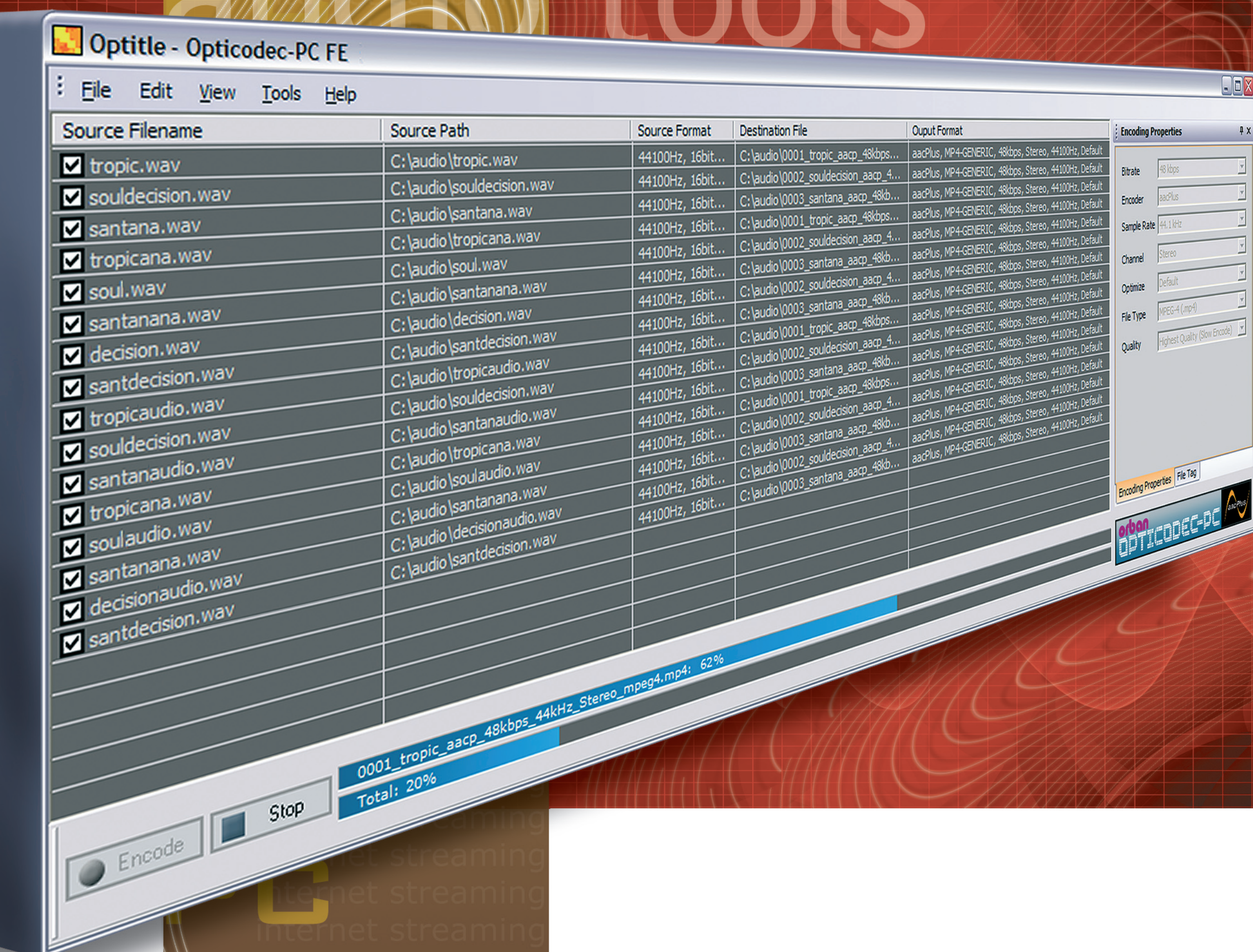
streaming

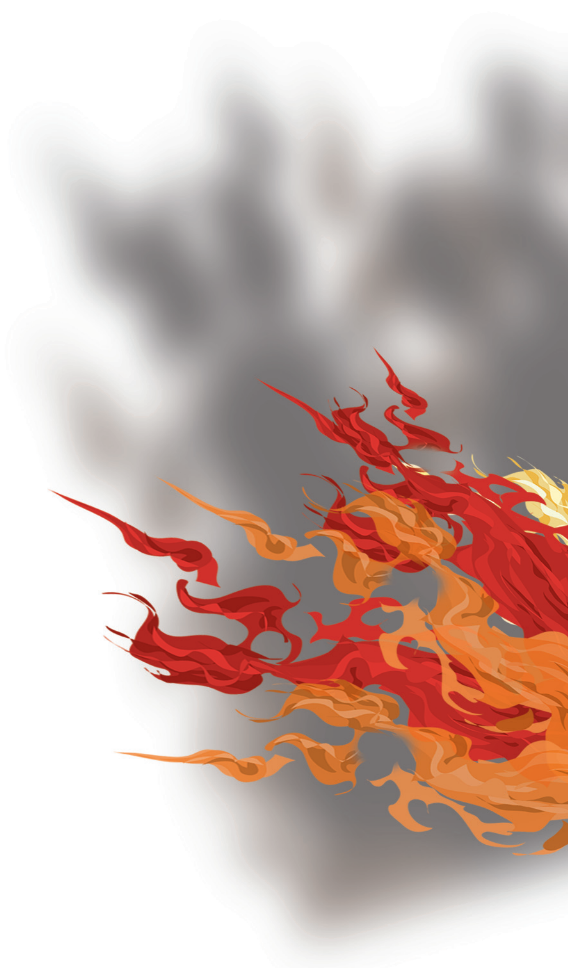
audio tools



OPTICODEC-PC 1020 FILE ENCODER

Low bitrate Internet audio
is red hot, and so is our
OPTICODEC-PC 1020 File Encoder.
Just remember, though —

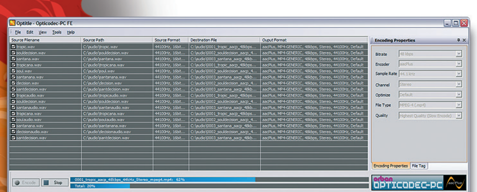
fire extinguisher
not included.



OPTICODEC-PC FE is professional quality MPEG-2/MPEG-4 AAC/aacPlus™ software for highly optimized, fast audio file encoding. OPTICODEC-PC FE offers the most important feature that the basic netcaster is looking for in a file-encoding product — entertainment-quality sound at economical file sizes.

OPTICODEC-PC FE software lets you supply files encoded with the Coding Technologies® AAC / HE AAC / aacPlus codec, widely acknowledged as offering the highest available audio quality at the lowest possible bitrate. AAC/aacPlus™ is changing the way that low bitrate audio is perceived. For a given bitrate, it sonically outperforms any other codec currently available.

Introducing OPTICODEC-PC 1020 File Encoder: Entertainment-quality sound at economical file sizes



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 streams to FM and other codecs.

Files encoded with OPTICODEC-PC can be experienced through RealPlayer® 10, QuickTime 6.5, or Winamp 5.08 and higher as well as Apple iTunes and iPods® and all other AAC/aacPlus-enabled 3GPP cellular devices. OPTICODEC-PC FE is ideal for podcasters and can even encode ringtones, which are now an important new source of revenue for production houses.

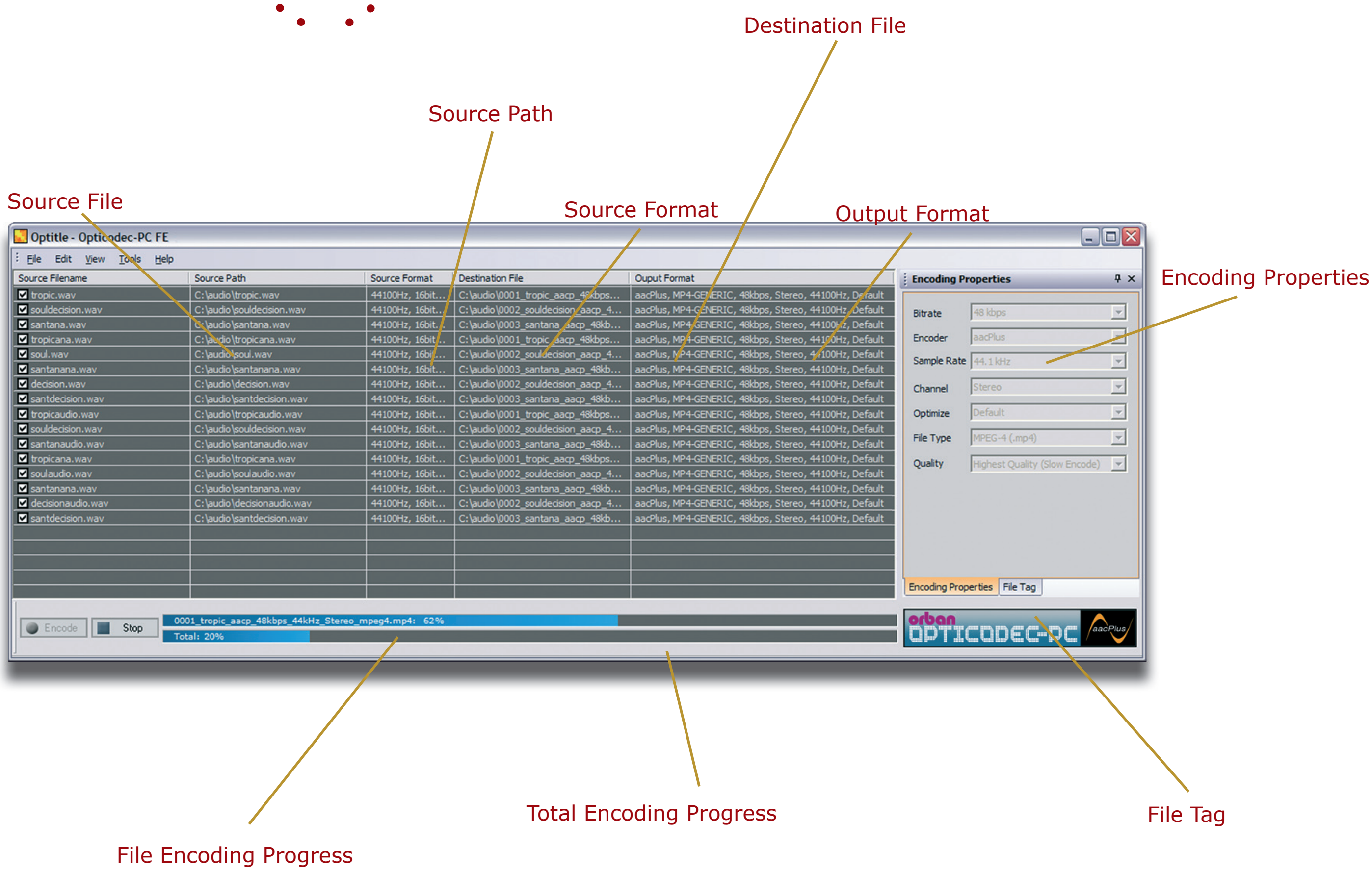
OPTICODEC-PC FE is available for Microsoft Windows® XP/2003 Server. It is available in three versions, ranging from the Personal edition for light users all the way to the Enterprise edition, which is an industrial-strength batchable tool for users with heavy production requirements.

NOW SUPPORTS:

MPEG-2 ADTS
MPEG-4 ISMA
MPEG-4 3GPP

3 Versions to Choose From:

Personal
Professional
Enterprise



Uncompromised MPEG-4 and 3GPP support means that whatever the target device or server, OPTICODEC-PC file encoder will create a correctly hinted file that plays right every time, while tagging options ensure that your target audience will see the title and artist on their PCs or mobile cellular devices.

Orban's OPTIMOD-PC audio processor on a PC card is an excellent companion for OPTICODEC-PC FE. 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 1100, a professional PCI sound card designed for streaming media, provides "genuine radio"™ audio processing for Internet broadcasters. With three on-board DSP's providing equalization, AGC, multi-band compression, and look-ahead limiting, OPTIMOD-PC 1100, especially when combined with aacPlus encoding technology, delivers a polished and produced stream that has the same loudness, consistency, and punch as satellite and major-market FM radio.

OPTIMOD-PC ordinarily processes the output 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 is also a capable mixer, having one stereo analog input, two AES3 / SPDIF digital inputs, and one wave input, all of which can be mixed. Thanks to onboard samplerate converters, the two digital inputs can accept and mix asynchronous sources. In practice, the four inputs might be used for a local feed, a network feed, a voice channel, and a wave player, making OPTIMOD-PC the heart of a "desktop production studio". Thanks to separate "processed" and "unprocessed" mixers, any of the inputs in any combination can be processed or passed directly to the input of OPTICODEC-PC without processing — the user can always choose how much processing (if any) to apply to the audio.

high audio quality

low bitrate

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. Low bitrate Internet audio is red hot, and OPTICODEC-PC is the fuel that puts you ahead of the pack.

details

Supported Players

WINAMP · realPlayer · QuickTime · Get VLC! · foobar2000 · CoreCodec.org · iPod · iTunes

Supported Servers

SHOUTcast · icecast.org · Steamcast Server · Darwin

features + benefits = cost savings

ALL VERSIONS
FREE players available now from RealNetworks, RealPlayer, Apple Computer, QuickTime and Nullsoft Winamp.
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.
Keeps network bandwidth and storage costs down.
When used with OPTIMOD-PC, OPTICODEC-PC provides true entertainment-grade audio quality.
Available in three versions, Personal, Professional, and Enterprise.
Microsoft Windows XP/2003 Server Application.
Graphical user interface uses standard Microsoft Windows menu structures for ease of learning and use.
Supports any high quality Microsoft Windows Sound Card. May be used with OPTIMOD-PC professional signal processing sound card.
Up to 128 kbps bitrate using aacPlus codec.
Up to 320 kbps bitrate using AAC codec.
To ensure best quality input, uses linear PCM .wav file or direct capture from sound device.
Creates MPEG-2/MPEG-4 compliant AAC/aacPlus Files.
Apple iTunes/iPod AAC/aacPlus portable player compatible.
Winamp 5.08 compatible.
File asset tagging.
PROFESSIONAL VERSION
Controllable from GUI or command line. Batchable.
Creates MPEG-4 hinted AAC/aacPlus files for streaming servers.
ENTERPRISE VERSION
Controllable from GUI or command line. Batchable.
Creates MPEG-4 hinted AAC/aacPlus files for streaming servers.
Creates 3GPP compliant AAC/aacPlus files for mobile devices.
Creates 3GPP hinted AAC/aacPlus files for mobile streaming servers for on-demand delivery.
3GPP mobile device compatible.

reduces

> network streaming bandwidth requirements and costs

> storage requirements for personal audio players and mobile devices

increases

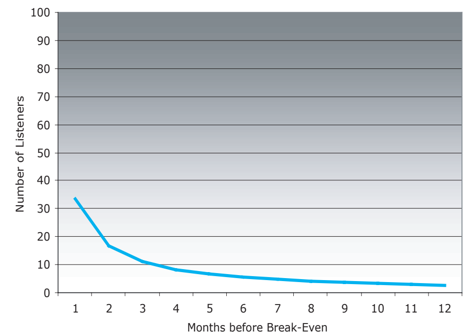
> audio quality for a given bitrate by comparison to free codecs

features + benefits = cost savings

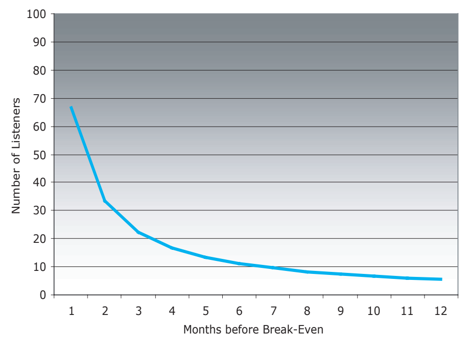
The top three graphs show, as a function of the number of clients, how long it will take before the cumulative cost of file serving at 128 kbps with no encoder license fee becomes higher than the cumulative cost of file serving at 32 kbps plus the purchase price of a given version of OPTICODEC-PC FE. After the break-even time indicated on these graphs, savings at 32 kbps continue to accrue. We assume continuous serving at the stated bitrate.

The middle three graphs show the cost of continuously serving a file per hour or month, broken down as a function of bitrate. Cost is directly proportional to bitrate, so low bitrates are better if the associated quality satisfies your audience.

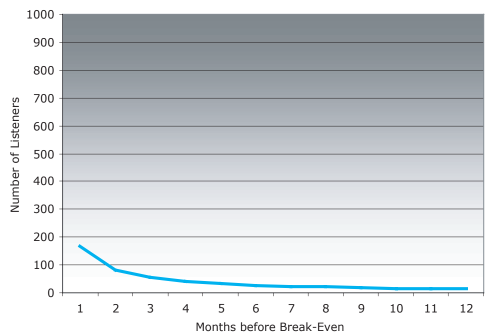
OPTICODEC-PC FE Personal
Cost of Ownership and Operation
Comparitive Break-Even Analysis
32 kbps Compared to 128 kbps
(Based on US$4.00 / Month / 128 kbps)



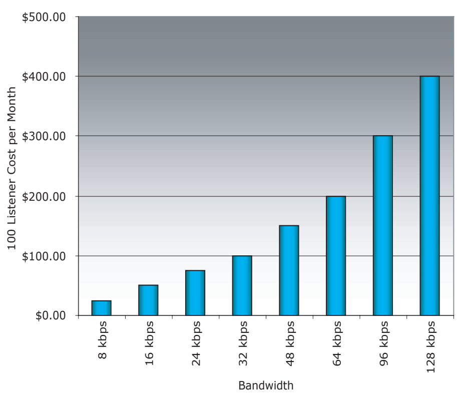
OPTICODEC-PC FE Professional
Cost of Ownership and Operation
Comparitive Break-Even Analysis
32 kbps Compared to 128 kbps
(Based on US$4.00 / Month / 128kbps)



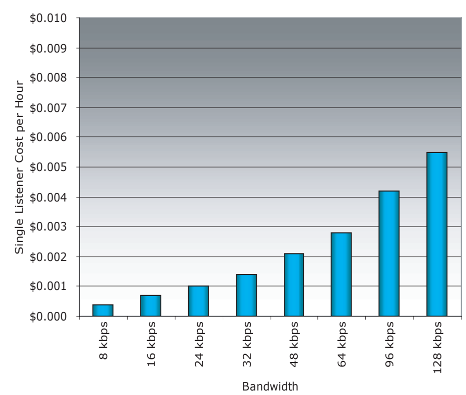
OPTICODEC-PC FE Enterprise
Cost of Ownership and Operation
Comparitive Break-Even Analysis
32 kbps Compared to 128 kbps
(Based on US$4.00 / Month / 128 kbps)



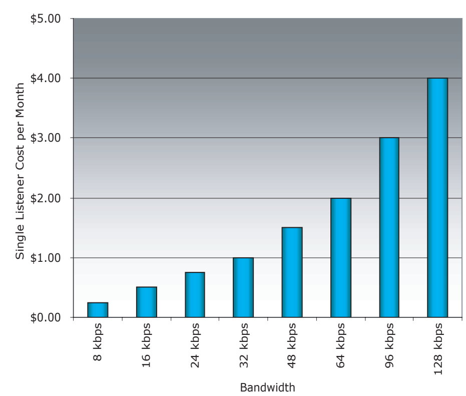
Streaming Bandwidth 100 Listener Cost per Month
(Based on US$4.00 / Month / 128 kbps)



Streaming Bandwidth Single Listener Cost per Hour
(Based on US$4.00 / Month / 128 kbps)

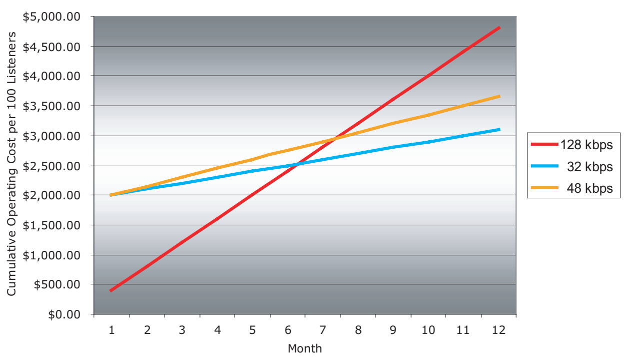


Streaming Bandwidth Single Listener Cost per Month
(Based on US$4.00 / Month / 128 kbps)



The bottom graph shows how operating costs accrue over time if you purchase OPTICODEC-PC FE Enterprise Edition plus OPTIMOD-PC and serve at 32 or 48 kbps, compared to serving 128 kbps with a free encoder. After a short time, the bandwidth savings alone will more than pay for the OPTICODEC/OPTIMOD combo, giving you a very quick return on investment. Coupled with the fact that the sound quality will attract and hold listeners, this means that serious netcasters cannot afford to be without OPTICODEC-PC FE and that free codecs at higher bitrates actually end up costing more in the long term.

OPTICODEC-PC FE / Streaming Bandwidt
Operating Cost Comparison - Cumulative
(Based on US$4.00 / Month / 128 kbps)
100 Listeners : 32/48 kbps : 128 kbps



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.

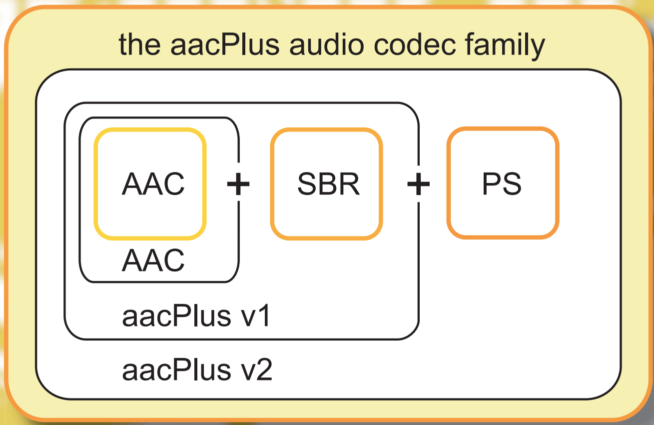
SBR and PS are forward and backward compatible methods to enhance the efficiency of any audio codec. AAC was chosen as the core codec for aacPlus because of its superior performance over older generation audio codecs such as MP3 or WMA. This was the reason why Apple Computer chose AAC for their market-dominating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

aacPlusV2codec

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

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+)

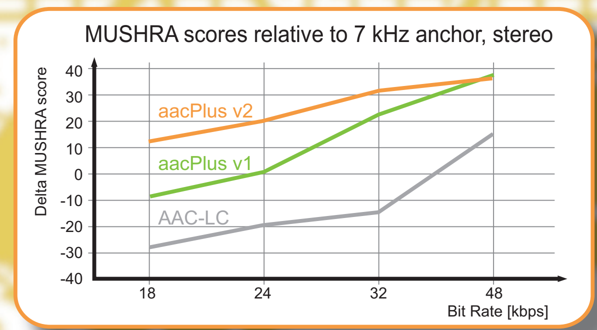
MPEG ISO/IEC 14496-3:2001/AMD-2: Parametric Coding for High Quality Audio

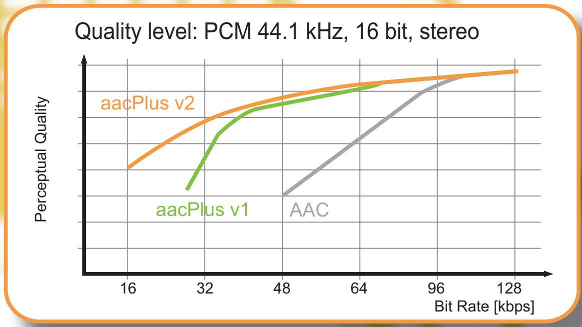
aacPlus v1 is standardized by 3GPP2 (3rd Generation Partnership Project 2), ISMA (Internet Streaming Media Alliance), DVB (Digital Video Broadcasting), the DVD Forum, Digital Radio Mondiale, and many others. aacPlus v2 is specified as the high quality audio codec in 3GPP (3rd Generation Partnership Project)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

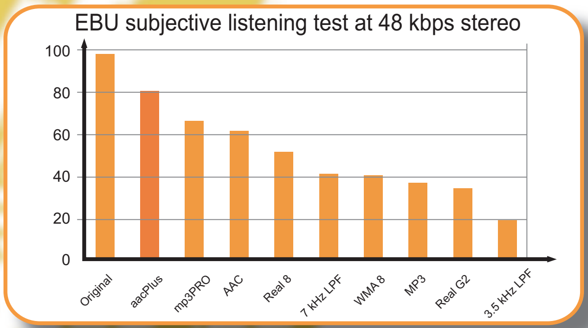
INDEPENDENT QUALITY EVALUATIONS OF aacPLUS

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.



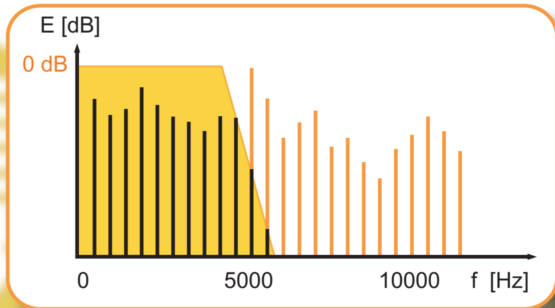
SPECTRAL BAND REPLICATION

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.

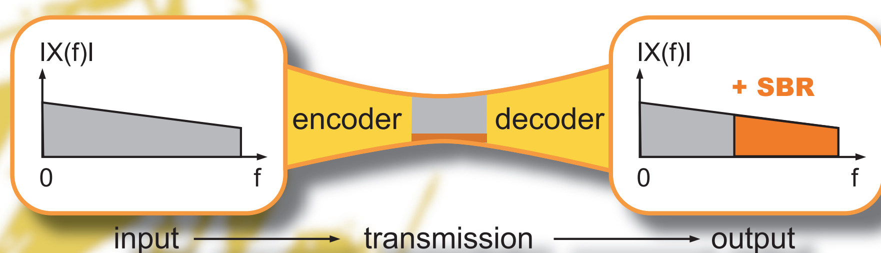
Using perceptual codecs at low bitrates, however, is not without its downside. State-of-the-art perceptual audio codecs such as AAC, achieve "CD-quality" or "transparent" audio quality at a bitrate of approximately 128 kbps (~ 12:1 compression). Below 128 kbps, the perceived audio quality of most of 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.



example of band-limiting of a typical signal

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 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.

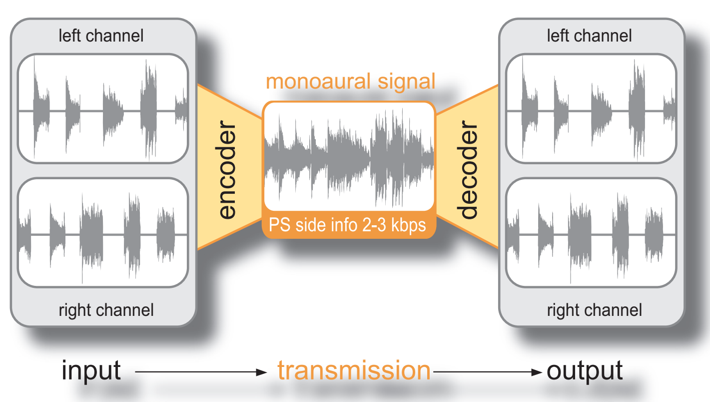
SBR can enhance the efficiency of perceptual audio codecs by ~ 30% (even more in certain configurations) in the medium to low bitrate range. The exact amount of improvement that SBR can offer also depends on the underlying codec. For instance, combining SBR with AAC achieves a 64 kbps stereo stream whose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



FE Version – File Edition

Personal / Professional / Enterprise Versions

PERFORMANCE	***Compatible Players***	
	MPEG-4 ISMA	**RTSP/RTP Streaming**
		RealNetworks® RealPlayer® 10 and above: AAC/HE-AAC/aacPlus
		Apple Computer® QuickTime® Player 6.0 and above
		Apple Computer® iTunes/iPod: AAC Audio Streams; Currently does not support HE-AAC/aacPlus; Will play AAC portion of HE-AAC/aacPlus streams
		VLC
	MPEG-2 ADTS	**RealNetworks® RealPlayer® 10 and above:** AAC/HE-AAC/aacPlus
		HTTP/ICY Streaming, Nullsoft Winamp 5.05 and above, Foobar2000 and VLC
	3GPP Mobile	Several Mobile PDAs and Cellular Telephones
INSTALLATION	***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 optional Windows sound device is required for audio capture. An Orban OPTIMOD-PC 1100 audio processor / sound card is recommended for processing files for on-demand services.
	Interface	Graphical User Interface (GUI), standard Microsoft Windows. Command Line Interface (CLI), supports batch file execution for easy launching and automation capability.
	Encoder	
	Codec Technology	MPEG-2 AAC ADTS, ISO/IEC 13818-7 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 and 2-Stereo
	Coding Options	Optimize for Voice
	Encoder Instances per Computer	1
	Speed	1 Minute Stereo PCM file Encodes to: 32 kbps / 44.1 kHz, 48 kbps / 44.1 kHz, and 64 kbps / 44.1 kHz
INSTALLATION	***Input***	
	Input Audio Source	Microsoft Windows or Virtual Sound Device required for direct audio capture encoding.
	Input File Formats	Linear PCM [.wav] and Linear PCM Broadcast Wave Format [.bwf]
	Output File Formats	
	MPEG-2 : ISO/IEC 13818-7	ADTS [.aac] andADTS - ID3v2 Tag [.aac]
	MPEG-4 : ISO/IEC 14496-1/3/12/14	MPEG-4 [.mp4], MPEG-4 Hinted ISMA [RFC-3640] [.mp4], MPEG-4 QT Tag [iPod/iTunes] [.m4a], and MPEG-4 QT Tag Hinted ISMA [RFC-3640] [iPod/iTunes] [.m4a]
	3GPP : ISO/IEC 14496-1/3/12/14 *Enterprise Version Only*	3GPP [.3gp], 3GPP 3GP Tag [.3gp], 3GPP Hinted LATM [RFC-3016] [.3gp], and 3GPP 3GP Tag Hinted LATM [RFC-3016] [.3gp]
	Asset Tagging	Title and Copyright Information
		ID3v2 Version 3
		MPEG-4
		3GPP
	Accepts XML formatted Input	
INSTALLATION	***Metering / Status Indication***	
	Graphical Interface	File Processing – Individual – Progress Bar
		File Processing – Total – Progress Bar
		Audio Capture Level, True Accurate Peak Indicating
	Console Interface	File Processing Status
	Log File	
	Functions	Enable/Disable, ASCII Text and Log File Size Specifier
	Content	Encoding and Error Reporting
INSTALLATION	***Compatible Servers for Streaming Applications***	
	Server Requirements – RTSP/RTP Unicast	Free Darwin Streaming Server 5.0 or later http://developer.apple.com/darwin/projects/streaming/
	Server Platform – RTSP/RTP Unicast	http://developer.apple.com/darwin/projects/streaming/ 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.
	Server Requirements – HTTP/ICY SHOUTcast	Free Nullsoft SHOUTcast DNAS 1.9.4 or later http://www.shoutcast.com/download/serve.phtml
	Server Platform – HTTP/ICY SHOUTcast	http://www.shoutcast.com/download/serve.phtml Available for Microsoft Windows 2000 Professional/Server, Windows XP, Windows 2003 Server, Red Hat Linux 9, FreeBSD, Sun Solaris 9.
	Server Requirements – HTTP/ICY Icecast2	Free Icecast2 Server 2.2.0 or later http://www.icecast.org/download.php
	Server Platform – HTTP/ICY Icecast2	http://www.icecast.org/download.php Available for Microsoft Windows 2000 Professional/Server, Windows XP, Windows 2003 Server, Red Hat Linux 9, FreeBSD, Sun Solaris 9.

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

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