streaming

netcasting tools


# OPTICODEC-PC 1010 v3.1 STREAMING ENCODER

The Absolute Highest Quality
AAC/HE-AACv1/HE-AACv2 Encoder

With OPTICODEC-PC,
your low bitrate netcast
doesn't have to sound
like it is underwater.
Thanks to the aacPlus codec,
you can attract and hold a mass audience
with broadcast-quality audio —

*without*
breaking the bank.

**NOW SUPPORTS:**

RTMP Flash Media Server

RTMP Wowza Media Server

RTSP/RTP QTSS/DSS Servers

RTP Real/Helix ™ Mobile Servers

HTTP/ICY SHOUTcast ™ / Icecast2 Servers

OPTICODEC-PC is the first MPEG-4 AAC/HE-AACv1/HE-AACv2 aka aacPlus™ encoding software for high quality streaming audio. OPTICODEC-PC offers the most important feature that the basic netcaster is looking for in an encoding product — entertainment-quality sound at economical bitrates.

AAC/HE-AACv1/HE-AACv2 aka aacPlus™ is changing the way streaming audio and netcasting is perceived. For a given bitrate, it sonically outperforms any other codec currently available. It is perfect for crowded Intenet infrastructures and mobile streaming.

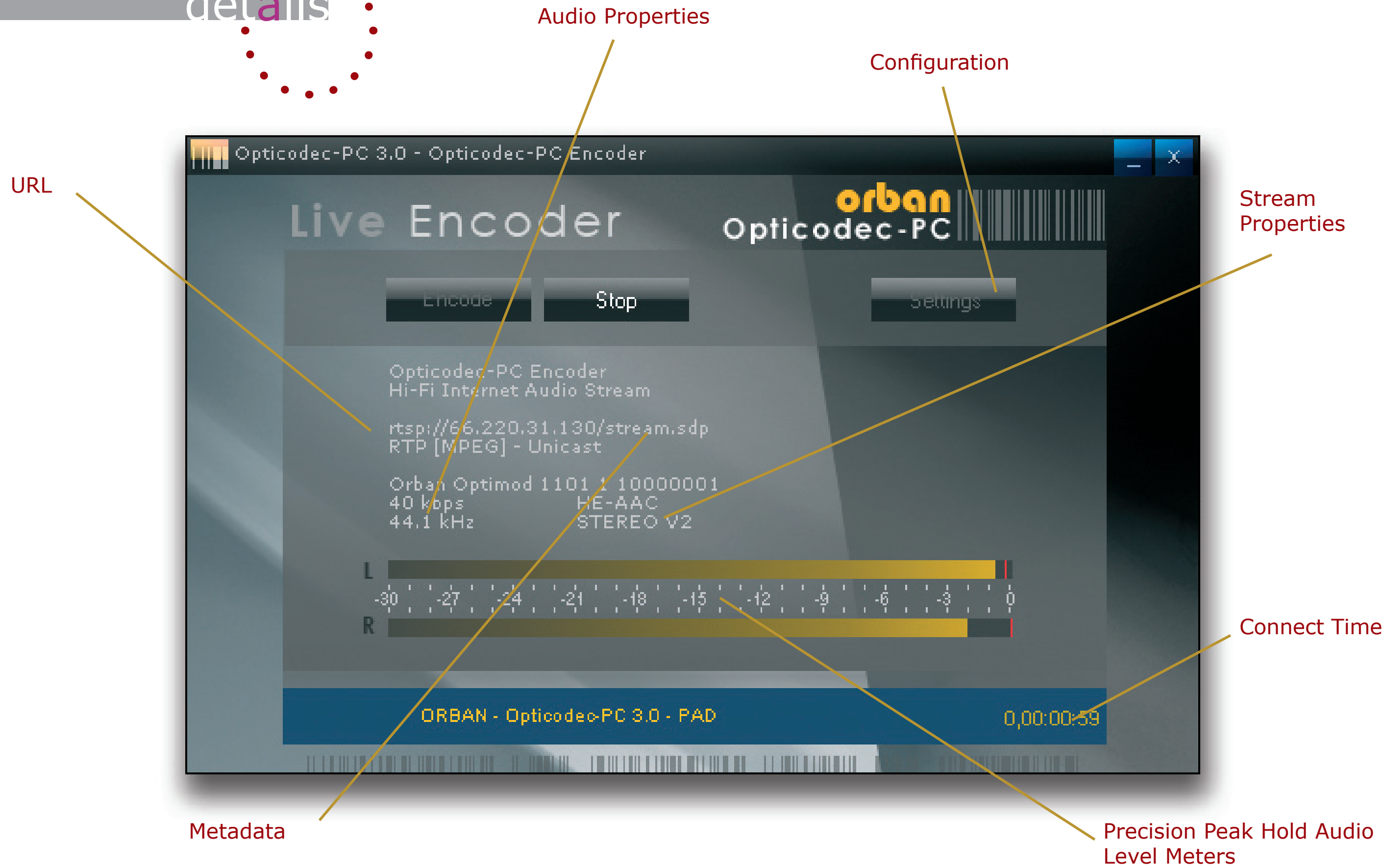
The software lets streaming providers supply content encoded with the Coding Technologies® AAC / HE-AACv1 / HE-AACv2 / aacPlus codec, widely acknowledged as offering the highest available audio quality at the lowest possible bitrate. Streams encoded with OPTICODEC-PC can be experienced through several players, including Adobe Flash®, RealPlayer®, Winamp®, Apple QuickTime®, and many iPhone Players such as Tuner2 HiFi. Streams can be listed on Tuner2, a Directory service for AAC / HE-AACv1 / HE-AACv2 streams.

## "Genuine Radio" ™ AAC / HE-AACv1/ HE-AACv2 / aacPlus Streaming from Orban


For the last 30 years, Orban's patented OPTIMOD technology has helped radio and television broadcasters everywhere shape their sound to grab and hold their listening audiences. Orban's OPTIMOD-PC 1101, a professional PCI sound card designed for streaming media, provides "genuine radio"™ audio processing for Internet broadcasters for professional smooth consistent level controlled audio. With three on-board DSP's providing equalization, AGC, multi-band compression, and look-ahead limiting, OPTIMOD-PC 1101, especially when combined with HE-AAC/aacPlus encoding technology, delivers a polished and produced stream that has the same loudness, consistency, and punch as major-market FM radio.

OPTICODEC-PC is available for Microsoft Windows® XP/Vista and 2003/2008 Server. It supplies streams compatible with the open standard RTSP/RTP free

# details




Darwin Streaming Server, available for multiple platforms including Linux®, FreeBSD®, Sun Solaris®, Microsoft Windows®, and Apple Macintosh®, the popular standard HTTP/ICY free Nullsoft SHOUTcast™ and Icecast2 servers, available for Microsoft Windows®, Linux®, FreeBSD® and Sun Solaris®.

OPTICODEC-PC PE is offered solely in a premium package coupled with an Optimod-PC and can encode multiple simultaneous streams at bitrates from 8 to 320 kbps, and additionally can encode streams targeting Flash, MPEG-4, and 3GPP mobile devices. OPTIMOD-PC ordinarily processes the resulting stream or file for consistency and punch, but it also comes with presets that allow it to do simple protection limiting.

In addition to sound card and audio processing functionality, OPTIMOD-PC 1101 is two separate sound devices, and also a capable mixer, having one stereo analog input, two AES3 / SPDIF digital inputs, and two wave inputs, all of which can be mixed or switched. Thanks to onboard samplerate converters, the two digital inputs can accept and mix asynchronous sources. In practice, the five inputs might be used for a local feed, a network feed, a voice channel, and two wave players, making OPTIMOD-PC the heart of a "desktop netcasting studio".

Thanks to separate "processed" and two "unprocessed" mixers, any of the inputs in any combination can be processed or passed directly to the input of OPTICODEC-PC without

the heart of a
professional
netcasting
studio

processing – the user can always choose how much processing (if any) to apply to the audio. Since 1101 is two separate Sound Devices, content insertion systems, such as commercial replacement no longer require two separate sound cards. It can all be done with OPTIMOD-PC 1101.

reduces bandwidth and costs

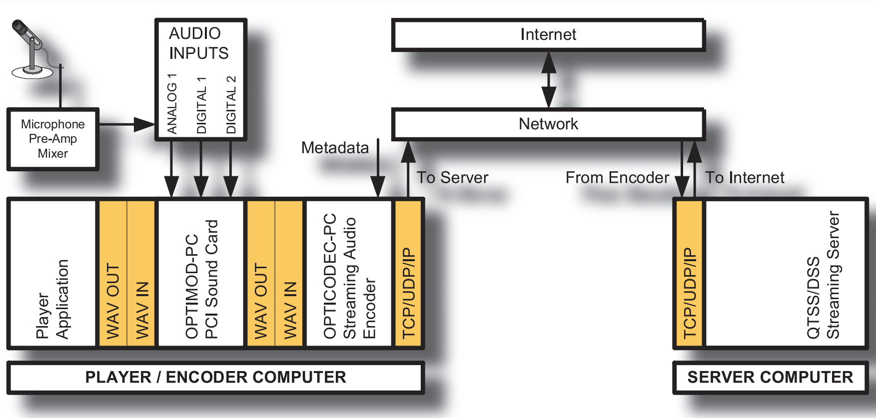
Compared to MP3, OPTICODEC-PC's aacPlus codec provides a better than 60% improvement in audio quality versus bitrate, reducing network streaming bandwidth requirements and costs accordingly. At 32 kbps, OPTICODEC-PC streams offer close to FM quality, without the phasey, watery character of other popular codecs operating at this bitrate. Many listeners prefer the audio quality of 48 kbps HE-AACv2 streams to FM and other codecs.

The marketplace has been screaming for broadcast quality from low bitrate Internet streaming and audio file serving. For the first time in this young industry, combining OPTIMOD audio processing with OPTICODEC-PC makes it possible to offer the sonic texture of major-market FM broadcasting via the Internet. Streaming is red hot, and OPTICODEC-PC is the fuel that puts you ahead of the pack.

"Streaming has just become profitable", said Rusty Hodge of netcaster SomaFM after first hearing the aacPlus codec at 32 kbps. Netcaster and respected broadcast engineer Gary Blau described it as "the quality breakthrough we've been waiting for".

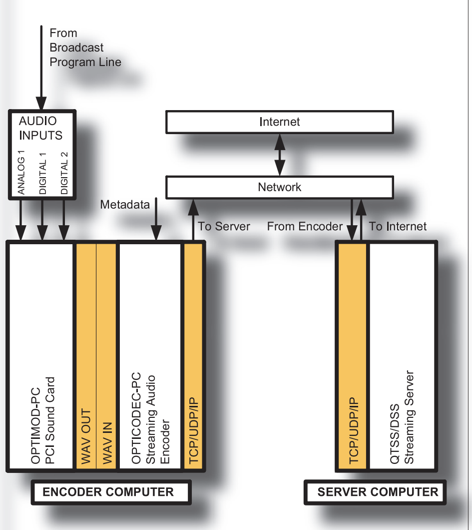
details

## Streaming Infrastructure

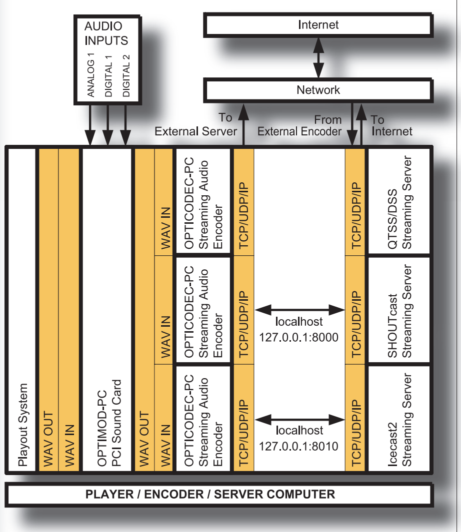



Typical streaming infrastructure where program material is sourced from a play-out system application with live assist

# Streaming Infrastructure




Typical streaming infrastructure where program material is sourced from a radio station on-air studio




Typical multiple streaming encoder/server infrastructure where program material is sourced from a player application

technology

## WHAT MAKES OPTICODEC-PC CODECS SOUND BETTER THAN OTHER POPULAR CODECS?

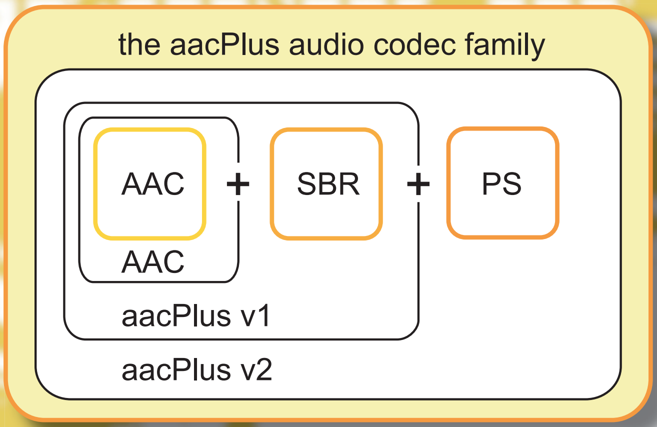
OPTICODEC-PC Streaming Audio Encoder uses Coding Technologies AAC/aacPlus codec technology, which combines three MPEG technologies: Advanced Audio Coding (AAC), Coding Technologies Spectral Band Replication (SBR) and Parametric Stereo (PS). SBR is a unique bandwidth extension technique that enables audio codecs to deliver the same quality at half the bitrate. Parametric Stereo increases the codec efficiency a second time for low bitrate stereo signals.

SBR and PS are forward and backward compatible methods to enhance the efficiency of any audio codec. AAC was chosen as the core codec for aacPlus because of its superior performance over older generation audio codecs such as MP3 or WMA. This was the reason why Apple Computer chose AAC for their market-dominating iTunes downloadable music service.

aacPlus delivers streaming and downloadable audio files at 48 kbps for FM-quality stereo, entertainment-quality stereo at 32 kbps, and good quality for mixed content even below 16 kbps mono. This efficiency makes new applications in the Internet, mobile, and digital broadcast markets viable. Moreover, unlike certain other proprietary codecs, AAC/aacPlus does not require proprietary servers for streaming.

## MEMBERS OF THE aacPlus CODEC FAMILY

aacPlus is the latest MPEG-4 Audio technology. aacPlus v1 combines AAC and SBR. aacPlus v2 builds on the success of aacPlus v1 and adds more value where highest compression efficiency for stereo signals is required. aacPlus v2 is a true superset of aacPlus v1, as aacPlus v1 is of AAC. With the addition of Parametric Stereo in MPEG, aacPlus v2 is the state-of-the-art low bitrate open-standards audio codec.




The members of the aacPlus codec family are designed for forward and backward compatibility. Besides aacPlus v2 bit streams, an aacPlus v2 encoder is also capable of creating aacPlus v1 and plain AAC bit streams

.

Every decoder is able to handle bit streams of any encoder, although a given decoder may not exploit all of the stream's advanced features. An aacPlus v2 decoder can fully exploit any data inside the bit stream, be it plain AAC, aacPlus v1 (AAC+SBR), or aacPlus v2 (AAC+SBR+PS). An AAC decoder decodes the AAC portion of the bit stream, not the SBR portion. As a result, the output of the decoder is bandwidth limited, as the decoder is not able to reconstruct the high frequency range represented in the SBR data portion of the bit stream.

If the bits stream is aacPlus v2, an AAC decoder will decode it as limited-bandwidth mono and an aacPlus decoder will emit a full-bandwidth mono signal; an aacPlus v2 decoder is required to decode the parametric stereo information.

## STANDARDIZATION

AAC/aacPlus is an open standard and not a proprietary format unlike other lower performing codecs. It is widely standardized by many international standardization bodies as follows:

MPEG-2 AAC ADTS
MPEG ISO/IEC 13818-7:2004 Advanced Audio Coding

MPEG-4 AAC
MPEG ISO/IEC 14496-3:2001 Coding of Audio-Visual Objects - Audio

MPEG-4 aacPlus v1 = AAC LC + SBR (aka HE AAC or AAC+)
MPEG ISO/IEC 14496-3:2001/AMD-1: Bandwidth Extension

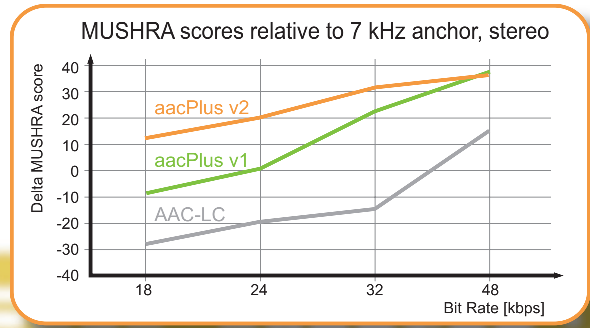
MPEG-4 aacPlus v2 = AAC LC + SBR + PS (aka Enhanced HE AAC or eAAC+)
MPEG ISO/IEC 14496-3:2001/AMD-2: Parametric Coding for High Quality Audio

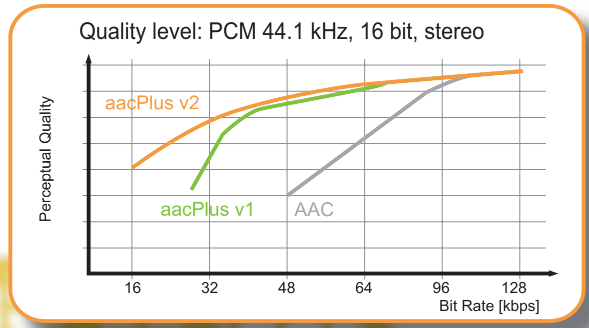
aacPlus v1 is standardized by 3GPP2 (3rd Generation Partnership Project 2), ISMA (Internet Streaming Media Alliance), DVB (Digital Video Broadcasting), the DVD Forum, Digital Radio Mondiale, and many others. aacPlus v2 is specified as the high quality audio codec in 3GPP (3rd Generation Partnership Project) and all of its components are part of MPEG-4.

As an integral part of MPEG-4 Audio, aacPlus is ideal for deployment with the new H.264/AVC video codec standardized in MPEG-4 Part 10. The DVD Forum has specified aacPlus v1 as the mandatory audio codec for the DVD-Audio Compressed Audio Zone (CA-Zone). Inside DVB-H, aacPlus v2 is specified for the IP-based delivery of content to handheld devices. ARIB has specified aacPlus v1 for digital broadcasting in Japan. S-DMB/MBCo has selected aacPlus v1 as the audio format for satellite multimedia broadcasting in Korea and Japan. Flavors of MPEG-4 aacPlus or its components are also applied in national and international standards and systems such as Digital Radio Mondiale (worldwide), iBiquity’s HD Radio (US), XM Satellite Radio (US), or the Enhanced Versatile Disc EVD (China).

## INDEPENDENT QUALITY EVALUATIONS OF aacPLUS

Independent tests have clearly demonstrated aacPlus v2's value. In rigorous double-blind listening tests conducted by 3GPP (3rd Generation Partnership Project), aacPlus v2 proved its superiority to its competitors even at bitrates as low as 18 kbps. aacPlus v2 provides extremely stable audio quality over a wide bitrate range, making it the first choice for all application fields in mobile music, digital broadcasting, and the Internet.




aacPlus v1 has been evaluated in multiple 3rd party tests by MPEG, the European Broadcasting Union, and Digital Radio Mondiale. aacPlus v1 outperformed all other codecs in the competition. Below is the results graph from the European Broadcasting Union testing at 48 kbps stereo. The full "EBU subjective listening test on low bitrate audio codecs" study can be downloaded at: http://www.ebu.ch/CMSimages/en/tec_doc_t3296_tcm6-10497.pdf.

## SPECTRAL BAND REPLICATION

Low bitrate audio coding is an enabling technology for a number of applications like digital radio, Internet streaming (netcasting/webcasting) and mobile multimedia applications. The limited overall bandwidth available for these systems makes it necessary to use a low bitrate, highly efficient perceptual audio codec in order to create audio that will attract and hold listeners.

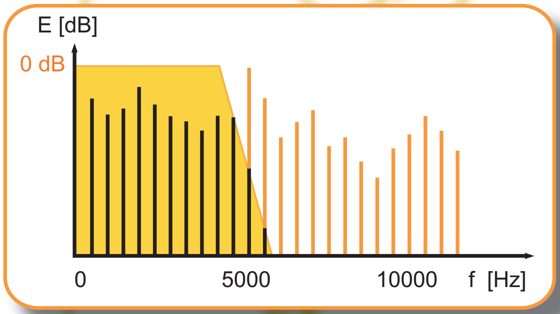
In Internet streaming applications, the connection bandwidth that can be established between the streaming server and the listener's client player application depends on the listener's connection to the Internet. In many cases today, people use analog modems or ISDN lines with a limited data rate — lower than the rate that can produce appealing audio quality with conventional perceptual audio codecs. Moreover, even if consumers connect to the Internet through high bandwidth connections such as xDSL or CATV, the ever-present congestion on the Internet limits the connection bitrate that can be used without audio dropouts and rebuffering. Furthermore, when netcasters pay for bandwidth by the bit, using a highly efficient perceptual codec at low bitrates can make netcasting profitable for the first time.

In mobile communications, the overall bandwidth available for all services in a certain given geographic area (a network cell) is limited, so the system operator must take measures to allow as many users as possible in that network cell to access mobile communication services in parallel. Highly efficient speech and audio codecs allow operators to use their spectrum most efficiently. Considering the impact that the advent of multimedia services has on the data rate demands in mobile communication systems, it is clear that even with CDMA2000, EDGE, and UMTS, cellular networks will find it necessary to use perceptual codecs at a relatively low data rate.

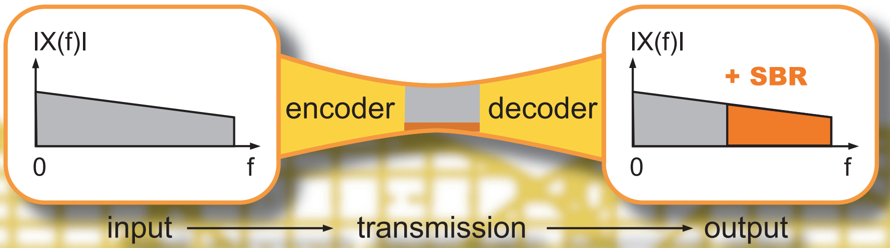
Using perceptual codecs at low bitrates, however, is not without its downside. State-of-the-art perceptual audio codecs such as AAC, achieve "CD-quality" or "transparent" audio quality at a bitrate of approximately 128 kbps (~ 12:1 compression). Below 128 kbps, the perceived audio quality of most of these codecs begins to degrade significantly. Either the codecs start to reduce the audio bandwidth and to modify the stereo image or they introduce annoying coding artifacts caused by a shortage of bits when they attempt to represent the complete audio bandwidth. Both ways of modifying the perceived sound can be considered unacceptable above a certain level. At 64 kbps for instance, AAC either would offer an audio bandwidth of only ~ 12.5 kHz or introduce a fair amount of coding artifacts. Each of these factors severely affects the listening experience.

SBR (Spectral Band Replication) is one of the newest audio coding enhancement tools. It can improve the performance of low bitrate audio and speech codecs by either increasing the audio bandwidth at a given bitrate or by improving coding efficiency at a given quality level.

SBR can increase the limited audio bandwidth that a conventional perceptual codec offers at low bitrates so that it equals or exceeds analog FM audio bandwidth (15 kHz). SBR can also improve the performance of narrow-band speech codecs, offering the broadcaster




example of band-limiting of a typical signal speech-only channels with 12 kHz audio bandwidth used for example in multilingual broadcasting. As most speech codecs are very bandlimited, SBR is important not only for improving speech quality, but also for improving speech intelligibility and speech comprehension. SBR is mainly a post-process, although the encoder performs some pre-processing to guide the decoding process.




From a technical point of view, SBR is a method for highly efficient coding of high frequencies in audio compression algorithms. When used with SBR, the underlying coder is only responsible for transmitting the lower part of the spectrum. The higher frequencies are generated by the SBR decoder, which is mainly a post-process following the conventional waveform decoder. Instead of transmitting the spectrum, SBR reconstructs the higher frequencies in the decoder based on an analysis of the lower frequencies transmitted in the underlying coder. To ensure an accurate reconstruction, some guidance information is transmitted in the encoded bitstream at a very low data rate.

The reconstruction is efficient for harmonic as well as for noise-like components and permits proper shaping in both the time and frequency domains. As a result, SBR allows full bandwidth audio coding at very low data rates and offers significantly increased compression efficiency compared to the core coder.

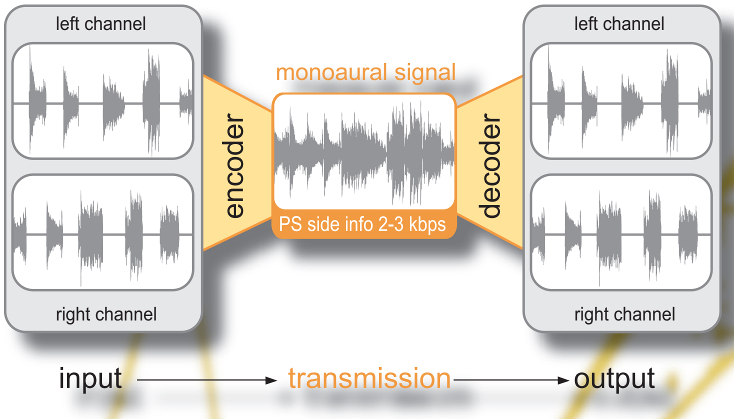
SBR can enhance the efficiency of perceptual audio codecs by ~ 30% (even more in certain configurations) in the medium to low bitrate range. The exact amount of improvement that SBR can offer also depends on the underlying codec. For instance, combining SBR with AAC achieves a 64 kbps stereo stream whose quality is comparable to AAC at 128 kbps stereo. SBR can be used with mono and stereo as well as with multichannel audio.

SBR offers maximum efficiency in the bitrate range where the underlying codec itself is able to encode audio signals with an acceptable level of coding artifacts at a limited audio bandwidth.

## PARAMETRIC STEREO

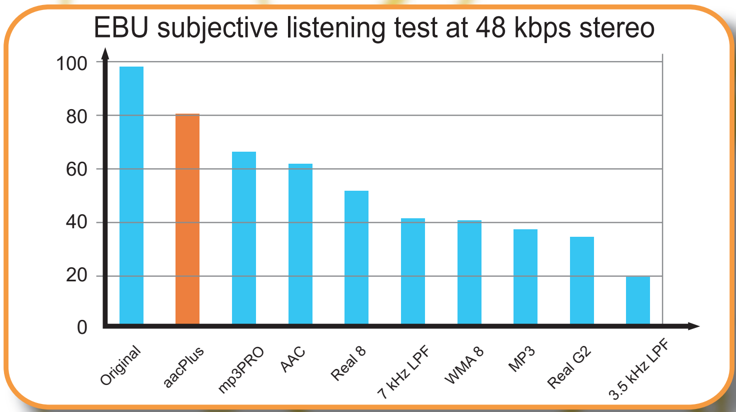
Parametric Stereo is the next major technology to enhance the efficiency of audio compression for low bitrate stereo signals. Parametric Stereo is fully standardized in MPEG-4 and is the new component within aacPlus v2. As of today, Parametric Stereo is optimized for the range of 16 – 40 kbps and provides high audio quality at bitrates as low as 24 kbps.

The Parametric Stereo encoder extracts a parametric representation of the stereo image of an audio signal. Meanwhile, a monophonic representation of the original signal is encoded via AAC+SBR. The stereo image information is represented as a small amount of high quality parametric stereo information and is transmitted along with the monaural signal in the bit stream. The decoder uses the parametric stereo information to regenerate the stereo image. This improves the compression efficiency compared to a similar bit stream without Parametric Stereo.




## PERFORMANCE COMPARISON

OPTICODEC-PC Streaming Audio Encoder uses Coding Technologies AAC/aacPlus codec technology. Multiple 3rd-party tests have demonstrated the value of aacPlus. In careful double-blind listening tests conducted by Digital Radio Mondiale, MPEG, and the European Broadcasting Union, aacPlus outperformed all other codecs in the comparisons.



CODEC PARAMETERS

## aacPlus CONFIGURATIONS

| aacPlus Mode | Channel Mode | Bitrate [kbps] | Sample Frequency [kHz] | Audio Bandwidth [kHz] |
|---|---|---|---|---|
| aacPlus | 1ch-Mono | 8 | 24 | 8.3 |
| aacPlus | 1ch-Mono | 10 | 24, 32 | 10.9, 11.0 |
| aacPlus | 1ch-Mono | 12 | 24, 32, 44.1 | 10.7, 11.0, 11.4 |
| aacPlus | 1ch-Mono | 16 | 32, 44.1, 48 | 11.7, 12.0, 12.3 |
| aacPlus | 1ch-Mono | 20 | 32, 44.1, 48 | 14.5, 14.8, 15.4 |
| aacPlus | 1ch-Mono | 24 | 32, 44.1, 48 | 14.8, 15.3, 15.4 |
| aacPlus | 1ch-Mono | 28 | 32, 44.1, 48 | 16.0, 16.2, 16.9 |
| aacPlus | 1ch-Mono | 32 | 32, 44.1, 48 | 16.0, 16.2, 16.9 |
| aacPlus | 1ch-Mono | 40 | 32, 44.1, 48 | 16.0, 17.6, 18.4 |
| aacPlus | 1ch-Mono | 48 | 32, 44.1, 48 | 16.0, 20.3, 20.3 |
| aacPlus | 1ch-Mono | 56 | 32, 44.1, 48 | 16.0, 20.3, 20.3 |
| aacPlus | 1ch-Mono | 64 | 32, 44.1, 48 | 16.0, 20.3, 20.3 |
| aacPlus | 2ch-Stereo | 24 | 32, 44.1, 48 | 12.7, 13.1, 13.8 |
| aacPlus | 2ch-Stereo | 32 | 32, 44.1, 48 | 14.8, 15.3, 15.4 |
| aacPlus | 2ch-Stereo | 40 | 32, 44.1, 48 | 16.0, 16.2, 16.9 |
| aacPlus | 2ch-Stereo | 48 | 32, 44.1, 48 | 16.0, 16.2, 16.9 |
| aacPlus | 2ch-Stereo | 56 | 32, 44.1, 48 | 16.0, 17.6, 18.4 |
| aacPlus | 2ch-Stereo | 64 | 32, 44.1, 48 | 16.0, 20.3, 22.1 |
| aacPlus | 2ch-Stereo | 80 | 32, 44.1, 48 | 16.0, 20.3, 22.1 |
| aacPlus | 2ch-Stereo | 96 | 32, 44.1, 48 | 16.0, 20.3, 22.1 |
| aacPlus | 2ch-Stereo | 112 | 32, 44.1, 48 | 16.0, 20.3, 22.1 |
| aacPlus | 2ch-Stereo | 128 | 32, 44.1, 48 | 16.0, 20.3, 22.1 |

## AAC CONFIGURATIONS

CODEC PARAMETERS

CODEC PARAMETERS

| AAC Mode | Channel Mode | Bitrate [kbps] | Sample Frequency [kHz] | Audio Bandwidth [kHz] |
|---|---|---|---|---|
| AAC | 1ch-Mono | 16 | 24 | 5.2 |
| AAC | 1ch-Mono | 20 | 24 | 7.2 |
| AAC | 1ch-Mono | 24 | 24, 32 | 7.2, 7.2 |
| AAC | 1ch-Mono | 28 | 24, 32 | 10.0, 10.0 |
| AAC | 1ch-Mono | 32 | 24, 32, 44.1, 48 | 10.0, 10.0, 10.0, 10.0 |
| AAC | 1ch-Mono | 40 | 24, 32, 44.1, 48 | 12.0, 12.4, 12.4, 12.4 |
| AAC | 1ch-Mono | 48 | 24, 32, 44.1, 48 | 12.0, 13.5, 13.5, 13.5 |
| AAC | 1ch-Mono | 56 | 24, 32, 44.1, 48 | 12.0, 15.5, 15.5, 15.5 |
| AAC | 1ch-Mono | 64 | 32, 44.1, 48 | 15.5, 15.5, 15.5 |
| AAC | 1ch-Mono | 80 | 32, 44.1, 48 | 16.0, 17.6, 17.6 |
| AAC | 1ch-Mono | 96 | 32, 44.1, 48 | 16.0, 22.1, 24.0 |
| AAC | 1ch-Mono | 112 | 32, 44.1, 48 | 16.0, 22.1, 24.0 |
| AAC | 1ch-Mono | 160 | 32, 44.1, 48 | 16.0, 22.1, 24.0 |
| AAC | 2ch-Stereo | 24 | 24 | 6.6 |
| AAC | 2ch-Stereo | 28 | 24 | 6.6 |
| AAC | 2ch-Stereo | 32 | 24 | 6.6 |
| AAC | 2ch-Stereo | 40 | 24, 32 | 8.5, 8.5 |
| AAC | 2ch-Stereo | 48 | 24, 32 | 8.5, 8.5 |
| AAC | 2ch-Stereo | 56 | 24, 32, 44.1, 48 | 12.0, 12.5, 12.5, 12.5 |
| AAC | 2ch-Stereo | 64 | 32, 44.1, 48 | 12.5, 12.5, 12.5 |
| AAC | 2ch-Stereo | 60 | 32, 44.1, 48 | 13.5, 13.5, 13.5 |
| AAC | 2ch-Stereo | 96 | 32, 44.1, 48 | 15.5, 15.5, 15.5 |
| AAC | 2ch-Stereo | 112 | 32, 44.1, 48 | 16.0, 16.0, 16.0 |
| AAC | 2ch-Stereo | 128 | 32, 44.1, 48 | 16.0, 16.0, 16.0 |
| AAC | 2ch-Stereo | 160 | 32, 44.1, 48 | 16.0, 17.6, 17.6 |
| AAC | 2ch-Stereo | 192 | 32, 44.1, 48 | 16.0, 22.1, 24.0 |
| AAC | 2ch-Stereo | 224 | 32, 44.1, 48 | 16.0, 22.1, 24.0 |
| AAC | 2ch-Stereo | 256 | 32, 44.1, 48 | 16.0, 22.1, 24.0 |
| AAC | 2ch-Stereo | 320 | 32, 44.1, 48 | 16.0, 22.1, 24.0 |

## CODEC COMPARISONS

### AUDIO FILES

This section allows you to assess the subjective audio quality of different permutations of several currently available lossy audio codecs. It is one of the most comprehensive comparisons available on the Internet.

We chose these particular audio sources because they tend to stress lossy codecs for various reasons, which will become apparent after you listen to some of these samples. The source audio was digitally extracted from the original label CDs. There were no physical cables or any analog-to-digital or digital-to-analog conversions in the extraction process. The source audio extraction used software error correction and is thus bit-accurate.

**Track 1** — There is a slight left and right channel delay caused by slight misalignment of the analog tape recorder used to master this recording. It should sound fine in stereo. This can be verified by listening to the mono sum or the mono PCM sample. Many poorly designed codecs will smear this in stereo.

**Track 2** — An abundance of high frequency energy will cause high frequency smearing in many codecs.

**Track 3** — Female voice is the most difficult test for codec voice performance. Many codecs will produce a tunnel effect on voice.

**Track 4** — Uncorrelated stereo material taxes the codec ability to take advantage of common material between left and right channels. This file is mainly useful in assessing stereo codecs.

A high-speed Internet connection will facilitate your evaluation. However, since the files are not streamed and must be downloaded, it is not required.

It is important to use the correct player application and version when you listen to these files. Many player features are changing to accommodate the new AAC/HE-AAC/aacPlus codecs in the various formats and containers. All files are in their native file format and need the correct player application to ensure that they will play properly with the correct reciprocal codec. We did this intentionally so that you can verify that these files are in the format they claim to be. Therefore, it will be necessary to install these players to perform an accurate evaluation. In order to play all the files here, you will need a minimum of Winamp 5.2, RealPlayer 10 and Windows Media Player 9. QuickTime 6 is optional, but useful to assess downward compatibility of HE AAC/aacPlus on an AAC only player, hopefully only for the time being. Apple, are you listening?

Installing multiple media players on a single computer can sometimes be problematic. However, through careful installation, it is possible to have these players coexist without conflict. You must make sure that file associations are properly assigned to their player applications and that no player application appropriates an incompatible file type.

Unfortunately most Microsoft Windows installations include Windows Media Player by default. WMP supports some non-Microsoft audio file formats, including some it should not, as they are either not correctly or not fully supported. This causes either poor quality or complete failure to play. Below is a chart of tested and recommended players and file formats for Microsoft Windows computer systems. Use this chart to make sure the correct player is associated with the correct file types.

## RECOMMENDED PLAYER FILE TYPE ASSOCIATIONS

FILE TYPE ASSOCIATIONS

| ***File Type*** | ***Winamp*** | ***RealPlayer*** | ***QuickTime*** | ***Windows Media Player*** |
|---|---|---|---|---|
| **.aac** | **X** | | | |
| **.mov** | | | **X** | |
| **.mp4** | **X** | **X** | **X** | |
| **.mp3** | **X** | **X** | **X** | **X** |
| **.mp2** | **X** | **X** | **X** | |
| **.ogg** | **X** | | | |
| **.rm** | | **X** | | |
| **.wma** | | | | **X** |
| **.m3u** | **X** | | | |
| **.pls** | **X** | | | |
| **.asx** | | | | **X** |
| **.ram** | | **X** | | |
| **.qtl** | | | **X** | |

## SUPPORTED PLAYOUT SYSTEM

## METADATA FOR STREAM AND CONTENT INFORMATION

OPTICODEC-PC supports one of the most important features of streaming media: sending real-time stream name, artist, and title. Many competing encoders cannot send this information at all or cannot send it in real-time. Some expensive servers these competing encoders work with do not even have this real-time metadata feature available.

Below is a list of currently supported playout systems for artist and title metadata and their methods. If your system is not listed, please contact Orban.

| PLAYOUT SYSTEMS | | Real-Time Data Injection Recommended | | Non Real-Time File Polling Not Recommended | |
|---|---|---|---|---|---|
| | **System** | **Ethernet** | **Serial** | **Text File** | **HTML File** |
| | BE AudioVault | X | X | | |
| | BSI Simian | X* | X | X | |
| | BSI Wavestation | X* | X | X | |
| | Dalet | X | X | X | |
| | Enco | X | X | X | |
| | Netia | | | | |
| | OtsDJ | | | | X |
| | Prophet Systems NexGen | X | X | | |
| | Prophet Systems NexGen 101 | X | X | | |
| | RCS Master Control | X | X | X | |
| | Winamp 2.x / 5.x | X* | X* | X | |
| | WinRadio | | | | |

* Extra software required.

# features + benefits

| GENERAL FEATURES |
|---|
| FREE software players include Adobe Flash, RealNetworks RealPlayer, Apple QuickTime, and Nullsoft Winamp, among others. |
| True streaming providing content security - not progressive downloads from a web server. |
| Hardware players include Apple iPhone/iPod Touch, Terratec Noxon2, Roku Soundbridge, Chumby. |
| Most efficient audio codec currently available. |
| Most natural sounding lossy audio codec currently available. |
| Provides higher audio quality for a given bitrate than other audio codecs. |
| Fewer audible coding artifacts for a given bitrate than other audio codecs. |
| Dramatically superior to MP3 and WMA. |
| AAC is the choice for the number 1 music download service: Apple iTunes. |
| HE-AACv1/v2 aka aacPlus is the choice for standards-based 3GPP mobile streaming. |
| Minimizes network bandwidth costs. |
| Maximizes network reliability minimizing buffering. |
| When used with OPTIMOD-PC, OPTICODEC-PC provides true entertainment-grade audio quality. |
| Accurate peak-hold audio level metering. |
| Robust network connectivity with auto-reconnect. |
| Packet optimization for increased efficiency. |
| Extensive, flexible metadata support using direct connect, Serial or Ethernet interfaces. |
| **PROFESSIONAL EDITION** |
| Microsoft Windows XP/Vista and 2003/2008 Server Application. |
| Graphical user interface uses standard Microsoft Windows menu structures for ease of learning and use. |
| Supports batch file execution to enable easy launching and automation. |
| Must be used with OPTIMOD-PC professional signal processing sound card. |
| Up to 128 kbps bitrate using HE-AAC / aacPlus codec. |
| Up to 320 kbps bitrate using AAC codec. |
| Unlimited number of encoder instances per stream. |
| MPEG-2 / MPEG-4 / 3GPP / RTMP Enterprise Class Streaming. |

## reduces

> network streaming bandwidth requirements and costs

> storage requirements for personal audio players and mobile devices

## increases

> audio quality for a given bitrate by comparison to free codecs

## PE Version – Professional Edition

PROFESSIONAL EDITION

| | |
|---|---|
| ***Compatible Players*** | |
| **RTMP Flash** | **Adobe Flash® Browser Plugin** AAC/HE-AACv1/HE-AACv2. |
| | **Tuner2 HiFi iPhone/iPod Touch** AAC/HE-AACv1/HE-AACv2. |
| | **Orban/Coding Technologies AAC/HE-AAC Windows Media Player Plugin** AAC/HE-AACv1/HE-AACv2. |
| **RTSP/RTP** | **RealNetworks® RealPlayer® 10** and above AAC/HE-AACv1 / aacPlus v1 Audio Streams. |
| | **Apple Computer® QuickTime® Player 6.0** and above AAC Audio Streams; Currently does not support HE-AAC/aacPlus Will currently play AAC portion of HE-AAC/aacPlus streams. |
| | **Orban/Coding Technologies AAC/HE-AAC Windows Media Player Plugin** AAC/HE-AACv1/HE-AACv2. |
| **HTTP/ICY** | **Nullsoft Winamp® 5.05** and above**.** AAC/aacPlus. Will currently play AAC portion of HE AAC/ aacPlus streams). |
| | **Orban/Coding Technologies AAC/HE-AAC Windows Media Player Plugin** AAC/HE-AACv1/HE-AACv2. |
| ***Computer*** | |
| **Minimum System Requirements** | **Microsoft Windows® XP** Intel Pentium II 400 MHz, RAM 128 MB, 256 MB recommended. |
| | **Microsoft Windows® Server 2003** Intel Pentium III 500 MHz, RAM 128 MB, 256 MB recommended. |
| | This specification denotes the minimum CPU power necessary to control one OPTIMOD-PC card with external audio sources and one instance of the OPTICODEC-PC encoder. Additional cards, audio player and/or encoder software will require additional CPU power. |
| **Processor and Chipset** | This software has been tested and qualified with Intel CPU and chipsets. |
| **Sound Device** | An Orban OPTIMOD-PC 1100/1101 audio processor / sound card must be installed in the host computer in order to run the OPTICODEC-PC PE application. |
| **Interface** | Graphical User Interface (GUI), standard Microsoft Windows. May be executed from command-line using configuration file for each stream. |
| ***Encoder*** | |
| **Codec Technology** | MPEG-2 AAC ADTS, ISO/IEC 13818-7 http://www.iis.fraunhofer.de/amm/techinf/aac/index.html |
| | MPEG-4 AAC, ISO/IEC 14496-3 http://www.iis.fraunhofer.de/amm/techinf/aac/index.html |
| | MPEG-4 HE-AAC / aacPlus v1 / aacPlus v2 ISO/IEC 14496-3 : Amd-1 : Amd-2 http://www.codingtechnologies.com |
| **Sample Rates (kHz)** | 24, 32, 44.1, 48. |
| **Bit Rates (kbps)** | 8, 10, 12, 16, 20, 24, 32, 40, 48, 56, 64, 80, 96, 128, 160, 192, 224, 256, 320. |
| **Number of Channels** | 1-Mono 2-Stereo. |
| **Coding Options** | Optimize for Voice. |
| **Encoder Instances per Computer – Multicast** | One. |
| **Encoder Instances per Computer – Unicast** | Limited only by available CPU power. |
| ***Streaming*** | |
| **Transport Protocols** | RTSP/RTP: RFC 2326/3550 - HTTP/ICY SHOUTcast: RFC 2616 - HTTP/ICY Icecast2: RFC 2616 - RTMP Flash: Adobe Protocol. |
| ***Methods:*** | |
| **Multicast RTP/UDP (Internal Server)** | Transmission: Multicast TTL: 1-255. |
| **Unicast RTSP/RTP/UDP (Internal Server)** | Transmission: Unicast. |
| **Unicast RTP/TCP (External Server)** | Transmission: Manual Unicast |
| **Unicast RTSP/RTP/TCP (External Server)** | Transmission: Automatic Unicast - Announce Session Description Protocol (.sdp) file per stream generated and sent to server: RFC 2327. |
| **RTP Payload Format** | ISMA (audio/mpeg4-generic): RFC 3640 3GPP/3GPP2 (audio/MP4A-LATM): RFC 3016. |
| **Unicast HTTP/ICY/TCP** | Content Type: audio/aac Content Type: audio/aacp. |
| **Packet Size** | 1450 Bytes plus IP Header Bytes = Total < 1500 Byte MTU. |
| **Packet Optimization (PE only)** | Multiple audio frames per packet without fragmentation. Packet size not to exceed specified MTU. |

specifications

PROFESSIONAL EDITION

PROFESSIONAL EDITION

| | |
|---|---|
| ***Streaming** (continued)* | |
| **Connection Fallback - Unicast** | Automatic Reconnection to External Server upon Connection Failure. |
| **Stream Information** | Stream Name and Description. |
| | Artist and Title. |
| | URL. |
| | All server-supported metadata. |
| **Metadata Input** | Text File. |
| | Ethernet TCP/UDP. |
| | Manual Entry. |
| | Several Audio Playout Systems Supported Directly. |
| **Server Requirements/Platform – RTSP/RTP Unicast** | **Free Darwin Streaming Server 5.0** or later<br>http://developer.apple.com/ darwin/projects/streaming/<br>Available for Microsoft Windows 2000 Professional/Server, Windows XP, Windows 2003 Server, Apple Mac OS X 10.2.8 or later Server and Proxy, Red Hat Linux 9, FreeBSD, Sun Solaris 9<br>Helix Server 10.0 or later<br>http://www.realnetworks.com/ products/media_delivery.html<br>Available for Microsoft Windows 2003 Server, Linux (RHEL 4.0), Sun Solaris 8 (Native) and 9. |
| | **Helix Mobile Server**<br>http://www.realnetworks.com/ industries/serviceproviders/ mobile/products/server/index. html<br>Available for Microsoft Windows 2003 Server, Linux (RHEL 4.0), Sun Solaris 8 (Native) and 9. |
| **Server Requirements/Platform – HTTP/ICY SHOUTcast** | **Free Nullsoft SHOUTcast DNAS 1.9.4** or later<br>http://www.shoutcast.com/ download/serve.phtml<br>Available for Microsoft Windows 2000 Professional/Server, Windows XP, Windows 2003 Server, Red Hat Linux 9, FreeBSD, Sun Solaris 9. |
| **Server Requirements/Platform – HTTP/ICY Icecast2** | **Free Icecast2 Server 2.2.0** or later<br>http://www.icecast.org/ download.php<br>Available for Microsoft Windows 2000 Professional/Server, Windows XP, Windows 2003 Server, Red Hat Linux 9, FreeBSD, Sun Solaris 9. |
| **Server Requirements/Platform – RTMP** | **Adobe Flash Media Server**<br>Available for Microsoft Windows XP, Windows 2003/2008 Server, Linux. |
| | **Wowza Media Server Pro**<br>http://www.wowzamedia.com<br>Available for Microsoft Windows XP, Windows 2003/2008 Server, Linux, Mac OS. |
| ***Metering / Status Indication*** | |
| **Graphical Interface** | Audio Level, Peak Indicating. |
| | Connected to Remote Server. |
| | Disconnected from Remote Server. |
| | Bitrate of Connection to Remote Server. |
| | Connection Elapsed Time - DDD:HH:MM:SS. |
| ***Log File*** | |
| **Functions** | Enable/Disable, ASCII Text, Log File Size Specifier. |
| **Content** | Connect Date/Time, Connect Remote Server, Connect Bitrate, Disconnect Time. |

*Because engineering improvements are ongoing, specifications are subject to change without notice.*

## SUPPORTED PLAYERS & SERVERS

www.orban.com

r13-2010_GO/Lo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