Source Codecs for Terrestrial and Satellite based Broadcast Systems in the US and Europe

Thomas Ziegler, Coding Technologies, Nürnberg, Germany; zie@CodingTechnologies.com
Daniel Homm, Coding Technologies, Nürnberg, Germany; homm@CodingTechnologies.com
Reinhold Böhm, Coding Technologies, Nürnberg, Germany; boe@CodingTechnologies.com

Abstract

A new generation of digital audio broadcast systems has established in the past years. The common characteristic is the need for an extremely low bit rate for the source coder. Reasons are e.g. the requirement to offer a very large number of simultaneous audio channels or an extremely narrow banded RF-carrier. Bit rates down to about 20 kbps are common. Today, only codecs using Spectral Band Replication (SBR) technology reach high quality audio while maintaining the full bandwidth at these low bit rates. The combination of SBR with Parametric Stereo (PS) technology offers a perfect stereo image at a bit rate of 20 kbps. This paper will discuss three examples using SBR-enhanced codes in the digital broadcast domain: XM Satellite Radio, iBiquity's HD Radio and Digital Radio Mondiale (DRM).

1 Introduction

The idea of digital broadcast is not new. There are already systems in the satellite domain like Worldspace [1] and in the terrestrial domain, e.g. the European ETSI standard for terrestrial Digital Audio Broadcast (DAB) [2]. However, they are not too successful yet. The reasons are manyfold. One reason for the limited success of the previous generation of satellite based systems is the limited number of channels. DAB also does not provide substantial more channels than the existing analog FM radio. In addition the DAB modulation does require a new RF infrastructure. The limiting factor for the number of audio channels is in both examples the "old" audio codec: DAB is using MPEG-1 Layer 2, Worldspace MPEG-1 Layer 3 –both audio codecs where the standardization process has started in the late eighties of the previous century.

As a logical result, the current generation of digital broadcast systems is using State-of-the-Art audio codecs to exploit the available bandwidth efficiently. Low bit rate audio codecs are mandatory to achieve a high number of audio channels. Low bit rate audio codecs are also a mandatory requirement if the existing RF infrastructure should be "reused" for digital transmission. To achieve a broad market acceptance, the audio quality has to meet current standards; i.e. a CD-like experience is the minimum.

Today only audio codecs using Spectral Band Replication (SBR) technology are able to touch the sweet spot. SBR-enhanced codecs achieve an outstanding quality level at typical bit rates of 32 kbps (stereo) resp. 20 kbps (mono). As a natural consequence, the three broadcast systems addressed within this article are using SBR-enhanced audio codecs for source coding:

- XM Satellite Radio, a US based nation wide radio service offering up to 120 music and talk channels via satellite,

- iBiquity's HD Radio, offering a terrestrial radio service in the US using the existing FM- and AM-infrastructure and

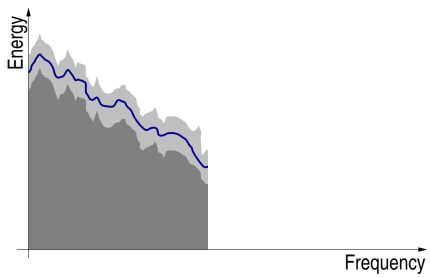
- Digital Radio Mondiale (DRM), a worldwide standard designed to replace the current short-, medium and long wave AM in a step-by-step approach.

In addition to SBR, the Parametric Stereo (PS) technology allows to further reduce the bit rate for a stereo signal. PS provides a transparent stereo image at typical bit rates of 20 kbps. PS is used in HD Radio and DRM. Both technologies have been developed nearly completely by Coding Technologies. Coding Technologies is also holding all major patents for Spectral Band Replication. It should also be mentioned that SBR- and PS-enhanced codecs are not limited to the broadcast domain. Among other standardization bodies, 3GPP has recently standardized aacPlus v2 – the combination of AAC, SBR and PS – as recommended codec for mobile phones.

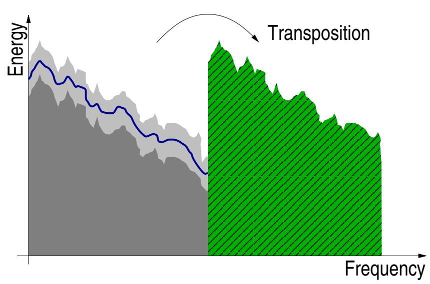
After a brief introduction of the technology behind SBR and PS, this article will focus in detail on the source codecs behind the broadcasting systems above.

2 The Technology

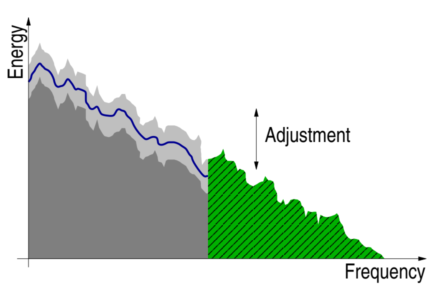
Spectral Band Replication and Parametric Stereo break with the paradigm of conventional perceptual audio codecs such as the series of MPEG audio codecs. A traditional codec is based on transmitting single spectral lines usually at a high frequency resolution. For a stereo signal, similarities between the left and the right channel are exploited but in general the single spectral line approach is also present for stereo coding.



(a) Traditional Perceptual coder operating at extremely low bit rates. Limitation of bandwidth avoids artifacts



(b) Creation of high frequencies by transposition of the low band into the high band



(c) Adjusting the spectral envelope of the highland to its original shape

Figure 1: SBR reconstructs the high frequencies based on the low frequencies in a perceptually accurate manner.

Spectral Band Replication and Parametric Stereo combine conventional coding with parametric coding in a psychoacoustic perfect manner.

2.1 Spectral Band Replication

In general SBR is a technology to increase the bandwidth of a traditional codec. Instead of transmitting the complete spectrum, the traditional codec (core codec) is operating only on the lower frequencies (low band) of the original signal. On the decoder side, the SBR-decoder re-creates the missing upper part of the spectral data (high band) by a “guided reconstruction” based on the low band and a very small amount of control data (SBR-data). Among other items, a parameterized shaping of the high band’s spectral envelope as outlined in Figure 1 is one technique used here. The amount of SBR-data varies in the range of 1-3 kbps resulting in a large gain of coding efficiency. Although the basic SBR principle is simple, the perceptually accurate reconstruction of the original audio signal is non-trivial. For more details on SBR, please refer to [3], [4], [5] and [6].

In general SBR could be combined with any core codec, e.g. also speech codecs. Some combinations have become international standards. E.g. the combination of AAC and SBR is called aacPlus v1™ and has been standardized by MPEG as HE-AAC [7, 8]. The main tool within the SBR-decoder (see Figure 2) is a high quality quadrature mirror filter bank (QMF). The tools recreating the missing high band operate in the QMF domain.

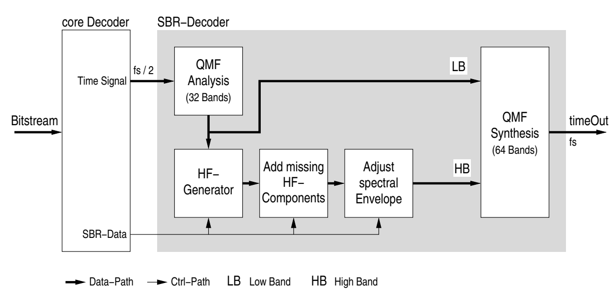


Figure 2: SBR decoder block diagram. The SBR-decoder operates as a post processor to the core decoder.

The quality improvement has been proven in many independent listening tests. A representative example is given in Figure 3. It shows the results of a MUSHRA listening test [10], carried out in the course of an Internet Audio comparison test of several low-bitrate audio codecs by seven independent EBU member organizations [9]. The MUSHRA[1] test has been explicitely designed by the European Broadcast Union (EBU, [10]) to

[1]**MU**ltiple **S**timulus with **H**idden **R**eference and **A**nchors

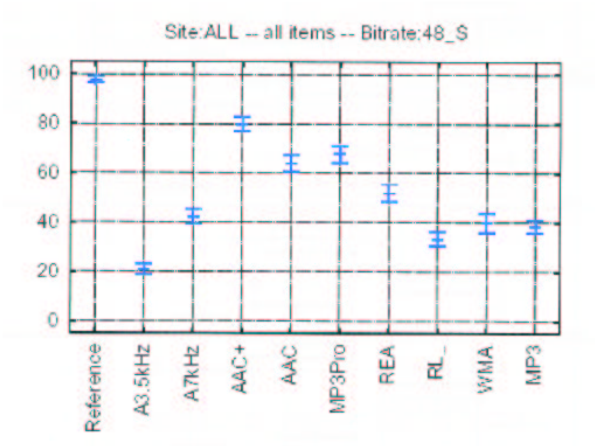


Figure 3: MUSHRA listening test results as taken from [9, page 15], comparing 28 items at the original full bandwidth reference, 3.5 kHz and 7 kHz band limited anchors, aacPlus, aac, mp3PRO, Real Audio 7, Windows Media 8, and mp3 – all codecs at a bitrate of 48 kbps, stereo

subjectively compare audio codecs. Each 20-point interval of the numeric scale is mapped to a quality level starting from 'bad' for the lowest interval via poor-fair-good to the highest one called 'excellent'. The combination of AAC + SBR performs significantly better than plain AAC at the same bit rate. For lower bit rates the difference is even larger.

2.2 Parametric Stereo

PS follows a similar principle as SBR does. Instead of transmitting two independent channels, only a mono signal (left plus right, mid) is transmitted from the encoder to the decoder. At the decoder, a guided reconstruction re-creates the missing stereo signal using only a small amount of parametric data. The major components are:

- Inter-channel Intensity Differences (IID),
- Inter-channel Cross Correlation (ICC)
- Inter-channel Phase Differences (IPD) and Overall Phase Difference (OPD)

Again the processing is done in the QMF domain which allows for an elegant combination of both SBR and PS technology. Two versions of PS with different characteristics and complexity are available:

- A simple version with low frequency and time resolution and
- a more sophisticated version with high resolution for both frequency and time.

The second version is based on the first one and has been a joint development between Coding Technologies and Philips. It has been standardized within MPEG [11], the more simple one within DRM [12]. The combination of AAC, SBR and MPEG version of PS is called aacPlus v2™.

Independent listening tests during the 3GPP standardization process (see Figure 4) show a significant improvement between aacPlus v1 (Codec1) and aacPlus v2 (Codec2).

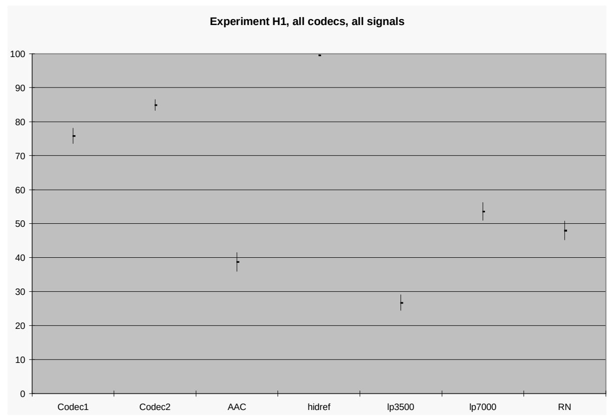


Figure 4: MUSHRA listening test results achieved in the course of the 3GPP standardization [13], comparing codecs at a bit rate of 32 kbps, stereo. Codec1 is aacPlus v1, Codec2 is aacPlus v2.

3 XM Satellite Radio

XM Satellite Radio is a satellite based digital broadcast system in the US with two geostationary satellites and terrestrial repeater networks in all major cities. Official service launch was end of 2001. The system is not restricted to stationary use but has been explicitely designed for the automotive area. All major car radio manufacturers offer XM Satellite car radios. XM-Radio provides up to 120 channels based on a monthly subscription fee, i.e. the contents are encrypted. The current source codec is aacPlus v1with a bit stream format adapted to the specific XM Satellite Radio requirements. Due to the excellent audio quality, XM Satellite Radio is by far the commercially most successful satellite broadcaster for mobile applications in the US with currently more than two million subscribers.

All current XM Satellite receivers are based on a receiver chipset manufactured by ST Microelectronics [14, 15]. The source decoder is implemented as a typical mask programmable 24-bit DSP-core with Harvard architecture and single-cycle MAC operations whereas the channel decoder has been realized as a ASIC [15].

Besides the SBR technology, Coding Technologies has also provided the optimized DSP software.

4 HD Radio

HD Radio™ is a terrestrial broadcast solution for FM and AM developed by iBiquity. The idea behind HD Radio is to keep the existing RF-infrastructure – namely the channel spacing but also existing hardware – from the analog world and to provide a smooth transition to the digital world. iBiquity is using the so-called In-Band-On-Channel (IBOC) technique to additionally transmit a digital signal on the side bands together with the analog carrier. Unlike as for DAB, this approach allows to decide on a per-station basis if switching to a digital service is acceptable. In a future scenario, the analog carrier might be replaced by a digital one.

It is obvious, that the side bands of the current analog radio do not allow high bit rates if there should be no interference with the next adjacent channels, which may eventually still operate in analog mode. This is especially true for the more narrow banded AM system. Only an extremely efficient audio codec is capable to provide the expected audio quality. HD Radios source codec is called HDC which is the combination of a proprietary perceptual core codec with Spectral Band Replication and Parametric Stereo. The system provides excellent audio quality in stereo for the FM- and the AM-modes. According to iBiquity more than 400 stations in the US have licensed HD Radio today.

Current receiver chipsets such as TI's award winning TMS320DRI250 [16] are based on a c64xx DSP. Both, channel decoding and source decoding are performed on that DSP, i.e. in pure software. TI and iBiquity have published a roadmap describing integration of additional blocks such as the AM/FM tuner at lower cost. A second single chip solution – manufacturer is Philips – has been presented at the CES-2004. The Philips solution (see Figure 5) is based an ARM-946E core with additional hardware accelerator units.

Besides the source decoder technology Coding Technologies' contribution includes optimized decoder libraries for both receiver chips set flavors.

5 Digital Radio Mondiale

Digital Radio Mondiale (DRM) has originally been developed as an European ETSI standard for digital broadcasting in the frequency range below 30 MHz, i.e. short, medium and long wave. In the meantime DRM has been adopted by the IEC (IEC 62272-1) and the ITU (ITU BS 1514). The goal is to offer an improved audio quality

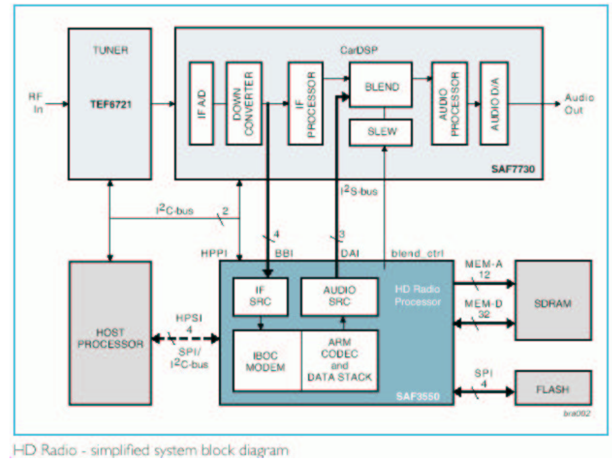


Figure 5: ARM-946E-based HD Radio block diagram from [17]. Source decoding is performed on the ARM-946E.

and to cope with channel effects such as fading or interference from other stations. The DRM system uses the same channel spacing and the same channel width as the analog system to allow a smooth transition like the IBOC system. The plan is to attract a large number of new listeners by just switching from analog to digital. Only by using SBR and PS in conjunction with AAC it was possible to reach the goal of high quality audio in stereo through the existing narrow banded AM-channels. To increase error robustness, Huffman Codeword Reordering (HCR) from the MPEG-4 Error Protection (EP) tools is used. For pure talk channels, DRM provides two speech codecs: HVXC and CELP. Both might also be used in combination with SBR. By using SBR, an improved intelligibility is achieved.

The official service launch was mid 2003. Today more than 60 broadcasters worldwide are transmitting live, daily or weekly DRM broadcasts and special programs.

The low bandwidth of the modulated DRM signal allows realtime A/D-conversion using the sound card of a typical personal computer. Also demodulation, channel decoding and source decoding can be completely realized as PC-software. There are commercial [18] and Open-Source [19] versions available. The commercial version is based on SDKs provided by Coding Technologies.

Today, several companies are developing DRM receiver chipsets. The goal of the Eureka project E!2390-DIAM also was to develop a DRM receiver chipset. The resulting chipset includes two ARM-cores, one for demodulation and channel decoding the other for source decoding. In addition to the source decoder technology, Coding Technologies has provided an optimized DRM decoder library for an ARM-946E.

6 Conclusion

A new generation of digital broadcast systems – satellite based as well as terrestrial – has successfully entered the market. Nearly all of them are using Spectral Band Replication technology developed by Coding Technologies – one of the leading companies in audio compression technology for mobile, broadcast and internet. Receiver chips sets for all digital broadcast systems are either available or under development. All major receiver chip set manufacturers such as TI, ST Microelectronics or Philips rely on optimized decoder libraries by Coding Technologies. Latest developments cover true 5.1 surround sound over HD Radio recently presented as an over-the-air demo at the NAB Radio 2004.

References

[1] Rainer Buchta, Stefan Meltzer, and Oliver Kunz. The WorldStar Sound Format. In *101st AES Convention, Los Angeles, November 8 - 11*, 1996. Preprint 4385.

[2] ETSI EN 300 401 V1.3.3, Radio Broadcasting Systems; Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers, May 2001.

[3] Thomas Ziegler, Martin Dietz, Kristofer Kjörling, and Andreas Ehret. aacPlus – Full Bandwidth Audio Coding for Broadcast and Mobile Applications. In *International Signal Processing Conference (ISPC), Dallas, March 31 - April 3*, 2003.

[4] Martin Wolters, Kristofer Kjörling, Heiko Purnhagen, and Daniel Homm. A closer look into MPEG-4 High Efficiency AAC. In *115th AES Convention, New York, October 10 - 13*, 2003. Preprint 5871.

[5] Andreas Ehret, Martin Dietz, and Kristofer Kjörling. State-of-the-Art Audio Coding for Broadcasting and Mobile Applications. In *114th AES Convention, Amsterdam, March 22 - 25*, 2003. Preprint 5834.

[6] Thomas Ziegler, Andreas Ehret, and Klaus Peichl. Implementations Aspects of aacPlus v2. In *Global Signal Processing Conference (GSPx), Santa Clara, Sept. 27 - 30*, 2004.

[7] ISO/IEC JTC1/SC29/WG11 MPEG, International Standard ISO/IEC 14496-3:2001/Amd. 1:2003, ”Information technology – Coding of audio-visual objects–”, Part 3: Audio, Amendment 1: Bandwidth Extension, 2003.

[8] ISO/IEC JTC1/SC29/WG11 MPEG, International Standard ISO/IEC 14496-3:2001/Amd. 1:2003/Cor. 1:2004, ”Information technology – Coding of audio-visual objects–”, Part 3: Audio, Amendment 1: Bandwidth Extension, Technical Corrigendum 1, 2004.

[9] EBU Project Group B/AIM Audio in Multimedia. The EBU Subjective Listening Tests on Low Bitrate Audio Codecs. Technical Report BPN049, EBU-UER, 2002.

[10] EBU Project Group B/AIM Audio in Multimedia. MUSHRA – EBU method for subjective listening tests of intermediate audio quality. Technical report, EBU, 2000. Draft Technical Recommendation BMC 607 (Rev. 1).

[11] ISO/IEC JTC1/SC29/WG11 MPEG, International Standard ISO/IEC 14496-3:2001/Amd. 2:2003, ”Information technology – Coding of audio-visual objects–”, Part 3: Audio, Amendment 2: Parametric Coding for High Quality Audio, 2003.

[12] ETSI ES 201 980 V2.1.1, Digital Radio Mondiale (DRM); System Specification, June 2004.

[13] Global Analysis Laboratory Report on 3GPP High-Rate Audio Codec Exercises. WWW page, 2003. `http://www.3gpp.org/ftp/tsg_sa/WG4_CODEC/TSGS4_30/Docs/S4-040172.zip`.

[14] STMicroelectronics. *XMRADIO SDARS Service Layer & Source Decoder*, 2003. Datasheet http://www.st.com/stonline/books/pdf/docs/-9803.pdf.

[15] STMicroelectronics. *XMRADIO SDARS Channel Decoder*, 2003. Datasheet http://www.st.com/-stonline/books/pdf/docs/9802.pdf.

[16] *TI and iBiquity introduce industry’s first single-chip AM/FM and HD Radio™Baseband plus a new Companion Analog front end*, 2004. `http://focus.ti.com/docs/pr/pressrelease.jhtml?prelId=sc03179`.

[17] Philips. *SAF3550 HD Radio™processor*, December 2003. Document order number 9397 750 12472.

[18] DRM. *The DRM Software Radio Website*, October 2004. `http://www.drmx.org`.

[19] TU Darmstadt. *Open-Source Software Implementation of a DRM-Receiver*, October 2004. `http://www.tu-darmstadt.de/fb/et/uet/fguet/mitarbeiter/vf/DRM/DRM.html`.