

Audio Processing In Car Infotainment Systems

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Abstract— Present day in-car infotainment systems use audio DSP to incorporate many advanced features. The paper briefs about the car radio architecture which is used in high end infotainment systems and explains how the audio mixing takes place in the superposition block of the audio DSP. The paper also explains the audio processing operations like filtering and scaling, which a signal undergoes before it reaches the output of the speakers. It also contains how audio mixing takes place in the superposition block of audio DSP.

Keywords— In-Car Infotainment, audio DSP, superposition block.

INTRODUCTION

Today's car manufacturers are integrating more and more complex multi-media equipment into their vehicles in order to meet the changes in consumers' buying behavior. This collection of firmware equipment installed into automobiles is called In-vehicle Infotainment. The in-vehicle infotainment includes audio and/or audio/visual entertainment, along with automotive navigation systems (SatNav). Present day car radios have features like:

- Tuner sources - FM, AM, XM
- Media Sources like CD/DVD, USB Audio, Picture, Video, SD Card support, iPod, AUX;
- Wireless connectivity: Bluetooth, Phonebook, Internet, Phone app access
- Telematics: GPS, Modem
- Speech Recognition
- Rear-seat entertainment

It also has features like internet radio (Pandora and Stitcher), traffic control radar.

But the DSP (Digital Signal Processor) is the core of the infotainment system. Digital signal processors (DSPs) are being used widely for sound field reproduction. There are various papers published which shows the design of digital audio signal processing circuitry having high fidelity and sound reception ([1] and [2]). Also papers are published which describe different architectures of DSP which is used in car radio ([3] and [4]).

2. ARCHITECTURE

The design to any electronic device is done based on the architecture followed. The overall car infotainment architecture is as shown in Figure 1. This is a layered architecture and can be broadly classified as HMI, Application Interface (API), Middleware and base drivers. The hardware and the OS are at the lowest layer. Software code resides in the middle ware and the application layer. The code is made platform independent so that the programming can be done across platforms [5].

2.1. Car Infotainment Architecture

Figure 2 shows the basic architecture used in present day car radios. The host microcontroller forms the core which is responsible for the flow of signal into and out of the DSP. Present day car radios follow a dual architecture involving: two processors. The processors deals with various features in the car radio like networking, navigation, entertainment, diagnostics, etc. Moreover they also monitor and regulate the power requirements by various components.

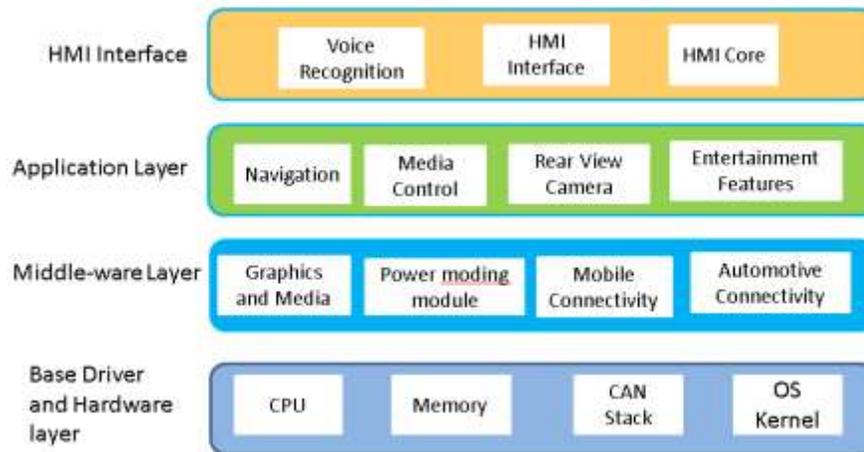


Fig. 1 Car Infotainment Software architecture

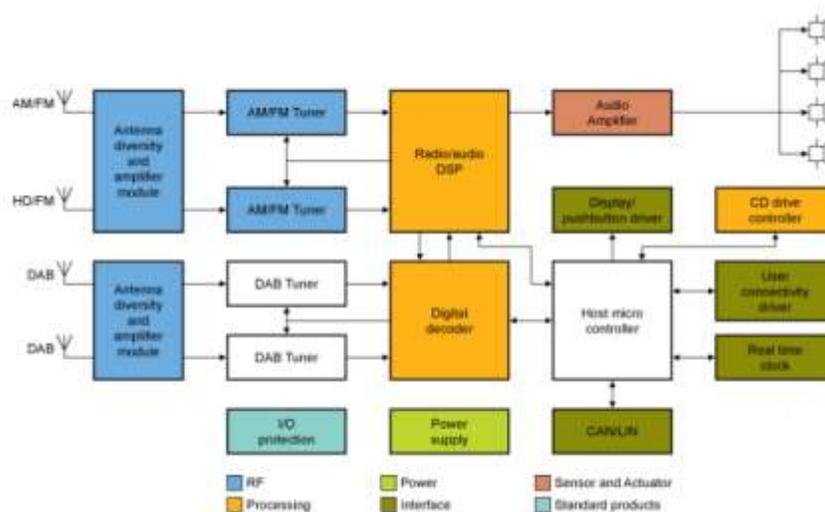


Fig. 2 Car radio Hardware architecture

Additionally there is a CAN (Control Area Network) which is responsible for the communication between the different ECUs (Engine Control Units) in the vehicle. The main audio processing is carried out in the DSP. The DSP can be divided into radio DSP and audio DSP. The output from the DSP is passed to the speakers through the amplifier.

The DSP has the following functional blocks:

2.1.1 Sample Rate Converter (SRC):

Sample rate conversion (SRC) is the process of changing the sampling rate of a data stream from a specific sampling rate (e.g. the input/output hardware rate) to another sampling rate (e.g. the rate at which application samples are processed). With the conversion of communication and software markets, SRC is a prime component in many of today's applications [6]. In most of those applications, a

very high quality sample rate converter is required. Most high quality SRCs employ a digital filter that provides the required quality by up-sampling the data to a very high sampling rate followed by down-sampling to the required output sampling rate.

2.2.2 Radio processing block:

The radio processing block digitizes signal at the tuner interface to digital converter. Additionally, it also performs blind equalization of the FM channel; this results in an outstanding rejection of the adjacent channels and any other interfering signal, even under severe multipath conditions. The DSP of the car radio includes the tuner part, which is responsible for the AM/FM reception and signal processing. The signals received undergo demodulation in this block. Frequency translation and filtering of the received signals takes place in this block. DSP based receivers incorporate digital demodulation to replace analog demodulation techniques.

2.2.3 Audio processing block:

The audio processing block takes care of the signal distortion, signal conditioning and equalizations. The audio processing block consists of various filters, noise generators, sine generators, limiters, equalizers, etc. These are explained in the following sections.

3. AUDIO PROCESSING

Following the audio input, the audio data is normally processed by a series of processing blocks programmed. These processing blocks are normally common audio blocks such as IIR (Infinite Impulse Response) filters, volume control, tone control, DRC (Digital Range Compression), loudness, delay, mixers, and many others. As the DSP is fully programmable, the processing that can be done specific to the customer requirements. The DSP programming is done according to the specifications and needs of the customer.

To accomplish the duty of post audio processing, the audio is transferred through the DSP and this audio is transformed in the desired fashion. Audio comes into the DSP via an Input channels. An Input along with a Converter is used to create a Source that is used to transform the audio into a format the DSP can utilize. The Audio travels through the DSP via a Physical Audio Path. This Physical Audio Path is constructed of Components that are used to transform the audio in desired ways. The audio then exits the DSP via an Output.

3.1 Signal flow in Car radio

When a user selects a source in the car radio, the corresponding audio is played through the speakers. The audio for the corresponding source undergoes various stages of processing before reaching the speaker.

Figure 3 shows the flow diagram of a signal in the car radio. The user selects the source from the HMI. The HMI sends the request to the source arbitration block. The main function of this block is to request to the DSP to play the corresponding source. The audio DSP provides the audio path for the corresponding source performs the necessary processing and outputs the signal to the DAC (Digital to Analog Converter). The converted analog signals are then provided to the amplifier. The input to the amplifier will be a low dB signal and finally to the speakers.

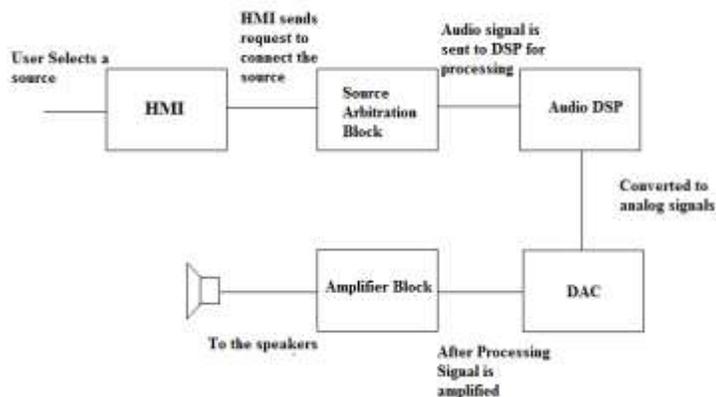


Fig 3. Flow diagram of signal a car radio

3.2 Digital Audio Scaling

In the DSP, the digital audio processing is designed so that noise produced by filter operations is maintained below the smallest signal amplitude of interest. Figure 4 describes the various terminologies used in the audio scaling. This low noise level is achieved by increasing the precision of the signal representation substantially above the number of bits that are absolutely necessary to represent the input signal. These scaling blocks act similar to gain blocks wherein the signals are attenuated at certain levels so that the net output is distortion free.

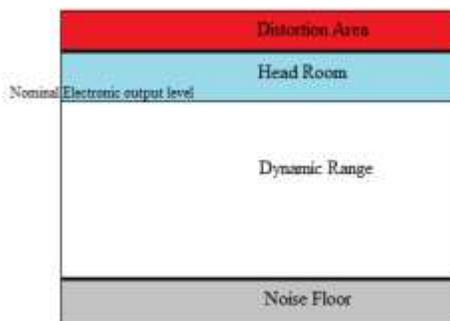


Fig 4: Audio Signal level: Relationship between SNR, Head room and Noise floor.

3.3 Digital Audio Processing in an audio-path

Figure 5 shows how the audio signal is processed in a single audio channel in the audio DSP. The audio signals can be boosted at various levels in a channel path to compensate for the headroom.

Scalar coefficients are used to adjust the volume of the audio passed through the control. Different Scalar coefficients will have different intended purposes. Some are intended for scaling of the audio of an audio path based on which audio application is currently selected. Others are intended for system level volume control. All of these Controls have a minimum and maximum gain property associated with it.

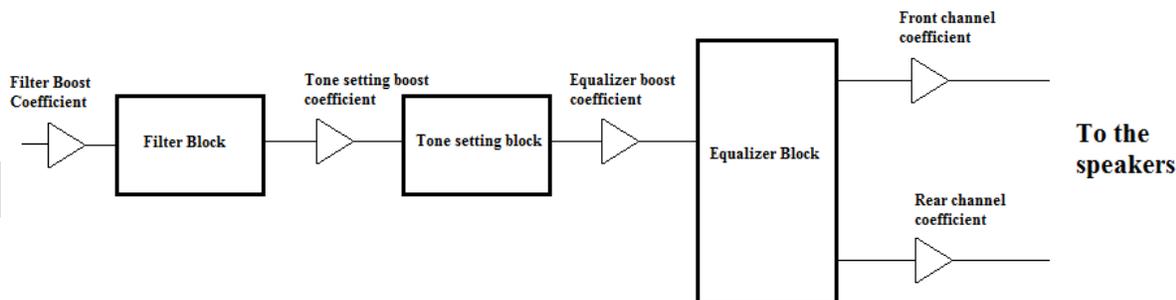


Fig 5. Audio scaling and boost coefficient in an audio path.

3.4 Digital Audio Mixing

When two different applications have to be played on a single channel, the sources have to be mixed. The mixing of sources is achieved by the superposition block. In this block the source signals are added together. The output volume of the sources that are mixed is set according to the customer requirement. For example if a USB/ SD source is playing and navigation prompt comes, the same channel plays both the above sources with varying volume output. This is achieved by the superposition block. Figure 6 shows a block diagram of a superposition block. The superposition block is responsible for the mixing of the audio signals. Appropriate mixing according to the requirement can be achieved by proper software programming.

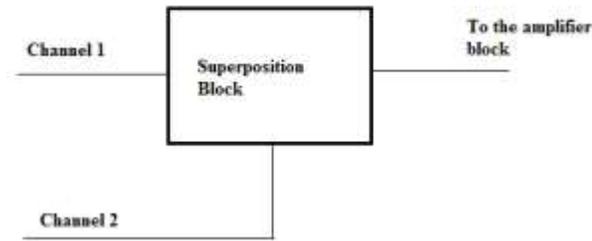


Fig6: Superposition Block in DSP

3.4 Equalizers

In order to alter or control the frequency response characteristics of an audio system, the equalizers are present. An audio equalizer provides more accurate control of tone or frequency than simple bass and treble controls. Equalizers and signal processors are two kinds of devices that allow to fine tune the sound in the vehicle's audio system.

The equalizers are usually present between the amplifier and the head unit. There are various equalizers. The two main equalizers are as follows:

- Graphic Equalizer
- Parametric Equalizer

3.4.1 Graphic Equalizer

These equalizers have fixed bandwidths. The sliders are present for each of the cut off frequencies so. The user can manually change the gain of each of the frequencies to obtain the desired level of attenuation. A graphic equalizer is the simplest type and consists of multiple sliders or controls for boosting or cutting bands or frequencies of sound.

Dynamic adjustment of frequencies of each of the bands can be achieved using the Graphic equalizer. For example, a typical five-band graphic equalizer has sliders for five fixed frequencies: 30Hz (low bass), 100 Hz (mid-bass), 1 kHz (midrange), 10 kHz (upper midrange) and 20 kHz (treble or high-frequency). Each of the bands is made of IIR filter. The equalizer can boost or cut each frequency but each frequency is fixed. With increase in the number of bands better tone control can be achieved.

3.4.2 Parametric Equalizer

A more complex type is the parametric equalizer, which controls more parameters of the sound than a graphic equalizer. A parametric equalizer can control three parameters of each frequency: level (boost or cut), the center or primary frequency and the bandwidth or range of each frequency. However, a parametric equalizer can also control the center frequency.

Some car Audio DSPs have both Graphical and parametric equalizers which help in wider control of audio.

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CONCLUSION

With the present technology, the various features included in the high-end infotainment are discussed. The paper also explains the general architecture followed to manufacture today's car radio systems. The Digital Signal Processor (DSP) forms the core for any car infotainment architecture. All the signal processing is performed in the DSP. A detailed description on the audio processing which occurs in the DSP is provided. The audio signal which is available at the input of the DSP undergoes stages of audio processing. These stages are described in detail. Proposed operations are manual. We can extend it to wireless or Bluetooth.

REFERENCES:

- [1] Matsuda, A., Shinada, A., and Morise, K., "Break Through Car Audio with DSP," SAE Technical Paper 910792, 1991, doi:10.4271/910792

- [2] Asami, T., Okamoto, M., Mochiyama, Y., and Nakamura, H., "Development of a Practical DSP Car Audio System," SAE Technical Paper 920081, 1992, doi: 10.4271/920081
- [3] Tatsuo Ito and Isogai, Y., "Digital Audio Signal Processor: Current Status and future Trends" FujiTsuTenTech.J. No.3, 1993
- [4] Edwin J. Tan, Wendi B. Heinzelman, "DSP Architectures: Past, Present and Future", University of Rochester, Rochester, NY 14627
- [5] Suresh Marisetty, Durgesh Srivastava, Joel Andrew Hoffmann, An Architecture for in vehicle infotainment system. R. Nicole, "Title of paper with only first word capitalized," J. Name Stand. Abbrev., in press.
- [6] Y. Yorozu, M. Hirano, K. Oka, and Y. Tagawa, "Electron spectroscopy studies on magneto-optical media and plastic substrate interface," IEEE Transl. J. Magn. Japan, vol. 2, pp. 740-741, August 1987 [Digests 9th Annual Conf. Magnetics Japan, p. 301, 1982].
- [7] M. Young, The Technical Writer's Handbook. Mill Valley, CA: University Science, 1989.

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