A HW/SW Co-design Environment for Multi-media Equipments Development using Inverse Problem

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ABSTRACT Multi-media equipment development must provide functions adjusted to human sensibilities. Realization of such functions depends on how the three transfer levels-- perception, recognition, susceptibility--are handled. In this paper, we deal with perception by employing inverse problem to characterize the system and correctly reproduce signals. To accommodate recognition and susceptibility, we propose an optimization method in which results are compared repeatedly with a model of human recognition characteristics. With respect to system response, numerical models, filter design, playback, evaluation, estimates of cost and performance when implemented as semiconductor circuits, and generation of a netlist for semiconductor production, we propose an environment for hardware/software co-design and development based on four steps which make possible comprehensive, systematic development and design, from the conceptual stage through to production. A procedure for solving the inverse problem is incorporated in the four areas of this environment, which are interconnected and efficiently linked to the development process, so that the overall development cycle can be shortened. This proposal was applied to the development of a television receiver and audio circuitry, and its effectiveness confirmed.

1. Introduction

Audio and video stimuli are conveyed to humans via a three-stage process: perception (acoustic) -> recognition (informational) -> susceptibility (preferences). For this reason, in multimedia equipment development, where the requirements of these three levels must be satisfied, it is difficult to produce and evaluate corresponding functions, and many tasks have to be repeated.

In response to the above-described problems, this paper proposes a HW/SW co-design environment for multi-media equipments development using inverse problem: four steps addressing each of the three levels of perception, recognition, and susceptibility. These four steps are concept-making, virtual-world prototyping, real-world prototyping, and synthesis, comprising a complete hardware/software co-design development environment. Techniques for inverse problem-solving are incorporated into these four steps, which are interconnected and efficiently linked to the development process. This proposal was applied to the development of a television receiver and audio circuitry, and its effectiveness confirmed.

2. Proposal

This paper proposes a HW/SW co-design environment for multi-media equipments development using inverse problem:

* Inverse problem method for stimuli on three levels
* Four steps environment in design and development

(1) Inverse problems for stimuli on three levels

Inverse problem-solving involves characterization of input and the system based on the system response. An inverse problem must satisfy the following three conditions[1].

(1) Existence: There must exist a solution z in the solution-space F for an arbitrary output.

(2) Uniqueness: The solution z in solution-space F for an output u must be unique.

(3)Continuity: The input z to be determined must vary continuously with the output u that is used to solve the problem.

If these three conditions are satisfied, inverse problem analysis can be used to determine the input or the system. On the other hand, stimuli pass through the three levels of perception recognition, and susceptibility. If we consider as an example the effect of audio input on perception, by placing microphones at the positions of the ears and applying methods for inverse problem analysis, we can determine the input characteristics. In other words, the output signal characteristics for the equipment can be found. In multimedia equipment and systems, if the input can be determined, then by controlling the input signal it is possible to reproduce a signal which is ideal at the perception level--that is, we can reproduce signal characteristics similar to those of the original sound or image. On the other hand, where recognition and susceptibility are concerned, the subject is a human being and things are not so simple. Solution methods include optimal control of audio, video and other stimuli according to models of human sensory perception and sensibilities, and equipment design including functions for the manual adjustment or selection of signal characteristics according to preference. In the optimal control, if the process signal output=y, and the model signal output=m, then the new model is adapted by y-m = minimum value, automatically or semi-automatically. In this paper, we propose a method for solution of broadly defined inverse problems relating to the three stimuli levels, including inverse problem analysis techniques and optimal control methods as shown in Fig. 1.

(2) Four steps in design and development

This paper proposes four steps for comprehensive, systematic development and design, extending from concept all the way to production. For each steps there is an environment for hardware/software co-design and development. The four proposed development environments are as follows.

1)**Concept-making** development environment: Creation of product concepts to meet the needs of a diversifying market.
2)**Virtual-world prototyping** development environment: An

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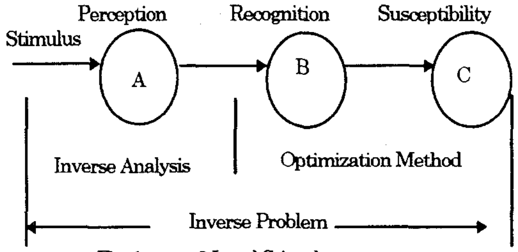
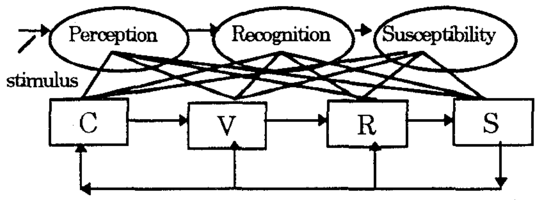


Fig. 1 3 Level Stimulus



C:concept-making V:virtual-world prototyping
R:real-world prototyping S:sythesis

Fig. 2 The relationship of 3 stimulus levels & 4 steps

environment enabling direct visual confirmation, in virtual space, that a product as defined satisfies users.

3) **Real-world prototyping** development environment: Performs real-world prototyping, in which audio and video are actually reproduced to confirm operation.

4) **Synthesizing** development environment: Finally, if the above results are satisfactory, the circuitry is logically synthesized as a semiconductor device circuit, the device operating frequency, chip size and other parameters are predicted, price is estimated, and logical synthesis is performed. Techniques for inverse problem-solving are incorporated into each of the four steps of this environment, and they are interconnected and efficiently linked to the development process. A schematic diagram illustrating co-design through the four steps and the three stimulus levels appears in Fig. 2.

3. Experimental Verification Through an Inverse-Problem Sound Field Development Environment

The proposed methodology was verified by constructing an actual inverse-problem sound field development environment for the development of multimedia equipment with specific audio functions.

(1) Concept-Making

Measurement and analysis environments are used to determine the amplitude and phase characteristics of the sound field corresponding to an unknown circuit. The measurement environment includes single-shot pulse generators, speakers, microphones and a personal computer. It serves to gather numerical data assessing the target circuit's response characteristics. The analysis environment consists of a workstation-based signal-analysis system with software for performing Fourier transforms, reverse Fourier transforms, and window-functions. The environment analyzes the collected sound data to determine numerical and physical response characteristics and to define an identified model.

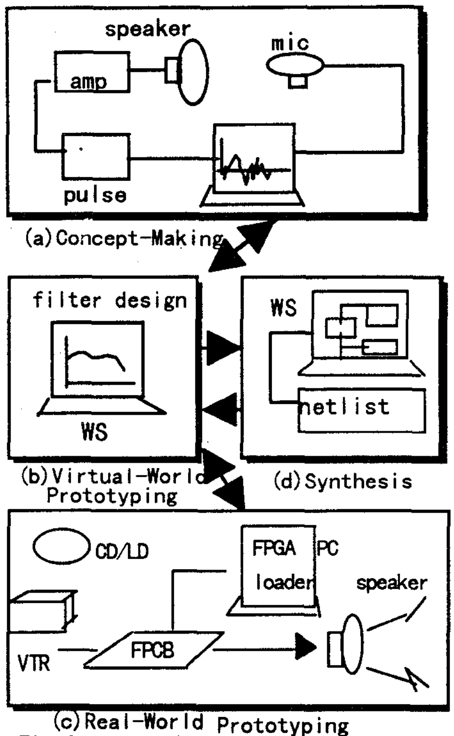


Fig. 3 Schematic diagrams for flow and configuration of the system

(2) Virtual-World Prototyping

This process creates a new model that combines the modeling of ideas for the audio characteristics of the end product developed in the concept-identified stage with the working model derived from the unknown circuit. Signal processing is performed on the initial models and a filter design environment is used to determine a new optimized solution for the system [2]. The workstation-based filter design environment includes software for designing finite impulse response (FIR) and infinite impulse response (IIR) circuits, digital signal processors (DSPs), and DAC and ADC circuits.

(3) Real-World Prototyping

A real-time processing environment reproduces the sound output corresponding to the new optimized solution for subjective evaluation in live listening tests. The environment includes a field-programmable circuit board (FPCB) that can automatically download the netlist created in the logic simulator, a reproduction unit and speakers.

(4) Synthesis

Once the developer has arrived at a satisfactory circuit solution, an ASIC development environment performs the processing required to implement this circuitry in silicon [3]. This workstation-based environment includes an EDAS tool that performs logic synthesis and creates a netlist[4]. The synthesis stage is necessary only if semiconductor devices are to be used.

4. Design Method on the Development Environment

4.1 Concept-Making

(1) Idea Generation

Faithful sound reproduction requires that the aggregate amplitude and phase frequency characteristics of the speakers and signal-processing system satisfy the following model

$$X(z)F(z) = 1 \qquad (1)$$

where $X(z)$ represents the frequency characteristics of the unknown circuit, $F(z)$ represents the inverse characteristics of $X(z)$ of the idea model.

(2) Modeling of Ideas

The sound is handled as discrete frequency data and FIR filters are applied toward satisfying Eq.(1). DSPs are designed to perform sum of products processing on the discrete frequency data to achieve the desired filtering effects.

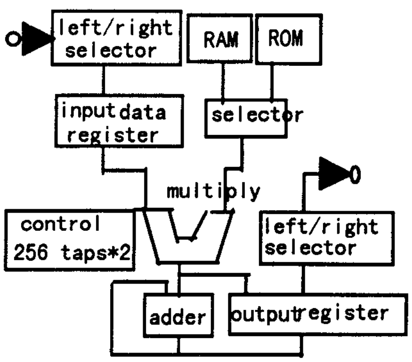


Fig. 4 Schematic diagram of sound LSI

(3) Analysis of Unknown Circuit Characteristics

Fig. 5 shows an amplifier and speaker circuit as unknown circuits intended for use in a multimedia audiovisual product. A single-shot pulse generator is applied to the circuit and a microphone connected to a personal computer is used to record the response. The smoothing of the impulse and window-function processing are performed on this data, followed by a discrete Fourier transform to determine the frequency characteristics $X(z)$ of the unknown circuit. If $xm(n)$ is the response to the n th pulse, $x(n)$ represents the arithmetic average of $xm(n)$ from $n = 0$ to L (the data length). If window-function processing is applied to perform wave-form correction at both ends of the pulse for filter length N, the impulse response $xw(n)$ follows from

$$xw(n) = hw(n) \bullet x(n) \qquad (2)$$

where $hw(n)$ Is Hamming window ;

$$hw(n) = 0.54 - 0.46\cos(2\pi\, n/N) \qquad (3)$$

Performing a Fourier transform on Eq. (2) yields:

$$X(z) = DFT(xw(n))$$
$$= 1/N \sum_{n=0}^{N-1} xw(n) \exp(-2\pi n \bullet k / N) \quad (4)$$

where $k = 0, 1, \ldots N-1$. The function $X(z)$ derived here represents the frequency characteristics of the unknown circuit.(Real part : Sound pressure amplitude characteristics, Image part : phase shifted angle characteristics). The wave-forms of Eq.(2) and (4) can be observed graphically on a personal computer or workstation.

4.2 Virtual-World Prototyping

(1) Calculating Inverse Filter Coefficients

From Eq. (1), $F(z)$, the inverse characteristic of $X(z)$ is given as follows:

$$F(z) = 1/X(z) \qquad (5)$$

The inverse filter coefficients $f(n)$ are determined by applying an inverse Fourier transform to Eq. (5), which yields

$$f(n) = IDFT(F(z))$$
$$= 1/N \sum_{n=0}^{N-1} F(k) \exp(-2\pi n \bullet k / N) \quad (6)$$

where $n = 0, 1, \ldots N-1$.

(2) Overall Characteristics Computation

From the inverse filter coefficients $f(n)$ and the pulse response $x(n)$, we perform convolutional arithmetic to derive the overall characteristics of a new model $y(n)$ for the unknown circuit.

$$y(n) = \sum_{k=0}^{L} f(k) \bullet x(n-K) \qquad (7)$$

where if $k>=N$,then $f(k)=0$.

(3) Response Customization

The frequency response characteristics can be tailored to particular purposes by changing Eq. (5) to the form

$$F(z) = G(z) / X(z) \qquad (8)$$

, and substituting the results into Eq. (6) and (7) where $G(z)$ represents the customized frequency characteristics. The desired characteristics can be entered graphically on a workstation screen and converted to data for substitution into Eq. (8).

(4) FIR Filter Design

Fig. 4 shows the basic FIR filter circuit in LSI. The development environment allows the sampling frequency, tap length, data word length, filter coefficient word length to be set to any desired value. But some limits are set by practical sound levels, the processing capability of the FPCB used for real world prototyping, and the processing capabilities of the ASICs developed in the final synthesis step. Eq.(7) shows that the sum of products $An \cdot X + B$ is calculated repeatedly for each tap length, where constants An are the inverse filter coefficient $f(n)$ and variable Xs are sampled values, and B is the sum of products at $N-1$. The FIR to implement this function consists of a delay unit, multiplier, adder and memory for storing the constants.

4.3 Real-World Prototyping

The specialized FPCB used to realize real world audio circuits. 6 consists of a field-programmable gate array (FPGA) and switches that provide for connections between standardized components and logic functions. The output signals of FPCB is connected to the target circuit.. For general sound reproduction, a low-distortion audio signal source supplies the input and the target speaker provides the output.

Once logic data for the circuitry is approved in the virtual prototyping stage, an EDAS tool downloads the logic data, implements the logic in the FPGA, and generates the appropriate switch connection data.

4.4 Synthesis

An EDAS tool takes the FIR filter logic simulated in the virtual world prototyping step and generates netlists, gate counts, interconnect counts, critical virtual wiring data, delay timing data, maximum and minimum operation timing data, and test vectors[3]. If the system represented by this data does not provide satisfactory performance, the developer returns to the virtual world prototyping process. The data can also be processed to estimate a die size of semiconductor, cost and performance.

5. Application of a TV receiver

Fig. 5 shows an application of the development environment to the audio circuit of a television receiver. The sound pressure level and phase characteristics have been improved over the audible frequency range. The circuit receives inputs from an external jack, terrestrial broadcast receiver and satellite broadcast receiver. The signals are routed by an analog switch, pass through compensation circuits and two-channel amplifier, and are output by two horn-type speakers. The circuitry was evaluated by supplying a pulse to the external jack and measuring the response of a point equidistant from the two speakers. The compensation circuitry was inserted into the original circuit to realize the improved circuit shown in Fig. 5.

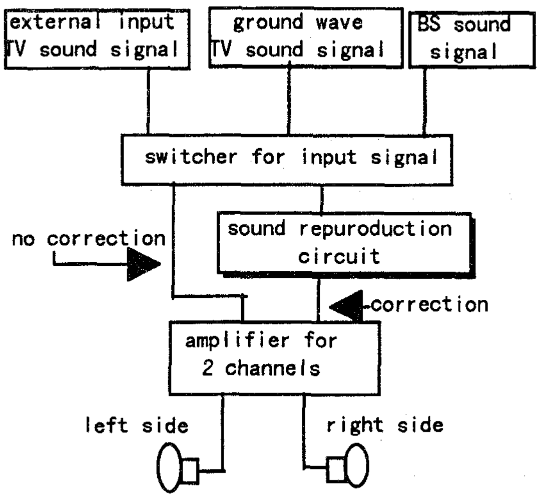


Fig. 5 The application of a television receiver audio circuit

6. Experimental Results

(1) Pulse Response

The single-shot pulse generator was used for testing maintains accurate wave-forms at frequencies up to 100 KHz. We selected a 10μs pulse width, 1V pulse amplitude, and 100 ms pulse separation. The response wave-forms were averaged and Hamming window-function applied to obtain corrected pulse wave-form response according to Eq.(2). Trade-offs in the number of filter steps and filter coefficient processing performance led to selection of a 32 KHz sampling rate and 256 step filters, with corresponding a window-function processing.

(2) Frequency Response, Inverse Filter Coefficients, Final Output Wave-forms

Fig. 6-(a) and Fig. 6-(c) show the frequency response and the phase response based on a discrete Fourier transform of the

Table 1 Development Time

	Department of Process	day
1	Concept-Making	7
2	Virtual-World-Prototyping	7
3	Real-World=Protopyping	12
4	Synthesise	4
	Total Time	30

Table 2 Circuit Scale and Cost Estimation of LSI

FIR Tap Length	DSP Ciuit ①	RAM ②	ROM ③	Cost ①	Cost ②	Cost ③	Total Cost ①②③
	gate	gate	gate	ratio	ratio	ratio	ratio
16	5048	5216	768	1	1	1	1
192	5276	18432	9216	1.05	3.53	12.00	3.00
256	5271	24576	12284	1.05	4.71	16.00	3.82
*	12.5	58.3	29.3	12.5	58.3	29.30	

* Ratio(%) of DSP:RAM:ROM in Scale and Cost of 256 Tap Length

impulse response respectively. Fig. 6-(b) and Fig. 6-(d) show the final output wave-forms which were determined by taking the frequency response and the phase response respectively, performing a reverse discrete Fourier transform to derive the inverse filter coefficients, and then convoluting the impulse response with the coefficients.

(3) Subjective Considerations

Fig. 6-(e) and Fig. 6-(f) show sound characteristics adjusted for optimum subjective listening pleasure of three generations of listeners. Fig. 6-(e) shows a target characteristics. And Fig. 6-(f) shows an adapted characteristics. These characteristics were designed graphically on a workstation, and substituted into Eq. (8) to generate the appropriate circuitry by the semi-automatic operation.

(4) Development Period

Table 1 shows a comparison of data for the period taken to complete development under previous methods and that of the proposal for operations involving otherwise similar functions. In previous methods, every project involves its own preparatory stage, ordering of equipment, adjustment of machinery, configuration of the environment and the evaluation of tools, and reviews of the development methods. For this reason, operations involving similar operations took three months(90 days) while the method of our proposal takes one month(30 days). Real-world prototyping took 12 days, but much time was consumed in setting up the interface for virtual-world prototyping.

(5) Scale of LSI Gate, RAM and ROM <Table 2>

Table 2 gives predictions for the scale of basic circuit elements and the cost of a device for which estimates were prepared. The functions of the EDAS synthesis tool output the scale and number of the elements required, facilitating the prediction of costs. The circuits are largely divided into the DSP section, the RAM section and the ROM section for prediction. The trade-offs between cost and performance can be made on the basis of structural scale for each filter lengths. This DSP is not a generally available LSI, and the ability to create it is an important advantage. The lengths of the filters dedicated to audio signal processing can be freely selected by changing the RAM and ROM capacity.

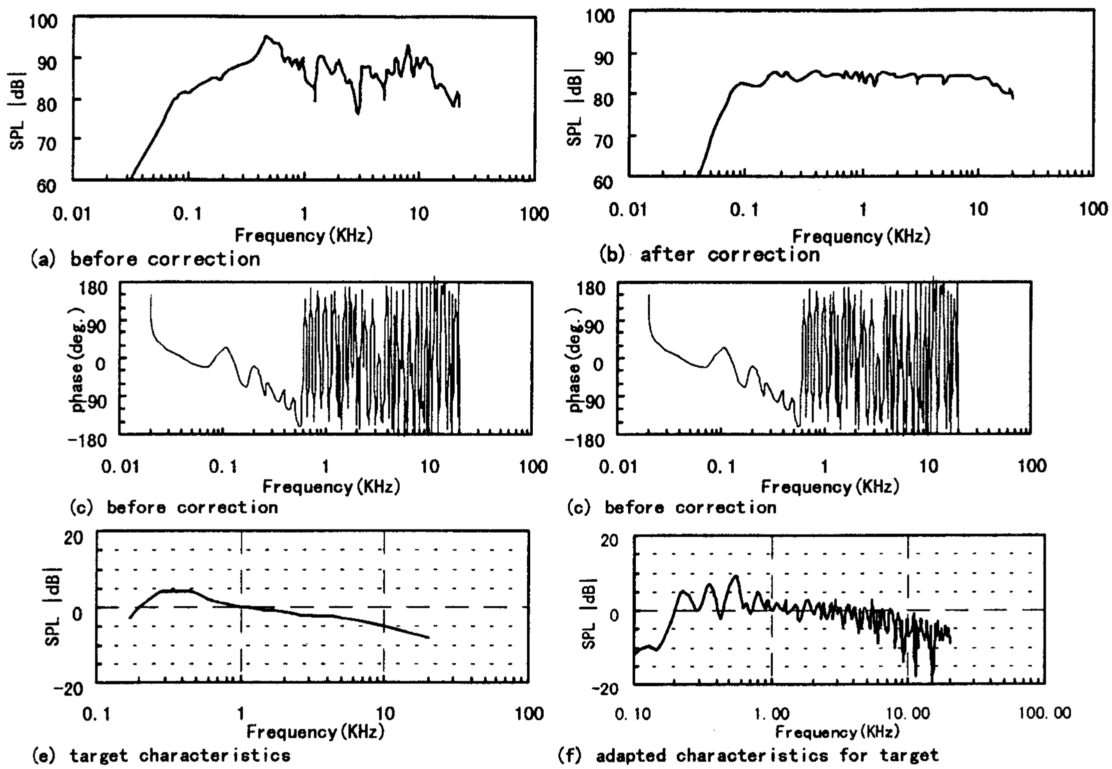


Fig. 6 The Results of Frequency Characteristics

7. Discussion

(1) Inverse Problem

In the inverse problem approach we have adopted for the sound reproduction control, we have used the principle that $X(z)F(z)=1$ using inverse analysis. The frequency response of the application circuit was limited to 16 Hz due to the 32 KHz sampling frequency used. Fig. 6-(b) and Fig. 6-(d) show the change in sound reproduction over this bandwidth with and without the insertion of FIR filtering that clearly shows how this filtering contributes to realization of a clear, deep and natural sound. And we have used the principle that $F(z)=G(x)/X(z)$ using optimization method. Fig. 6-(f) shows that the desired characteristics can be generated for the customer.

(2) The Four Step Approach

A four-stage approach to the digital sound design development environment using inverse analysis for the acoustic response was proposed. According to the data in Table 1, circuit development that would have required three months under a conventional environment was completed in one month using the environment proposed here. The development time was reduced to one third of the previous method due to the efficient linking of the component tools in the proposed environment with its division into four distinct standardized processes and functions.

(3) Device Scale and Cost Estimation

Co-design of device scale and cost can be estimated for the DSP, RAM and ROM used for varying numbers of filter taps. Table 2 shows that cost of a dedicated DSP is about 15% of the total cost, and that it does not vary significantly with the number of filter steps. This suggests that circuitry can be designed for upgrading by installation of additional RAM and ROM without sacrificing overall economy.

8. Conclusion

This paper has presented a hw/sw co-design environment for multimedia equipments development using inverse problem for the acoustic applications that divides the development process into four interrelated steps and thereby realizes substantial reductions in development time and satisfactions for the customer. The value of this environment has been verified by using it for actual circuit development work. The problem of sound design from concept to production has been treated as a single inverse analysis problem. This unified approach has significant implications for the design of other rapid prototyping systems. However, many other multi-media areas for technological improvements such as optimization methods remain. The authors plan to continue research on the application of inverse problem and rapid prototyping methods to the co-design development environment for multi-media equipments that satisfy subjective human sensibilities.

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