The Successful Implementation of High-Performance Digital Radio

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Evolution of Digital Radio

Amplitude modulation (AM) was the dominant form of radio broadcasting during the first 80 years of the 20th century, but channel fading, distortion, and noise led to poor reception quality. These problems were reduced to some extent with the introduction of frequency modulation (FM), which could also provide stereo transmission and CD-quality audio, but analog radio was still not devoid of channel imperfection effects and limited coverage area. During 2003, two commercial start-ups, XM and Sirius (these merged and became SiriusXM™), introduced the huge footprint of subscription-based digital satellite radio in the United States, with a revenue model similar to that of Pay-TV channels. Around the same time, WorldSpace Radio started satellite broadcasts for Asia and Africa.

The Satellite Digital Audio Radio Services (SDARS) enabled mobile car audio listeners to tune into the same radio station anywhere within the satellite's coverage map, limited only by intermittent blockage of satellite signal due to buildings, foliage, and tunnels. XM satellite radio took the lead in circumventing the blockage problem by installing terrestrial repeaters, which transmit the same satellite audio in dense urban areas and create a hybrid architecture of satellite and terrestrial broadcasts.

Around the same time the *traditional* terrestrial broadcasters also charted a digital course—for two reasons. First, they perceived that their life span on the *analog* concourse had to be quite short, as the world migrates to the higher quality *digital* runway. Second, the frequency spectrum is getting scarce, so additional content within the same bandwidth could be delivered only by digitizing and compressing the old and new content, packaging it, and then broadcasting it. Thus, the world started migrating from analog to digital radio. These techniques for radio broadcast had the advantages of clearer reception, larger coverage area, and ability to pack more content and information within the existing bandwidth of an available analog radio channel—as well as offering users increased control flexibility in accessing and listening to program material (Figure 1).

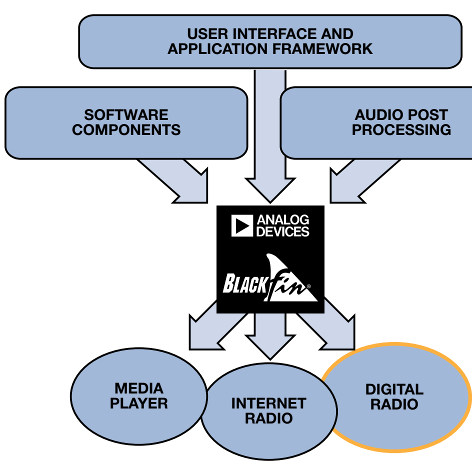

Figure 1. Digital radio on a convergence processor.

Digital Radio Development Example: India

In terrestrial broadcasting, there are two open standards—Digital Multimedia Broadcasting (DMB) and Digital Radio Mondiale™ (DRM)—and HD Radio,™ a proprietary standard from iBiquity (the only standard approved by the FCC for AM/FM audio broadcasting within the United States). DMB specifies several formats for digital *audio broadcasting*, including DAB, DAB+, and T-DMB, which use VHF Band III and L-band. DRM uses DRM30, which operates from 150 kHz to 30 MHz, and DRM+ in VHF Bands I, II, and III.

Useful propagation in the VHF bands is essentially limited to line-of-sight in small geographic regions. Propagation in short wave, on the other hand, can go almost anywhere in the world due to multiple reflections in the ionosphere. For countries that are densely populated and have small geographic regions, DMB transmitting in VHF Band III and L-Band functions very efficiently. For countries that have large geographic areas, transmissions in medium and short wave provide effective coverage. For this reason, after a few years of trials of DAB and DRM, India decided to adopt DRM.

During 2007, All India Radio (AIR), Asia-Pacific Broadcasting Union (ABU), and the DRM Consortium conducted the first field trial for DRM in New Delhi. The experimental trial was conducted over three days with three transmitters, with measurements of various parameters. Besides these tests in New Delhi, AIR also did these measurements at long distances. It became clear that DRM had the advantage of serving a larger population with a limited number of transmitters. In addition, the increasing need for energy conservation raises power saving considerations to paramount importance. DRM's 50% greater power efficiency plays a vital role in supporting the ecology and a "greener" Earth.

Digital Radio Receivers and DSP

The physical world is *analog*, yet scientists and engineers find it easier to do a lot of computation and symbol manipulation in the *digital* domain. Thanks to sampling theory, signal processing, and available data converters, the way is smoothly paved for engineers to design, implement, and test complex *digital signal-processing* (DSP) systems using *analog-to-digital* converters (ADCs) and *digital signal processors* with programmable cores.

Development of powerful and efficient DSPs—along with advancements in information and communication theory—enabled the convergence of media technology and communications. Digital radio owes its existence to these technological advances.

Digital radio receivers were initially designed as lab prototypes and then moved to pilot production. Like most technologies, the first generation products are generally assembled using discrete components. As the market size and competition increase, manufacturers find that markets can be further expanded by bringing down the price of the finished product. The prospect of higher volume attracts semiconductor manufacturers to invest in integrating more of these discrete components to bring down the cost. With time, the shrinking silicon geometries lead to further cost reductions and improvements in the product's capability. Such has been the continuing evolution in many products, including FM radios and mobile phones.

Signal Processing in Digital Radio

A typical digital communication system (Figure 2) converts the analog signal into digital, compresses it, adds error-correction code, and packs several signals to make best use of the channel capacity. To transmit RF signals (which exist in the "real" world of analog energy), the digital signal is converted into analog and modulated on a carrier frequency for transmission. At the receiver, the reverse process takes place, starting with demodulating the carrier frequency. The signal is then converted to digital, checked for errors, and decompressed. The baseband audio signal is converted to analog, ultimately producing acoustic sounds.

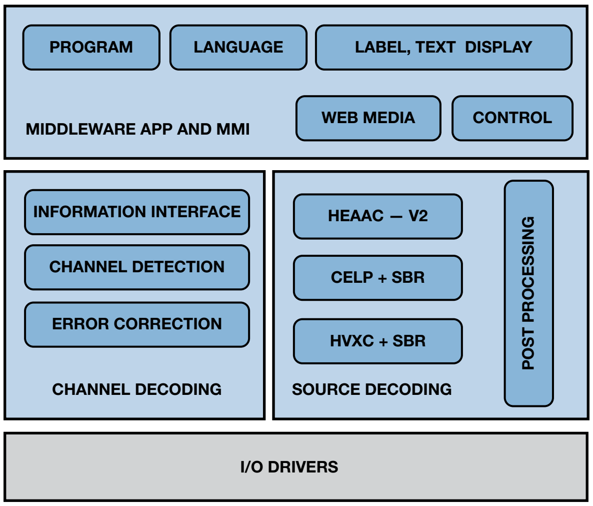

Figure 2. Software architecture of digital radio.

Signal processing algorithms in a digital radio receiver can be classified into the following categories:

- Channel decoding
- Source decoding
- Audio post processing
- Middleware
- User interface (MMI)

In digital radio, the *source coding* and *channel coding* can respectively be mapped to an efficient audio *codec* (coder-decoder) and *error control* system components. Practically, error control can be performed better if the codec is designed for error resilience.

An ideal channel coder should be resilient to transmission errors. An ideal source coder should compress the message to the highest information content (Shannon entropy), but highly compressed messages would lead to very high audio distortion if the input stream contains errors. Thus, effective source coding should also ensure that the decoder can detect the errors in the stream and conceal their impact so that overall audio quality is not degraded.

DRM applies relevant technological innovations in source coding and channel coding to deliver a better audio experience. The DRM audio source coding algorithm that is selected ensures:

- Efficient audio coding—higher audio quality with lower bit rate.
- Better error resilience—aesthetic degradation under transmission errors.

Efficient Audio Source Coding

Motion Picture Experts Group (MPEG) technology can be considered as the conduit and framework for effective collaboration of academic, industry, and technology forums. Success of such collaborative audio-specific efforts as MPEG Layer II, MP3, and AAC (advanced audio coding) for broadcasting and storage/distribution, respectively, has encouraged the industry to engage in further research initiatives. MP3 continues to be the most popular 'unofficial' format for web distribution and storage, but simpler licensing norms—and Apple's decision to adopt AAC as the media form for the iPod—have helped AAC to get more industry attention than MP3.

Let us consider AAC from the MPEG community to understand some of the important technologies involved in source coding. *Psycho acoustic model* (Figure 3) and *time-domain alias cancellation* (TDAC) can be considered as two initial breakthrough innovations in wideband audio source coding.

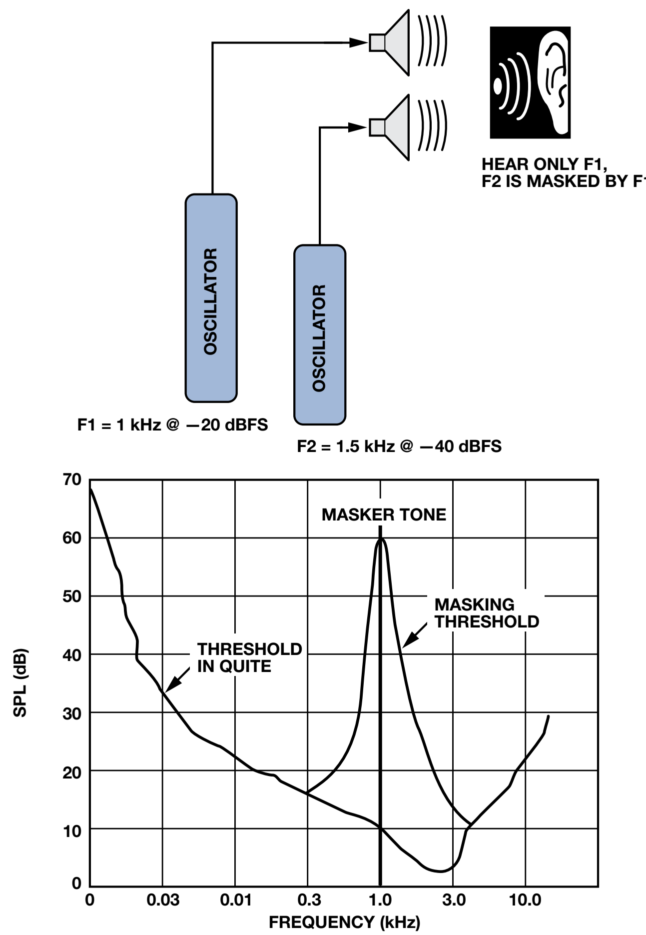

Figure 3. Understanding psycho-acoustic tonal masking.

Spectral band replication (SBR, Figure 4) and *spatial audio coding* or *binaural cue coding* techniques from industry and academia can be considered as the next two game-changing innovations. These two key breakthrough innovations further enhanced AAC technologies to give scalable coding performance, which resulted in standardization of HE-AAC v2 and MPEG *surround*—which received overwhelming responses from the industry. Industry-driven standards, like Dolby,® AC3, and WMA,® also took similar steps to leverage similar technological innovations for their latest media coding.

The *spectral band replacement* (SBR) tool doubles the decoded sample rate relative to the AAC-LC sample rate. The *parametric stereo* (PS) tool decodes stereo from a monophonic LC stream.

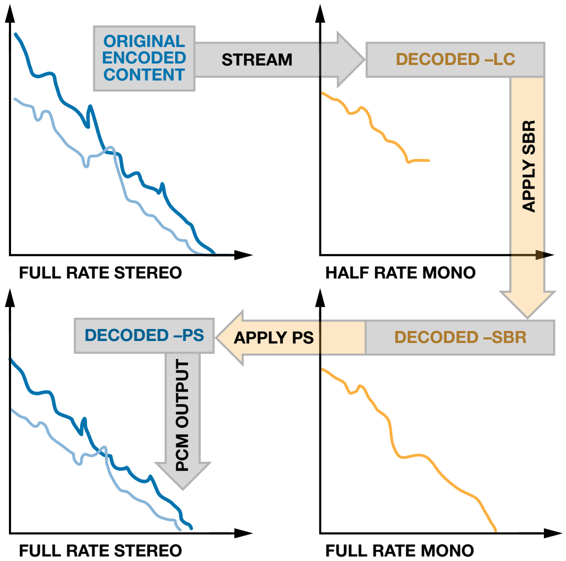

Figure 4. AAC-LR, SBR, and PS in audio decoding.

Like any other improvement initiatives, measurement technologies also played their role in audio quality improvement initiatives. Audio quality evaluation tools and standards, like *perceptual evaluation of audio quality* (PEAQ) and *multi-stimulus with hidden reference and anchor* (MUSHRA), aided faster evaluation of technological experiments.

Graceful Degradation/Error Resilience

In general, higher compression will lead to more audio artifacts from a given level of stream errors. For example, the MPEG Layer II stream is more robust to stream errors than AAC streams. A single-bit error in spectral data part of Layer II wouldn't make any annoying artifacts, as the spectral value maximum is decided by the bit-allocation value. However, in the case of AAC, the same single-bit error would cause the Huffman decoder to fail and apply frame error concealment; repetitive frame errors will mute the audio until the error rate is reduced to minimum. This long silence prevents the system from guaranteeing graceful degradation.

Error Resilience (ER) AAC coding guarantees graceful degradation from bit-stream errors with the help of these additional tools:

- HCR (*Huffman codeword reordering*): error propagation within spectral data is prevented by dividing the spectral data into fixed size segments. HCR places the most important data at the start of each segment.
- VCB11 (*virtual codebooks for codebook 11*): detects serious errors within spectral data with the help of special code word mapping.
- RVLC (*reversible variable length coding*): avoids error propagation in scale factor data.

The ER-AAC features, together with UEP, will provide adequate error resilience characteristics for DRM.

DRM Specification

Digital Radio Mondiale (DRM) is an open standard from European Telecommunication Standards Institute (ETSI) for digital narrow-band audio for short and medium-wave broadcasting. Although DRM supports bandwidths of 4.5 kHz, 5 kHz, 9 kHz, 10 kHz, 18 kHz, and 20 kHz within four modes of transmission and reception, bandwidth and bit rate must be limited to 10 kHz and 24 kbps, respectively, if compatibility with existing AM standards is desired.

Table 1. DRM bit-rate-bandwidth.

Bandwidth at 30 MHz	Bandwidth (kHz)	Bit Rate (kbps)
Nominal BW	9 to 10	8 to 20
Half BW	4.5 to 5	2 or 4
Double BW	18 to 20	20 to 80

This requirement demanded the use of highly efficient audio coding: Meltzer-Moser MPEG-4 HE-AAC v2 (International Standardization Organization/International Electrotechnical Commission—ISO/IEC) was a good choice, but the robustness against channel fading made an error-resilient version of HE-AAC v2 (Martin Wolters, 2003) the *best* choice.

Table 2. Different codecs supported by DRM.

Bit Rate (kbps)	20 to 80	8 to 20	2 to 4
Codec	AAC	CELP	8 to 20
Audio Rate	12, 24, or 48	8 to 16	2 or 4
SBR	Yes	Yes	Yes
PS	Yes	—	—
Double BW	Yes	Yes	Yes

Besides AAC, the DRM standard defines the harmonic vector excitation coding (HVXC) and code-excited linear prediction (CELP) codecs to be used for transmitting speech. Streaming raw data for image slideshows, HTML pages, and the like is also allowed by the DRM standard.

DRM Architecture

A DRM system comprises three main transmission paths: main service channel (MSC), service description channel (SDC), and fast-access channel (FAC). The FAC carries the orthogonal frequency-division multiplexed (OFDM) signal properties and the SDC/MSC configuration—and is limited to 72 bits/frame. The SDC contains the information needed for MSC decoding, such as the multiplex frame structure—and other information.

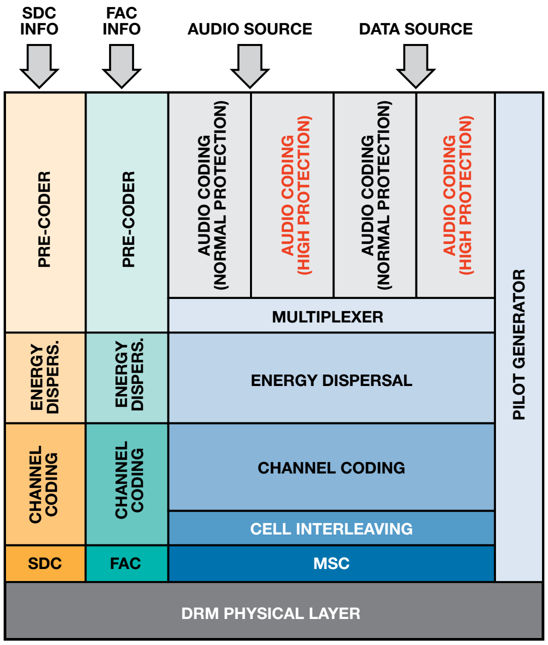

Figure 5. Multiplexing and channel coding in DRM.

The MSC encodes the frame generated by the multiplexer. One can choose between standard mapping, symmetrical hierarchical, or mixed hierarchical mapping. The MSC uses unequal error protection (UEP, Figure 6), in which the multiplex frame is split into two parts with different levels of protection: higher- and lower-protected data parts.

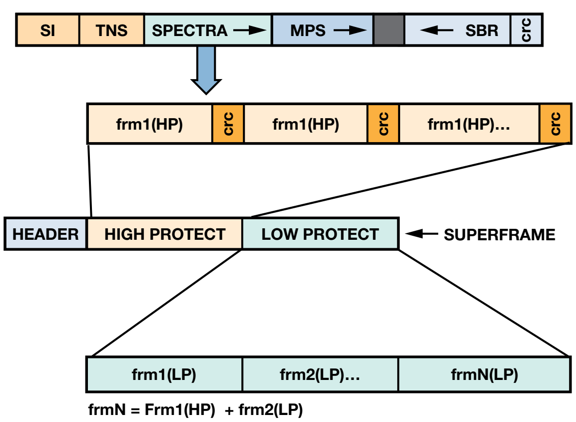

Figure 6. Unequal error protection in DRM.

Digital Radio with Blackfin

The Blackfin® processor (Figure 7) is an excellent fit for operations requiring both digital signal processing and a microcontroller function. The ADSP-BF5xx family is particularly suitable for these applications and also offers a variety of peripherals. Availability of hardware and software development tools, several software components from third parties, and reference designs make it an ideal platform for multifeatured products. Multiple generations of products, availability of mature software IP from dependable sources, reliable support from ADI, and the large portfolio of available high-performance analog integrated circuits support the quality of a designer's end products.

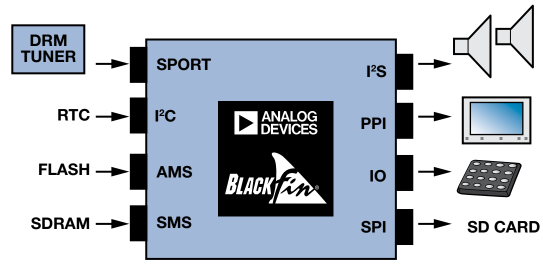

Figure 7. Blackfin processor based digital radio.

Blackfin processor based digital radios, Internet radios, and multi-featured products can be created using the existing ecosystem that ADI created for these products.

In addition to creating the required ecosystem and sourcing the various software modules, ADI also created its own decoder libraries for digital radio. One such key component is an HE-AAC v2 decoder, which optimizes the performance available from the large number of required MIPS.

Architecture of HE-AAC V2 Decoder

HE-AAC v2 decoder components (Figure 8) combine to form the DRM source decoder. The MPEG-4 HE-AAC v2 decoder (which can support ETSI DAB and DRM standards) combines advanced audio coding (AAC), spectral band replication (SBR), and parametric stereo (PS). The decoder is backward compatible with AAC-LC.

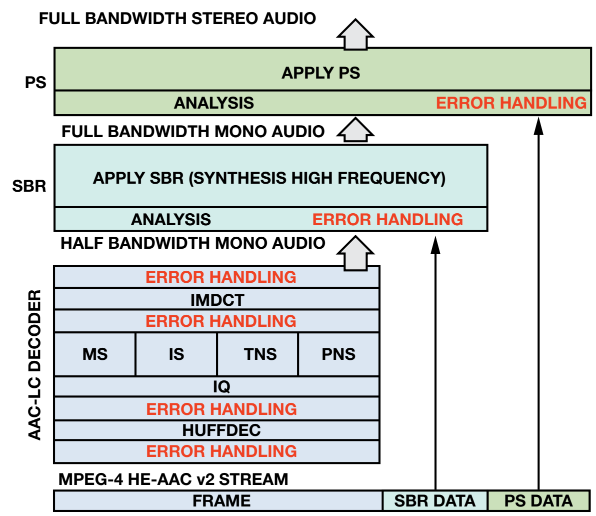

Figure 8. MPEG-4 HE-AAC v2 decoder.

Key features include:

- MPEG-4 ER-AAC scalable decoder that can handle 960 samples per frame
- AAC-LC/HE-AAC v1/v2/DRM/DAB support
- Error concealment support
- DRC support
- Highly optimized for memory and MIPS
- Validates against a complete set of ISO/DAB/DMB and ETSI vectors.

Table 3. MPEG-4 HE-AAC v2 decoder performance.

Memory in kB	Code	Table	Data	MIPS
DAB	115	61	182	8 to 20
DRM	115	62	182	2 or 4

The decoder implements all required audio coding tools specified by the standard, including:

- Higher frequency resolution and coding efficiency due to MDCT/TDAC
- Adaptive block switching reduces pre-echo
- Nonlinear quantization
- Huffman coding
- Use of Kaiser-Bessel derived window function to eliminate spectral leakage.
- Variable frame-size improves bit-allocation
- IS/MS stereo/TNS and PNS tools
- Spectral band replication (SBR)
- Parametric stereo (PS)

Digital Radio Test Results

A set of typical test results appears in Table 4.

Table 4. Digital radio test results.

Parameter	Results
Sensitivity	40 dB
Half BW	5 dB better than MRR
Inter-Modulation	>57 dB
Dynamic Range	25 dB more than MRR
Adj. Ch. Suppression	MRR +5 dB at ±10 kHz
Reception Freq. Offset	400 Hz better than MRR
Operating Voltage	6.5 V to 12 V

Conclusion

Analog Devices, Inc., (ADI) was an early participant in implementing digital radio and performing field trials of the reference design. A Blackfin processor-based DRM radio was one of the first designs that met all *minimum receiver requirements* (MRR) stipulated by the DRM standards. That success can be attributed to excellent teamwork, in which ADI managed and partnered with BBC in the UK, Dolby (erstwhile Coding Technologies) in the US, and Deutsche Welle and AFG Engineering in Germany. The technology and reference design was then adopted by apparatus manufacturers to engineer and make products.

Now, additional companies are using this design for making digital radios in India and other countries. The ADI Blackfin processor has the right combination of DSP and microcontroller features to form the core of a very cost effective DRM radio receiver. Availability of software tools, support by experienced applications teams, and the required software modules and reference designs from third parties make this implementation a good choice for manufacturers in India and elsewhere to adopt the design and mass produce DRM radios that use it.

References

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