation.

The tape output path has its own function selector and ADC. This supports the option of listening to one input and recording another. I elaborate upon the tape output path in Part 3.

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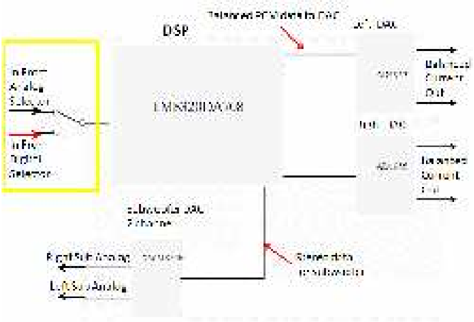
HK 990 Digital Signal Processing and DAC Block



The core of the HK 990 is the digital signal processor (DSP) and DAC that converts the DSP's computations to analog. The function shown in the block diagram directly above (Figure 11) is on the same board as the digital selector block discussed previously.

The switch at the left selects the LPCM data from the digital input board or the analog input board. In the case of the analog input board, the analog signal was converted to LPCM by the Cirrus CS5361 ADC before arriving at the switch.

The DSP in the HK 990 is a TI Aureus TMS320DA708. For room correction, the DSP has two different functions to perform. First, the room must be measured. The DSP generates the test signals to be sent to the speaker, records the resultant acoustic response of the room sent to the microphone (in the room-calibration mode, the microphone signal is routed to the ADC on the analog board), and generates the filter coefficients for room correction. Room-correction systems with on-board DSP can take several minutes to calculate the coefficient because the DSP is not designed to execute these types of off-line computations efficiently. The microcontroller on the main board is also involved in coordinating the room calibration process and transmits information to the MMI board so the user knows where to place the microphones and when it is safe to move them to a new position.

In normal operation, the incoming digital signal passes through the filter bank that is loaded with the room correction coefficients calculated as explained above. A significant part quality of the room correction depends on how well the coefficients are determined during the room measurement process. Systems that do the calculations with an external PC have an advantage.

The other metric of room correction quality is how much filtering can be accomplished in real time by the DSP. Each filter section requires multiplication and addition (subtraction) operations. The faster the DSP operates, more Multiply-Accumulate (MAC) operations can be applied to each sample. MACs (or FLOPS (floating-point operations per second), not the more familiar Million Instruction per second (MIPS), are one of the benchmark for DSP performance. The latter is the typical benchmark for microprocessors

Different DSPs have different MAC ratings that correlate with price. MACs are not the only consideration for a DSP engineer since each DSP has specialized instruction sets and data paths that also affect performance. For example, the DSPs designed for use in AVRs would drain a cell phone battery in a few minutes. Development tools differ significantly for different DSPs and different applications. Without good tools, it may be impossible to complete the design job on time.

In the HK 990, the DSP processor does two other jobs. First, is the bass management function. This will monopolize some of the available MACs. Recall the HK 990 supports two subwoofers that double the MACs for the room correction computations at the subwoofers (calculation requirements for the main channels are unchanged). Second, is to simulate the analog bass and treble control.

The fourth daughter board is a general purpose platform on which the TI Aureus TMS320DA708 DSP chip (Chip package logo shown in Figure 12 above; ™ Texas Instruments) and the associated RAM and Flash memory reside. It is unclear how Harman harnesses the board's computer power in the HK 990. They could have just used the code from the top of the line AVR (AVR 7550HD), leaving most of the computational power of the chip, which would normally process the other unused six channels in AVRs.

Alternatively, the resolution of the room correction system could have been improved over the AVR 7550HD, or the designers could have introduced high-order crossovers to enhance the bass management system. Unfortunately, my measurements of the bass management system indicate it is no better than the ones found on AVRs.

The DSP in an AVR has other chores aside from eight channels of room correction, bass management, and optional jitter suppression (soft ASRC):

- The decoding of all possible multichannel audio formats entering on the HDMI line to LPCM (example: DTS Master Audio)
- Volume compression (example: Dolby Volume)
- Surround synthesis function (example: SRS Circle Surround)
- Conversion of 5 or 7 input channel inputs to 9 or l1 (example: DTS Neo X).
- Signal processing to improve the sound of low bit rate compress signal such as MP3
- THX equalization
- Virtual Presence Speaker function
- Hall simulations using delay and reverb (often now expanded for game modes)

To accommodate the myriad functions, two or three DSPs may be present in an AVR. Often two DSP are placed on a silicon die and enclosed in a single package. Much of the code outlined above for an AVR may be proprietary to the DSP vendor. The company designing the AVR picks what options they want and thus have little added DSP coding. The more options, the more MACs used. Low-end AVRs with 5.1 outputs and simple room correction may use one low cost DSP, while a more elaborate unit may employ a trio of DSPs. The DSP vendor profits not only by supplying more DSP hardware, but also via fixed and variable (per unit) fees for the company's code blocks.

AVR manufacturers may develop code for a custom room-correction system, though some DSP suppliers also supply room-correction code. Many Sherwood units and the lower-priced Harman AVRs have room-correction algorithms sourced from Cirrus. Given the substantial research and development effort, an AVR manufacturer hopes to recover these costs in a custom implementation by producing better quality sound when the room correction is selected. In turn, the improved standard of performance should enhance sales.

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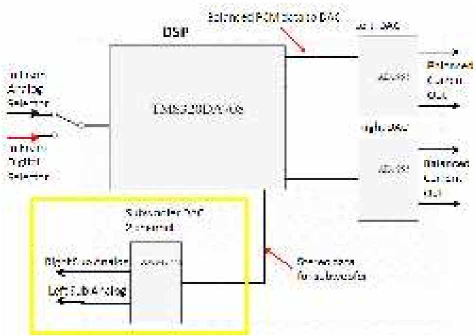
HK 990 Digital to Analog Conversion (DAC)



The block diagram of the DSP board is repeated as Figure 13 above with the Subwoofer DAC highlighted in the yellow box. All digital-to-analog converters are driven by the DSP. It is typical to place the DACs on the same board as the DSP since a significant amount of wiring runs between the two (a single wire is used in the diagram above to represent 3 wires).

The HK 990 can deal with two independent subwoofers. The DSP connection to the stereo subwoofer channel is typical of a DAC connection in an AVR. In the HK 990, the stereo DAC for the subwoofer is the Wolfson XMB8740. The Wolfson XWM8740 is below the DSP in the previous figure.

In an AVR, the number of DACs connected to the DSP depends on the number of main and subwoofer channels offered. The plethora of DACs complicates the block diagram, but nothing fundamentally changes. Typically, in a 7.1 channel AVR, eight channels must be converted with four stereo DACS.

The Wolfson XMB8740 DAC is above average. The bit-equivalent signal-to-noise ratio (SNR) is 18 bits worst case (A-Weighted). The bit-equivalent distortion is 15.5 bits worst case at 1kHz at full-scale. This is more than adequate for the subwoofer channels exceeding the performance of DACs for the main channels in most AVRs.

Some may find it strange that each data converter is sourced from a different vendor (Cirrus, AKM, Wolfson and Analog Devices). The DSP is from Texas Instruments, who also manufactures audio DACs. In general, large electronics enterprises (not just in the audio space) do this to ensure no single IC company commands significant pricing leverage and, at the extreme, ensures no vendor is indispensible. This makes life difficult for smaller companies who have no pricing power given their small order quantities.

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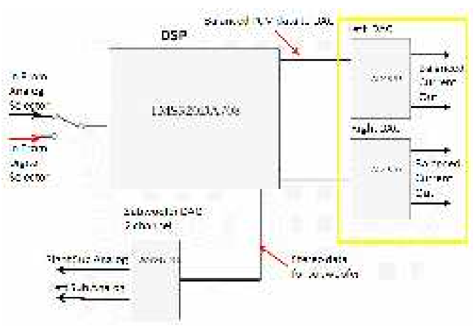
HK 990 Improving Performance by Operating a Pair of DACS in a Balanced Configuration



The HK 990 (and HD990) handles the main channel DSP/DAC interface in a novel manner. Figure 14 above provides the block diagram details in the yellow box. One channel is converted by two DACs in the single Analog Devices AD1955 package (on the left side of the block diagram). Normally the AD1955 is a stereo DAC, but in the HK 990 it will convert only one channel. This is accomplished in the fully balanced configuration. In a balanced circuit, one input is the inversion of the other. Instead of sending stereo data to one of the AS1955 the DSP sends the same LPCM data both in phase and out of phase. Again a digital wire in the diagram above to represent 3 wires.

After each DAC in the AD1955 completes its analog to digital conversion, there are two balanced analog outputs. In the analog stages that follow the DACs (not shown in the block diagram), the balanced signals are subtracted from each other to form a single-ended output. In the process of converting the balanced analog signal to the single-ended output, some distortion produced by each DAC in the AD1955 package is partially canceled. Using two DACs for one channel also reduces the noise level at the DAC's output.

The bit-equivalent signal-to-noise ratio (SNR) of the AD1955 in the mono mode is 18.6 bits equivalent bits worst case (A-Weighted). The bit-equivalent distortion exceeds the chip's stereo specification of 16.7 bits worst-case at 1kHz at full-scale. The data Harman Labs sent me exceeded this at the preamplifier output with all the additional analog electronics adding noise to the measurement.

The balanced DAC configuration of the HK 990 is atypical for an AVR. If used at all, the balanced configuration would apply only to stereo channels.

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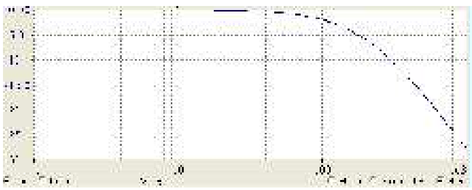
HK 990 Backend Analog Circuitry

The diagram above (Figure 15) shows the remaining circuits that are in place before the preamp output. We have already discussed the interface of the DSP to the DAC. Next, the reconstruction filter must remove the high-frequency folded spectra from the sampling process. Most of the filtering of the folded spectra was performed in the digital domain as was discussed above.

Below (Figure 16) is the simulation of the how second-order analog reconstruction filter in the HK 990. Only residual high frequency clock noise is removed by this filter. The filter is down -0.5db at 45 kHz (approximate maximum inband frequency for 96 kHz sampling). Since it is a low Q 2nd order filter it is down only 20dB at 600 kHz.



The next block in the diagram is the digitally-controlled volume control. Recall the volume knob on the front panel is a component of the MMI and is not a genuine potentiometer. As it is turned, the microprocessor senses movement and sends a formatted digital signal to the digitally-controlled volume control requesting a change in the tap position.

As shown in the diagram, the digital volume control does not connect to the preamp outputs. A buffer isolates the volume control output from the load at the preamp output and power amp inputs. A relay is in the signal path to prevent loud popping sounds that occur as the unit powers up or down from being transmitting to the output. The buffer and relay are also as close to the preamp outputs as possible.

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Conclusions

We ended at the analog preamp out. Our journey through the HK 990 DSP-centric integrated amplifier has concluded. While the unit's design requires a multi-disciplinary approach and hundreds of parts, the basics of the design at the block-diagram level is reasonably straightforward. Along the tour, I highlighted the similarities of the HK 990 to an AVR. With the HK 990, there were only two channels with which to deal. We also did not get lost in the weeds with issues related to video boards.

Part III drills down at the circuit level where patterns are very different from an AVR. It is at the circuit level where one gains an appreciation of the unit's cost. If you want to revisit Part I of this HK 990 three-part series....

Tags: **Amplifiers** | **Integrated Amplifiers** | **Stereo**

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Home . Integrated Amplifiers . Harman Kardon HK 990 Stereo Integrated Amplifier with Digital Room Correction and Dual Subwoofer Bass Management – Part III

Harman Kardon HK 990 Stereo Integrated Amplifier with Digital Room Correction and Dual Subwoofer Bass Management – Part III

Written by Dr. David A. Rich

Thursday, 03 November 2011 00:00

Article Index

Introduction to Harman Kardon HK 990 Stereo Integrated Amplifier with Digital Room Correction and Dual Subwoofer Bass Management – Part III

Circuit design and the tape recorder section are the focus of the final section of my HK 990 review. The topics are addressed to different groups. Those interested in the tape recorder section will find information on Page 9. The tape recorder interface section outlines matters of connectivity and usage. Usability issues are discussed.

The tape recorder interface section is at the level of the prior two parts and assumes no knowledge of circuit-level electronics.

The circuit-design section enters the land of high-end design, mostly guided by the design principles of Matti Otala. My intent here is two-fold: to identify anything that might make the unit sound better subjectively, and to objectively quantify whether any aspects of the design have inadvertently degraded performance.

Specifications

- Design: Solid State Stereo Integrated Amplifier
- Power: 2 x 150 watts RMS into 8 ohms @ 20 Hz - 20 kHz, 2 x 300 Watts into 4 Ohms
- MFR: 10 Hz – 100 kHz
- THD: <0.07% at Full Output (8 Ohm Load)
- Analog Inputs: 7, Plus 1 Phono MC, 1 Phono MM, and 1 Balanced XLR
- Digital Inputs: 1 HRS-Link, 2 Optical Digital, 2 Coaxial Digital
- Analog Input Sensitivity/Impedance: 350mV/43k ohms for tuner/CD, 10mV/47k ohms for Phono-MM, 1mV/100k ohms for Phono-MC
- Digital Input Capability: All Standard Digital Formats
- Dimensions: 6.4" H x 17.3" W x 17.5" D
- Weight: 43.2 Pounds
- MSRP: $2,599 USA
- Harman Kardon
- SECRETS Tags: Harman Kardon, HK 990

This circuit design section assumes the reader has knowledge of analog design equivalent to a 1980's Audio magazine. I started reviewing in the 1980s. I took my cues at what level to write from Audio and never let go. Some of the technical terms bubbled to the surface in the HK 990 literature as the marketing folks tried to capture the unique aspects of the unit. You get only words without graphs or block diagrams. Usually, only when engineers write the literature does it all jell. Accuphase is an example of this practice operating to perfection. The literature Harman produced in the 80s and early 90s included circuits, novel measurements and even Bode plots. Sansui and Kenwood were producing similar material. Sansui was a big loss from the creative circuit view when it went out of business.

Here are links to Part I and Part II of this review.

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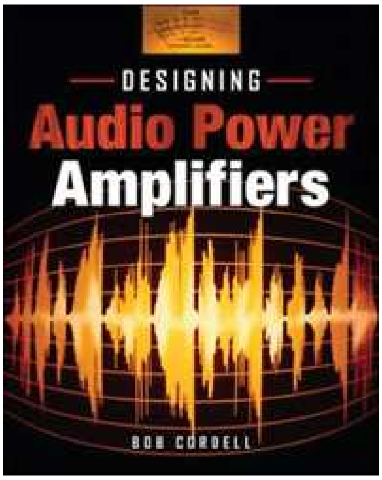
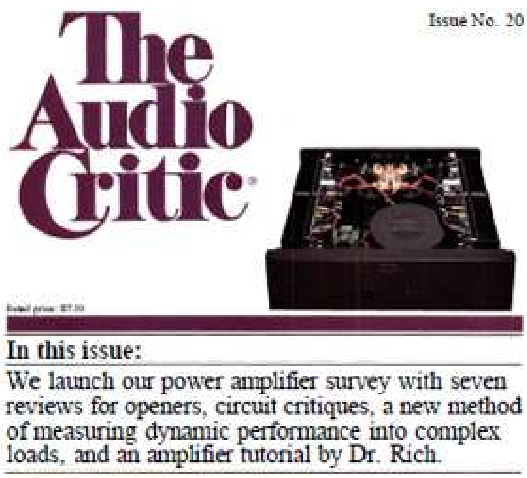
Construction of the Analog Blocks

A textbook by Bob Cordell is an excellent reference. The book is oriented to those interested in audio-electronics design.



During the early 90's, I summarized basic fundamentals in several issues of The Audio Critic and analyzed fully-designed commercial components as case studies. Issues 18 and 20, focus on preamplifiers and power amplifiers, respectively.



Free PDFs of these articles are available at >www.theaudiocritic.com.

For those with a limited knowledge of electronics, the classic The Art of Electronics (Horowitz and Hill) is a must-have. Though pricey and heavy (at 1100 pages), the breadth of material well serves the needs of both novices and experts. If you are lucky, it may be in your local library; otherwise, try to secure a copy through interlibrary loan.

Significantly cheaper is the 5th edition (2011) of Teach Yourself Electricity and Electronics written by Stan Gibilisco. I have not reviewed the text, but it has been recommended by others whose opinions I value.

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HK 990 Volume Control

The volume control is a good starting point because it is the one spot in the HK 990 where Otala design concepts are violated. Two stages of operational amplifiers are present when an analog source, including phono, is selected.

The Analog Device AD825 opamp is used. This is not a discrete circuit, which is surprising. A discrete circuit allows the amplifier stage to be optimized to all Otala requirements. A signal coming from digital media often sees many opamps in the recording studio, but this is not true for analog signals entering the HK 990 or signals from the phono preamp. It is especially surprising given that a discrete circuit is at the output of the HD 990 CD player.

The AD825 has some unusual characteristics that are more consistent with Otala design rules. That said, the opamp designers were probably unaware of these rules and coincidentally came upon them during optimization of the opamp for applications outside audio. The AD825 in the unity-gain configuration has a return-loop gain of 2000 (66dB) from 20Hz to 10 KHz. It then declines at 6dB per octave. A typical opamp has an open-loop gain of 100,000 to 1 million and will start rolling off around 10 – 100Hz. The AD825 has a high slew rate (115V/us), wide bandwidth (34MHz as a unity gain buffer), and sizeable open-loop linear input range. Indeed, Walt Jung references an Otala Audio Engineering Society (AES) paper in his Electronic Design article of December 1, 1994 on the AD825. Still, it is not fully complementary like other Harman discrete circuits and methods that that prevent oscillations when the feedback loop is closed are different.

The downsides of the AD825 include higher noise levels than the typical audio opamp and some discrete designs.

The digitally-controlled volume IC in the HK 990 is a JRC NJW1159 comprised of only silicon switches and a resister ladder. There is no operational amplifier on the chip. With a specified maximum supply voltage of maximum +/- 7V (limited by breakdown voltages of the switches in the IC fabrication process used), the JRC NJW1159 requires a pair of dedicated sub-voltage regulators.

The 50k ohm input impedance of the JRC NJW1159 is typical of an analog control. However, MOS transistors are sensitive to electrostatic discharge and stress when the input voltage exceeds the +/-7V power supplies. The AD825 unity gain buffer in the signal path protects the JRC part. A DC blocking capacitor is at the input of the buffer. Another is at the input of the volume control. A second AD825 with a gain of 2.4 (8dB) at the output of the volume control prevents it from being loaded by the preamp output jacks. I calculated the input impedance of the power amp at a low 10k ohm, meaning the second AD825 is required for the direct power amp connection. The power amp has an above-average gain of 32dB to compensate for the low gain in the volume control section (the line stage if it was an independent preamp).

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HK 990 Power Amplifier

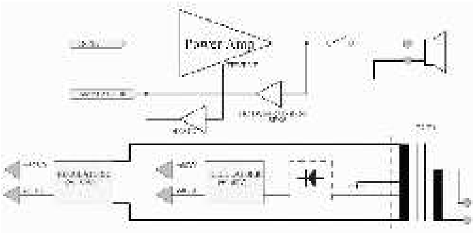
The power amplifier is full complementary from input to output. There are eighteen transistors just to implement the voltage gain section excluding the class AB bias stage. This complexity ensures the open-loop distortion is low so only a small, constant amount of feedback, as dictated by Otala, is required around the complete amplifier from 20Hz to 20 kHz.

The differential pair at the front of the amplifier is biased by current sources and a buffer stage isolates the first gain stage from the second. The differential pair and second gain stages are both cascaded. Both voltage gain stages have local feedback (emitter degeneration). The four circuit techniques linearize the open-loop distortion of the amplifier and keep the return-loop gain to Otala's desired minimum level. Just for reference, some AVRs over $1000 do a complete power amp with eight transistors including the AB bias and current gain stages.

Otala and other researchers showed much more current is required to drive a speaker than a resistor. In the HK 990, the current gain block is a triple Darlington terminating in five paralleled 15 amp continuous ON Semiconductor power transistors (MJL3281A and MJL1302A complementary pair) connected from one supply rail and the speaker terminals. These devices have 260V breakdown voltages.

Counting output transistors at the output of an amplifier is a futile exercise. You need to refer to the data sheet to find the short term and steady-state current each transistor can source or sink safely. The frequency at which the current gain goes to 1 (the point at which we could replace the transistor with a wire) should be near 30MHz. Current gain in the audio band should be greater than 20 when sourcing the maximum steady state current. Open-loop distortion under full load depends on the process technology. Typical specs sheets for an output transistor run more than four pages. Paralleling output devices has the disadvantage that the load the pre-driver sees is more difficult to deal with.

The HK 990 is dual mono down to its two transformers, a feature found on many older Citation products. All the metal in the transformers and output-stage heat sink literally weigh down this forty-five pound unit.



The HK 990 power amp uses separate transformer winding for the voltage and current gain stages. +/- 80V for the voltage stages and +/-60V for the output stages, as shown in the diagram above. This approach prevents the power supplies for the voltage stage from being modulated as the current stages send significant current to the speaker. A higher supply voltage for the voltage gain stage is rarely attempted because under a fault condition (shorted speaker terminals, for example) the number of pathways through which the transistors can be damaged multiplies. In a traditional amplifier with a high return-loop gain, the amplifier is relatively insensitive to modulation of the power supply (power supply rejection ratio). With the lower return-loop gain required by Otala, the modulation results in distortion.

Since spec sheets make a big deal of the size of the capacitor on the unregulated rails, I can report each cap on the +/-60V supply is 13600uF, slightly less than reported on the HK spec sheet. The size of these capacitors is important at low frequencies (20Hz) where the power supply is required to source or sink significant current in one direction as the sign wave remains at its maximum or minimum voltage level for a long period relative to the power supply refresh rate (120Hz). This is why THD goes up at low frequencies in some amplifiers when driving low impedance loads at the maximum output voltage swing.

The +/- 80V rails have 1000uF capacitors on the unregulated rails. This capacitor can be much smaller since the current drain on the 80V supplies from the voltage gain stages is smaller than what is flowing on the +/-60V rails driving the speaker.

The HK 990 spec sheet gives no FTC power rating into 4 ohms. The back panel of the HK 990 has a 1000 Watt rating for total power consumption from the AC line. Obviously, some of the power heats the amplifier owing to the efficiency of a class AB amp, but the construction points to the unit's capability of providing almost 300 average Watts per channel (RMS power does not exist except in the mind of some marketing people. The engineers should get the blame for not proof reading the material. HK avoids this error.) into 4 ohms under FTC test conditions across the full frequency band. Why Harman does not supply such a key spec is a mystery. The mystery deepens because fans are mounted to the large heat sinks to aid in passing the FTC preconditioning test.

Harman Kardon was the first company to adopt the Audio Graph Power Cube measurement system. The unit tests for stability into inductance and capacitive loads (+/-30 degrees, +/-60 degrees) when the amplifier is driven to full power into loads as low as 1 ohm (magnitude of the load impedance). The test signal is 20 periods at 1kHz, with the power reported at a THD of 1% I do not have results for the HK990.

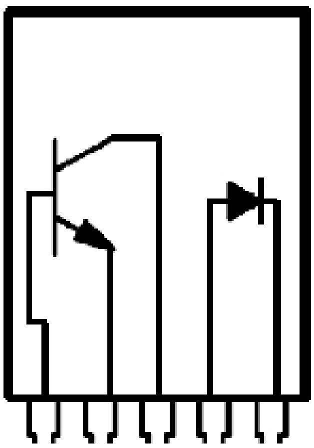
www.audiograph.se

The Audio Critic had access to a Power Cube measurement system. Harman products always did well while many other higher-priced units failed. That said, some amplifiers with traditional circuit topologies also did well. Class D amplifiers tend to create poor Power Cubes.

The idle current of the output stage is very high. The unit gets hot with no input signal, so it is advisable not to rest another component on the HK 990. The high idle current is an attempt to reduce crossover distortion with the low return-loop gain specified by Otala. Doug Self has shown crossover distortion can be made extremely small at low idle currents.

Not a single bypass or blocking capacitor is found in the power amp. A DC Servo circuit is designed to prevent a DC offset voltage from appearing at the speaker terminals. A DC servo provides a compensating voltage at the power amplifier’s input to remove the DC offset at the output. The servo circuit is designed to only respond to subsonic frequencies and DC. The use of a DC servo to eliminate capacitors is not a universally accepted method to improve sound quality. Some argue the additional active circuitry for the servo is more audible than a well-chosen capacitor.

Be careful with the power amplifier inputs. If the input has significant DC, the amplifier will go into protection (note the DC over-current sensor in the figure. The DC over-current detector also activates if the speaker terminals are shorted or an internal component fails in the amplifier.). An amplifier with a DC blocking cap at the input would remove the DC.



New to the HK 990 is the ThermalTrak class AB bias stage. Normally the bias circuit has a diode placed on the heat sinks so temperature of the output devices can be sensed. ON Semiconductors place a diode on the same die as the output device (NJL3281D and NJL1302D) to improve thermal tracking (please refer to the figure above). The thermal time constant associated with the heat sink is eliminated. ON semiconductor has shown ThermalTrack reduces distortion.

Harman has a odd specification in their literature: 200 amp instantaneous current. At first glance, it appears to be a typo. Just considering the primary capacitor size and power supply rail, the capacitor would be discharged in 4msec by my calculations. This spec turns out to be an old measurement from the Audio Graph Power Cube test system mentioned above. The amplifier is connected to an 0.1 ohm resistive load and driven with a 10kHz (100usec) square wave. Audio Graph looks to have deleted the made in the current system. I am hard pressed to see the value of the specification now, but some readers with long memories are going to post me that The Audio Critic did report it when I was Technical Editor.

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HK 990 Phono Stage

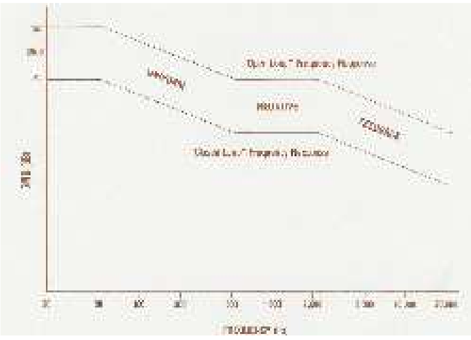
In his first preamp for Harman (Citation XXP), Otala addressed the challenge of achieving a flat return-loop gain across the frequency band. In a typical amplifier with a flat closed-loop response, this is readily achieved in a discrete circuit at the cost of distortion (attempts are made in the discrete opamp design to reduce open-loop distortion when the return-loop gain is purposely set to a low value as we saw in the power amp).

The problem is more difficult in a phono preamp with a varying frequency response (it follows the RIAA curve). The HK 990 uses a topology dating to Otala's tenure. Twelve transistors form a transconductance amplifier. The transconductance amplifier differs from an opamp since it has a high output impedance. Think of the transconductance amplifier as an opamp without the common emitter output stage. It is called a transconductance amplifier because a change in voltage at the input (open loop) results in a change of current at the output.

Iout=VinG where G is conductance which is the reciprocal of resistance.

The RIAA network is connected around the transconductance stage. As the frequency increases, the passive RIAA network loads the transconductance amplifier and lowers its open loop voltage gain (the current output of the transconductance amplifier flows in the RIAA network giving rise to a voltage). Since the transfer function of the passive RIAA network is increasing with frequency, the total return loop remains constant at the desired level chosen by the designer.

The plot below was taken from the original literature for the Citation XXP preamp.



Low frequencies pose a problem with this approach. Here, a closed-loop voltage gain of 60dB is required to match the inverse RIAA curve for a moving magnet cartridge. The open-loop gain of the transconductance circuit loaded by the RIAA network should be 80dB at a minimum. This may not be achievable because the intrinsic output resistance of a real transconductance amplifier limits the gain. The result can be a drop in the gain at low frequencies and increase an distortion.

I do not have measurements of the HK 990 to confirm if it exhibits this problem. Some earlier HK phono stages hinted at the problem, but the transconductance amplifier on the HK 990 is significantly enhanced. It remains fully complementary. Current sources replace a resistor to bias the differential input stage. A buffer between the first and second gain stages prevents the second stage from loading the first. The first voltage gain stage design trades off degraded noise performance for improved open-loop distortion.

The DC servo used in the phono stage eliminates all bypass and coupling capacitors. Surprisingly, no DC blocking capacitor is at the input to the phono stage. Since a bipolar transistor is used at the input, a small current always flows in the cartridge. I have rarely seen this direct connection implemented unless the input transistor is a FET with an extremely low gate-current flow.

An open-loop emitter-follower buffers the transconductance amplifier from the load presented by the line stage. Almost all phono stages are in the non-inverting feedback configuration. Such a configuration is limited to a gain no lower than one. The RIAA curve should continue rolling off and not stop at unity gain. A passive low-pass filter in this stage corrects the problem (The Audio Critic, Issue 18, page 16). This and other phono stage design issues are addressed in Chapter 7 of Small Signal Audio Design by Douglas Self (Focal Press, 2010).

A 330pf capacitor is at the input. This high value exceeds the specifications of most moving magnet cartridges when the turntable wiring is included. At this price, I would like to see a switch on the back to offer different loading options. In the absence of that, a low-valued capacitor is advisable since the value can be raised externally. In contrast, nothing can compensate for too much capacitance at the phono input.

Moving coil cartridges see a pre-preamplifier that is a two-stage open-loop design (no feedback). Since the signals from the moving coil cartridge are so small, the transistors almost act as linear devices; hence, no feedback is required to linearize the moving coil stage. Other designers might have employed feedback to reduce the distortion to lower levels. The pre-preamp has no DC servo. A standard DC blocking capacitor is at its output. In the moving magnet stage no blocking capacitor is at the stages input. Again its absence causes a small DC current flow in the cartridge.

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HK 990 Headphone Stage

The high performance Texas Instrument TP6120 obviates DC coupling or blocking capacitors in this stage. The TP6120 has very low distortion and noise. It has a current feedback topology and 1300V/us slew rate. I have no idea why it is designed with a slew rate 100 times larger than what is required. Without circuit details, I cannot conclude if Otala would approve, but with a THD under full load approaching 0.0002% at 1 kHz, I doubt this design has low return-loop gain.

One issue I identified involves the calibration microphone connected using the headphone output. If nothing is done, the microphone could potentially be damaged when plugged in while a large level signal is at the headphone jack. The headphone amps are disconnected when the HK 990 is put in calibration mode, but it is unclear if they are disconnected automatically when the microphone is inserted and the unit is not set to calibrate. I suspect all HK units with room calibration have the same issue but they use a small IC to drive high efficiency headphones. The TP6120 can deliver much more current and only a 20 ohm resistor is between it output and the headphone jack. The instruction book does not caution about these circumstances.

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HK 990 Analog Circuitry Connected to the DACs

The analog circuitry in this stage is a typical design and does not reflect any influence of Otala so I will not spend much time discussing it.

The Analog Devices AD1955 has a current source output. No on chip opamps are present.

Analog Devices OP275 and TI OPA2134 are used. The characteristics of these opamps are more conventional than those of the AD825. I suspect the higher power supply rejection, lower noise and lower input capacitance of these parts may have played a role in their selection. The absence of discrete circuitry is going to disappoint purists wedded to Otala design.

A DC servo circuit senses the DC offset voltage at the output of the DAC signal chain and introduces an offset current at the DAC to correct this (recall the DAC has a current output). The topology eliminates all coupling and DC blocking capacitors from the signal chain This is the first time I have seen a DC servo integrated in the digital-to-analog conversion block.

The output of the analog electronics connected to the DAC goes directly to the volume control bypassing the buffer in the direct path. In total three active opamp stages are present before the signal enters the power amp. One DC blocking capacitor is in the signal path. From the perspective of an engineer who does not slavishly adhere to Otala design rules, the execution of the direct analog and DAC paths is extremely well done.

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Conclusions About the HK 990 Circuit Design

I wrote much of this material in October 2010. At the time, my sole reference was the service manual (the actual unit arrived several months later). I intended the piece to be a preview of the HK 990, which had just been shown at CEDIA. Since then, the HK 990 received a positive review from Tyler Stripko and at least one other professional reviewer. It is rare for a relatively low-priced unit to achieve such consistently positive reviews. I will not comment about my impressions of the sound of this amplifier, other than to note that Tyler's positive sentiment may be justified by the work of Otala. Every design decision is backed up with some quantitative analysis. The HK990 is a polar opposite of designs offered by others where the design techniques are more closely allied to black magic than solid engineering and expensive metalwork wraps the questionable design to give the look of ultra high tech. A five-figure price is tacked on to enhance the aura of ultimate performance. Poor measured results are often a consequence when black magic replaces science. The component may sound different, but different is not necessarily better.

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HK 990 Dual-Domain Tape Recorder Outputs and Tape Monitor Details

Having been involved with tape recorders since reel-to-reel was the state of the art. I invested considerable effort in testing the dual-domain tape recorder interface function. Conceptually, a dual-domain tape output path should be a must-have for those seriously rooted in recording. With a dual-domain configuration, both analog and digital inputs appear at the analog tape output jack (to recorder in) for the two tape recorders (Harman calls these CD-R and Tape). Analog and digital inputs also appear on the single record digital output (to digital-in on the CD-R or MiniDisc player). The rear panel section with the tape outputs is shown below.

Harman allows the tape outputs to record a source (analog or digital) independently of what is heard on the speaker. This feature is relatively common in analog integrated amplifiers and preamplifiers replacing the single tape monitor button.

Listening to one input and taping another provides a tape monitor function. For example, one might tape the Tuner with the cassette deck connected to Tape using this sequence of operations:

1. Select Tuner using the record-out selector.
2. Select Tuner on the remote control to listen to the source through the speakers.
3. Press Tape on the remote to hear the recording off the cassette deck.

Even with a CD-R, monitoring is important. I cannot tell you how many times I thought I hit the record button, but inadvertently hit Play instead. With the meters moving, everything looks OK, but it's not. Another example where monitoring could save the day is when the Tuner is selected as record-out and not Phono that you wanted to record; again, the meters are moving but you are recording the wrong thing.

Source selection for the tape recorders is only available on the front panel, making the selection more than a nuisance when all the buttons on the front panel are similarly-sized with low-contrast lettering.

To select a recorder, the Record Output button is first selected, then the two Source Select buttons are toggled, and finally Record Output is re-pressed to return the source selector to its normal function so the speakers are operative. From the remote, toggling the source select for the output to the speakers is unnecessary. Each input is assigned a button.

Closely-spaced buttons may cause an adjacent button to be depressed. Put your finger a little to the right of source select and you press the input assignment setup button. Cancelling that is no fun.

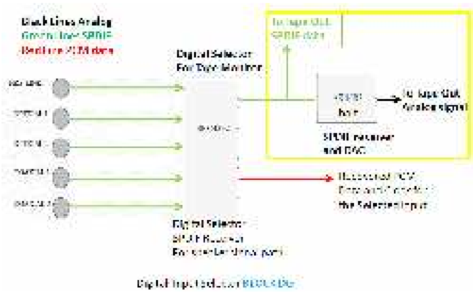
Many AVRs offer one or two tape outputs (with composite video outputs also supplied for at least one). These outputs are analog and transmit only analog inputs. No monitor function is available.

You can try to mimic the Harman's functionality using the Room 2 analog outputs on the AVR, but care is required as there is no protection for self loops under this setup: say goodbye to your tweeters were both Tape for Room 2 and source selected at the same time with the recorder activated. The oscillations are typically a square wave at the maximum swing of the tape recorder.

You may be able to work around this using the AVRs Room 2 input assignment GUI. One might try to assign Room 2 Tape In to an unused analog input. I make no guarantees you can perform a setup with an AVR Room 2 that will insure a disaster will not occur. I play it safe and live without the monitor function with an AVR. I use headphones connected to the recorder to do the monitoring function.

Almost all AVRs provide only analog inputs (single domain) at the Room 2 outputs.

Tape I/O signal flows at the block diagram level

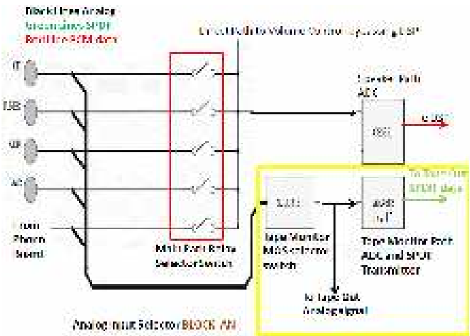


This diagram is similar to the block diagram of the digital input selector presented in Part 2, but the circuitry to support tape recorders has been included. The yellow box highlights the added circuits. The Texas Instrument SRC4392 multi-function chip has two digital input selectors. As can be seen one selector is used for the tape output path and one for the main path. In the tape output path the selector only routes SPDIF signals (green) and does not recover the PCM data as it does for the main path. A second set of input selectors are required for the tape recorder output path because the digital input selected is different from what is being sent to the speakers.

An extra DAC is required in a dual-domain tape system to convert the digital inputs to analog. This is highlighted in the yellow box. The DAC for the analog record outputs is part of a multifunction AKM 4683. The performance of the AKM DAC is lackluster, with a worst-case dynamic range equivalent to 15.5bits and distortion of 13 equivalent bits. The AKM 4683 also houses the requisite SPDIF receiver.

Older cassette and reel-to-reel machines need the analog output. The quality of the conversion of the digital inputs to analog is not that important since all CD-Rs and MiniDisc recorders have digital inputs; however, a problem with this assumption in the case of the HK 990 will be identified below.

A designate the diagram above BLOCK DG.

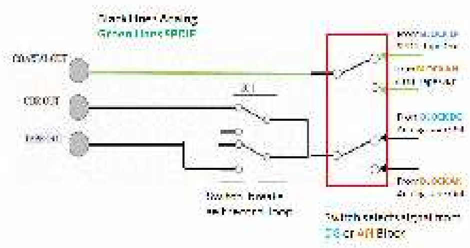


The analog selector block is shown in this diagram, with the circuitry to support tape recorder in yellow. Like the digital path (BLOCK DG), the analog input selector has second set of switch selectors that aggregates the analog inputs for the tape recorder outputs. Everything in this tape output channel (in the yellow box) is only AVR grade: the switches, for example, are MOSFETS, not relays.

The selected analog input must be converted to digital to provide a digital tape output. The ADC should be of high quality because it replaces the ADC in the digital tape recorder. Unfortunately, the HK 990 disappoints. The ADC is also in the AK4683 (with an ADC and DAC, it is called a CODEC). The ADC has a worst-case dynamic range equivalent to 15.5 bits and distortion of 13 equivalent bits.

You may ask why the output of the high-quality Cirrus CS5361 ADC in the main path (in the upper right of the diagram) is not used instead of adding an ADC just for the tape recorder outputs. The principle justification is the bifurcation of the input to be recorded from what is played on the speakers.

A designate the diagram above BLOCK AN.



This block diagram illustrates the conclusion of the recorder output's journey as it makes its way to the rear panel RCA jacks. This diagram clarifies the dual-domain aspect of the tape recorder path. Both digital and analog outputs are shown at the right. Depending on the component that the user has selected to record, a signal from the analog block (AB) or digital block (DG) is sent to the output (signals entering at right).

Note the switch prior to the two analog outputs CDR-Out and Tape-Out. The switch prevents self-oscillation. When listening and recording from CDR, the switch mutes CDR-Out, preventing a self-oscillation through the CDR recorder. Tape-Out is muted under similar conditions.

The digital Coaxial Out lacks the switch. The single digital output is live when you select to record from the CD-R or the Tape input. In contrast, the analog output for CD-R (Tape) mutes when to-record CD-R (Tape) is selected. Mixing analog and digital connections in a recorder can cause a self-loop and high-level oscillation that may damage your speakers.

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Proper Connection of the HK 990 Tape I/Os to a Digital Tape Recorder to Avoid Self-Oscillation

When connecting the HK990 to a tape recorder, I advise one of two methods to prevent self- oscillation:

1) Use only analog cables for the input and output connections.

Given the quality of the ADC in the HK 990 (AK4683) little is lost since the ADC in your CDR is likely to be as good as the one in the HK 990, especially if the ADR a semi-professional unit. Coming out of the CD-R analog to the HK 990 does degrade the signal since it travels through the DAC in the CD-R and the ADC (AK4683) in the HK 990. This is a redundant DAC – ADC path.

2) Use only the digital cables for the input and output connections.

Self-oscillation is avoided by coming into the HK 990 from a recorder's digital output (in the all-digital hookup) because both digital inputs mute when record selector is set to CD-R or Tape. In this manner, the oscillation is broken on the input connection for the digital inputs and the output connection (discussed above) for the analog connection. This is what makes mixing analog and digital connection so dangerous.

By simultaneously muting both CD-R and Tape digital inputs, tape-to-tape copying from CD-R digital in) to Tape is enabled. This is probably related to the Digital Millennium Copyright Act, whereby Tape-to-Tape copying is allowed with only analog interconnections.

At first glance, an all-digital connection of the CD-R appears preferable. The all-digital path eliminates the DAC – ADC redundancy on the recorder output side. However, conversion quality is crimped because the AK4683 ADC replaces the ADC in your CD-R.

With only one digital output, I do not understand why it is not marked digital output for CD-R and made to mute when CD-R is selected as the analog CD-R does. This should be a straight-forward software fix to allow mixed analog and digital connections.

The digital output is limited to a 16 bit depth and 48kHz sampling rate. Consumer digital recorders would mute if high resolution SPDIF data was allowed to appear at the output.

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Conclusions About HK990 Tape Recorder Functionality

Harman is to be congratulated for providing the dual-domain tape recorder interface. Regrettably, the front panel controls make it difficult to use. The quality of the internal ADC and DAC in the tape recorder path is disappointing. Mixing analog and digital connections to a CD-R or MiniDisc recorder can result in potentially damaging self-oscillation.

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Overall Conclusions About the HK 990

For those who started with Tyler's review and worked their way through this three-parter, 20,000 words have passed your eyes. Only a very special product requires that level of analysis. There is no comparable for the HK990. One could say that an equivalent could be crafted with three or more boxes. Unfortunately, this is not a viable option because many different functional blocks inside the HK990 interface with each other in ways that cannot be replicated with RCA cables running between multiple external boxes.

As with any debut product, the HK990 has some glitches. The front panel controls are very difficult to manipulate and the room-correction system has some software bugs. Putting these issues aside, the HK990 is a revolutionary product that will be on the list of the 100 most important audio components ten years from now.

Tags: Amplifiers | Integrated Amplifiers | Stereo

Comments (4)

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The Nationally Recognized Testing Laboratory safety mark on the HK990
written by David Rich , November 03, 2011

In my discussion of the protection system for the power amp I neglected to mention the ETL safety mark at the rear of the HK990. ETL is a Nationally Recognized Testing Laboratory (NRTL), certified by OSHA (Occupational Safety & Health Administration) to assess equipment safety. Underwriters Laboratories (UL) is also an NRTL and uses the same tests as all other NRTL's. The NRTL mark is found on most products sold at Best Buy, but high-end equipment is more likely to display a CE mark, referring to standards, including safety, required for sale in the EU. CE testing is undertaken by the manufacturer, not a third-party. CE certification is less comprehensive and less expensive to achieve since the tests are done internally.

The absence of replaceable fuses or resettable circuit breakers illustrates an enhanced safety element of the HK990. When the Amp Fault line goes high (see block diagram of power amp above), the power to the transformers driving the power amplifier is cut off by a relay in series with the transformer. This is done under the command of a protection circuit that monitors for faults. The protection circuit also commands another set of relays to open the connection of the amplifier to the speakers. This removes a fault that could be caused by a short in the speaker cable and protects the speaker from a large voltage at the amplifier's output when the fault is inside the amp. A third transformer supplies power to the protection circuit and remains connected to the AC line.

A user-induced fault might be a shorted speaker terminal or an overdriven amplifier input. The amplifier will recover after it is switched off provided the user removes the cause of the fault before turning the unit on again. If the amplifier is internally damaged, the fault signal status returns and the relays at the speaker and AC line again open. Now the amp most go for service. For the HK 990, it is never incumbent upon the user to change a fuse or reset a breaker as would happen if the amplifier did not have internal relays. That's to the good because a user might substitute a higher amperage fuse, which would be quite dangerous were the unit to remain on when an internal fault produces a potential safety hazard.

Circuit-breaker values cannot be change by the user, but the fault must be significant enough for the current flowing in the power line to exceed the breaker's pre-set level after damage has occurred. The HK 990's fault status line activates well before the line current reaches the maximum level. Some amps have many sub-fuses that activate when one amplifier draws too much current. The user must remove the cover and identify the tripped fuse. NRTL-tested equipment is affixed with a label that warns against opening the cabinet (for good reason).

When fuses and breakers are present, no relay may be in series with the speaker's output terminals. Your speaker could be damaged before the fuse or circuit breaker activates. The HK990 does have a couple of hidden fuses that trip if the protection circuit is defective and fails to act. Any component in the category of the HK990, including a home theater in a box, is vetted similarly by an NRTL and has a similar protection system. The website below identifies the marks that should appear on your amplifier after passing a safety inspection by a third-party NRTL.

www.osha.gov/dts/otpca/nrtl/nrtlmrk.html

Congratulations with an Exceptional Review!
written by Igor Khachaturyan , November 04, 2011

I would like to thank and thank you for the most comprehensive and informative review that i have read in many years!
Once could only wish that all audio reviewers had your competence and willingness to provide facts instead of based opinions.

Bravo!

Re: Congratulations with an Exceptional Review!
written by David Rich , November 05, 2011

Igor,

Your support of my work is appreciated and I am glad you found it useful.

Nothing like his would be possible without the support of the editor-in-Chief, managing editor, and the site manager. Besides the process of editing the document, the staff prepared the piece for on-line publication, then split it to improve ease of navigation. Also, many figures were modified to enhance readability.

It would have been impossible to present this in printed form, and I am heartened by the team's commitment to ready my work for the site.

Your feedback validates everyone's effort.

David Rich

HDMI
written by Nick , November 06, 2011

Bummer that there is no HDMI. I have a two channel system, but several sources that only output over HDMI.

Write comment

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comment

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email

Your email is kept private and will not be shown on the website.

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