A HISTORY OF FLAC

THE FREE LOSSLESS AUDIO CODEC

JOSH COALSON

ADC25 Bristol

ac

s audio codec


SONY
POWER
TIMER PLAY
OFF
ON
LEVEL
DISC
DISC SET
TRACK NO.
MINUTE
SECOND
RECEIVED
REMOTE
SENSOR
RESET
REPEAT
1
ALL
MEMORY
CLEAR
LAP/REMAINING
TIME
PLAY
PAUSE
BACKWARD
AUTOMATIC MUSIC SENSOR
FORWARD
OPEN/CLOSE
COMPACT
disc
DIGITAL AUDIO


SONY
POWER
RESET
REPEAT
ALL
CLEAR
TIME
OPEN/CLOSE
COMPACT
disc
DIGITAL AUDIO


mp3

flac

free lossless audio codec

mp3

mp3


HI-QUALITY DIGITAL SOUND
VOLUME
HOLD
OPEN
mpman


DIGITAL
DIAMOND
VOL
RANDOM
REPEAT
A↔B
Rio PMP300

free lossless audio codec

Year	HDD Name / Model	Capacity
1994–1997	Various Desktop HDDs	Up to ~16 GB
1999	IBM Microdrive (1″ form factor)	170 MB & 340 MB
2004–2006	Mainstream Desktops	~200–300 GB

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S/N	Year	Company	Model	Form Factor	Capacity	Cost (US$)	Price/GB (US$)*
11	1995	Conner	CP1275 (IDE)	3.5″	1.3 GB	278	214
12	2000	Seagate	Elite 47GB (SCSI)	5.25″	47 GB	695	14.8
13	2005	Seagate	400GB 7200.8 (ATA-150)	3.5″	400 GB	249	0.623

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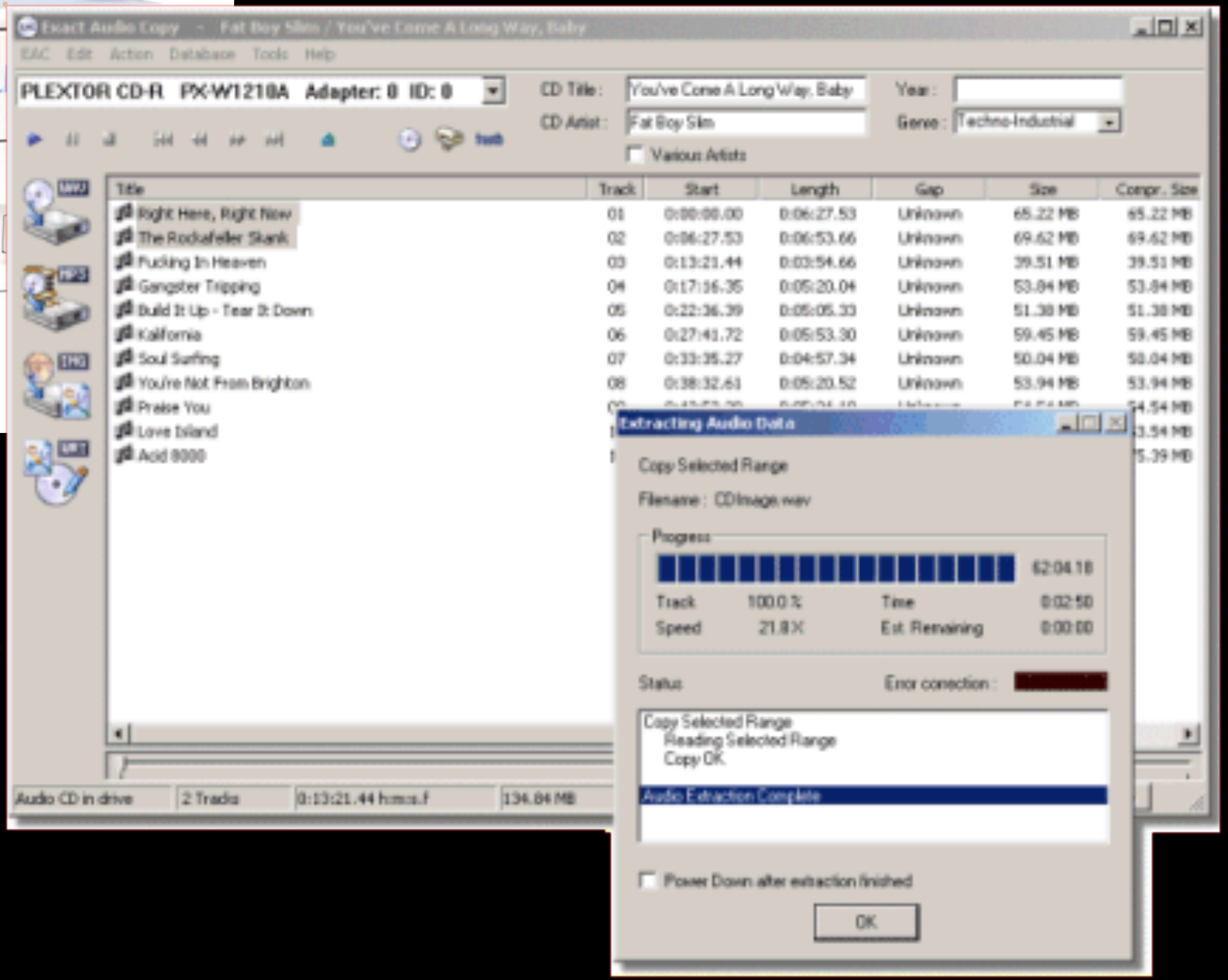
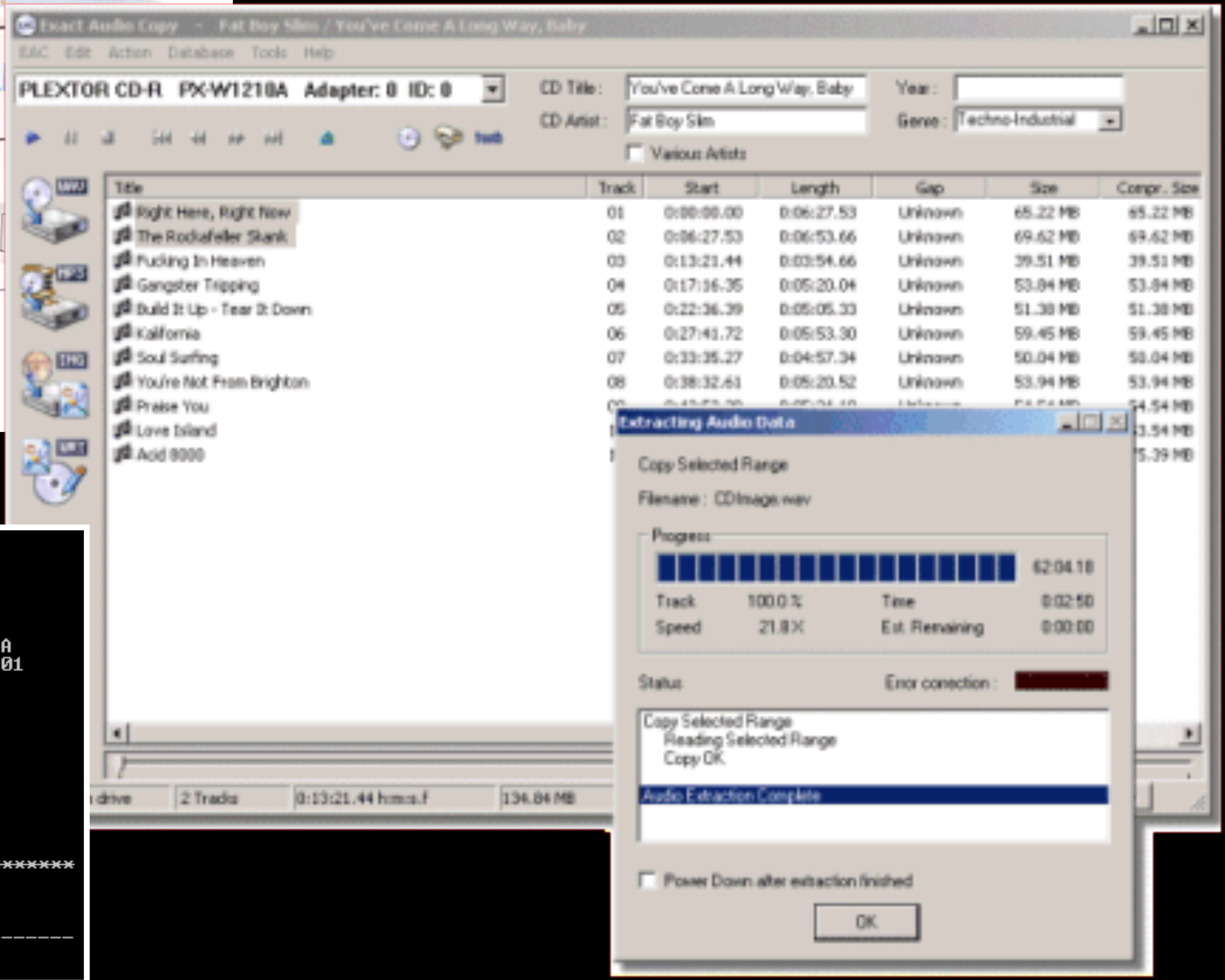
READ
WRITE


Exact Audio Copy - Fat Boy Slim / You've Come A Long Way, Baby
EAC Edit Action Database Tools Help
PLEXTOR CD-R PX-W1210A Adapter: 0 ID: 0
CD Title: You've Come A Long Way, Baby
CD Artist: Fat Boy Slim
Year:
Genre: Techno-Industrial
Various Artists
Title
Track
Start
Length
Gap
Size
Compr. Size
Right Here, Right Now 01 0:00:00.00 0:06:27.53 Unknown 65.22 MB 65.22 MB
The Rockafeller Skank 02 0:06:27.53 0:06:53.66 Unknown 69.62 MB 69.62 MB
Fucking In Heaven 03 0:13:21.44 0:03:54.66 Unknown 39.51 MB 39.51 MB
Gangster Tripping 04 0:17:16.35 0:05:20.04 Unknown 53.84 MB 53.84 MB
Build It Up - Tear It Down 05 0:22:36.39 0:05:05.33 Unknown 51.38 MB 51.38 MB
Kalifornia 06 0:27:41.72 0:05:53.30 Unknown 59.45 MB 59.45 MB
Soul Surfing 07 0:33:35.27 0:04:57.34 Unknown 50.04 MB 50.04 MB
You're Not From Brighton 08 0:38:32.61 0:05:20.52 Unknown 53.94 MB 53.94 MB
Praise You
Love Island
Acid 8000
Audio CD in drive
2 Tracks
0:13:21.44 h:m:s.f
134.84 MB
Extracting Audio Data
Copy Selected Range
Filename : CDImage.wav
Progress
62:04.18
Track 100.0 %
Time 0:02:50
Speed 21.8 X
Est. Remaining 0:00:00
Status
Error correction :
Copy Selected Range
Reading Selected Range
Copy OK
Audio Extraction Complete
Power Down after extraction finished
OK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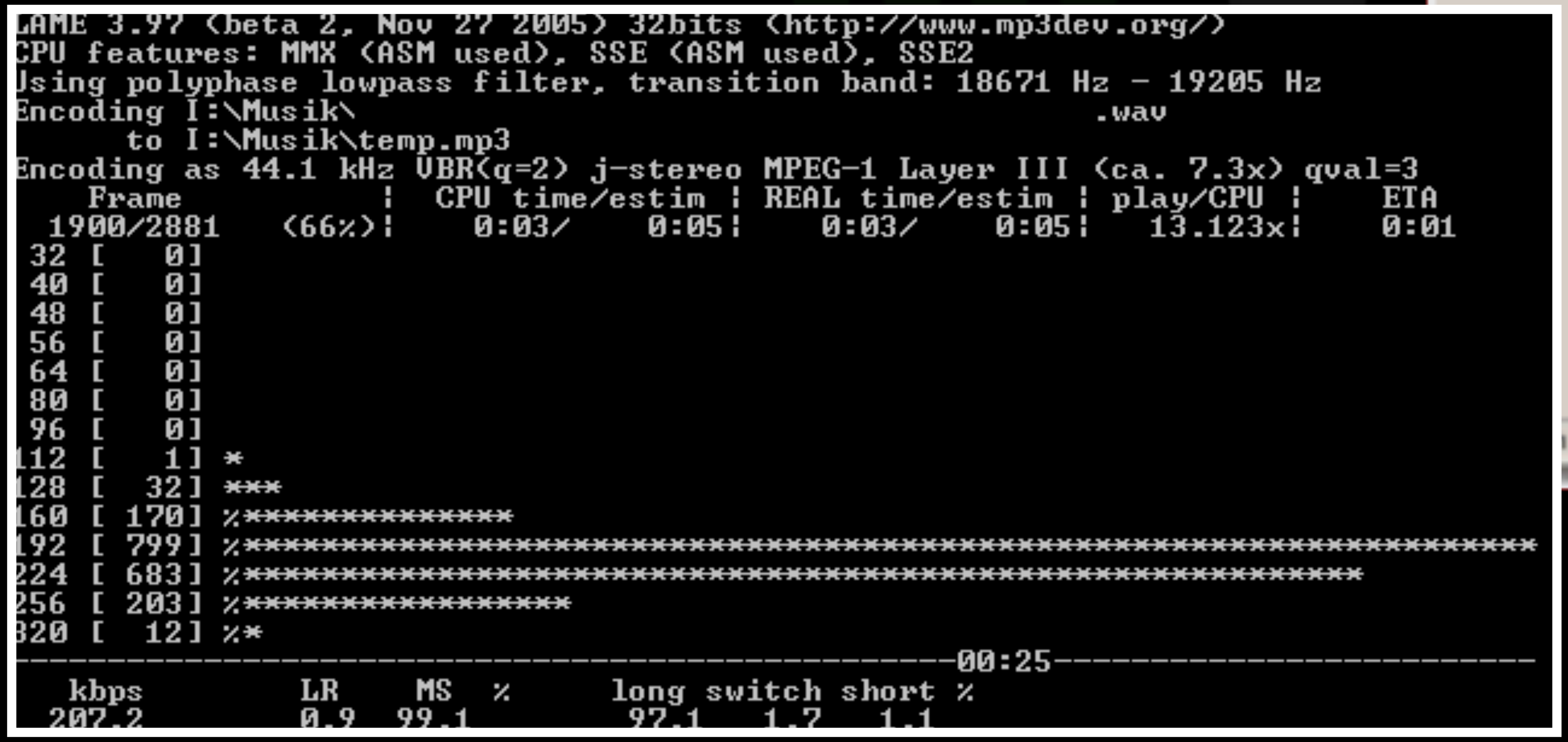
LAME 3.97 (beta 2, Nov 27 2005) 32bits (http://www.mp3dev.org/)
CPU features: MMX (ASM used), SSE (ASM used), SSE2
Using polyphase lowpass filter, transition band: 18671 Hz - 19205 Hz
Encoding I:\Musik\ .wav
to I:\Musik\temp.mp3
Encoding as 44.1 kHz VBR(q=2) j-stereo MPEG-1 Layer III (ca. 7.3x) qval=3
Frame | CPU time/estim | REAL time/estim | play/CPU | ETA
1900/2881 (66%)| 0:03/ 0:05| 0:03/ 0:05| 13.123x| 0:01
32 [0]
40 [0]
48 [0]
56 [0]
64 [0]
80 [0]
96 [0]
112 [1] *
128 [32] ***
160 [170] %**************
192 [799] %***
224 [683] %***
256 [203] %*****************
320 [12] %*
00:25
kbps LR MS % long switch short %
207.2 0.9 99.1 97.1 1.7 1.1


PL
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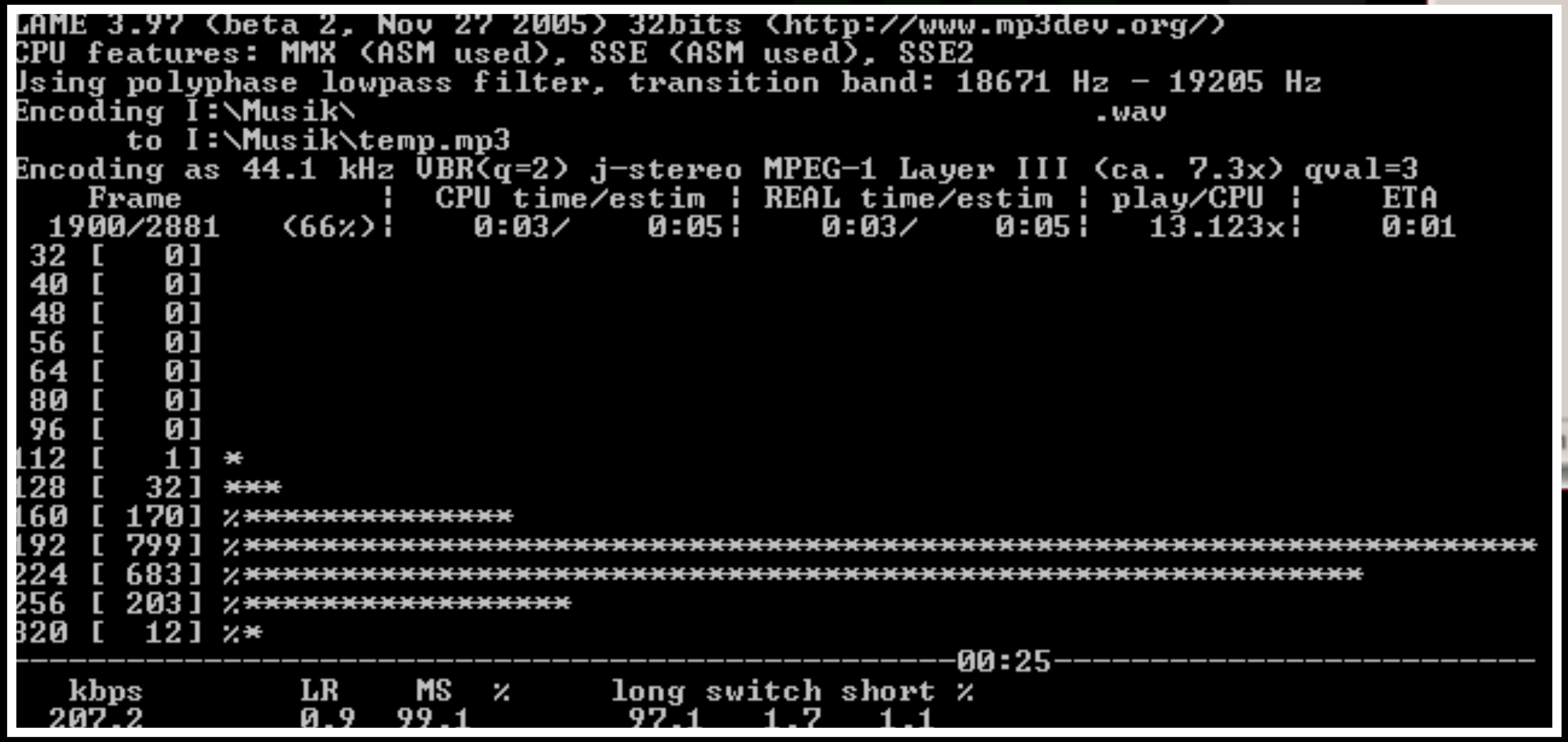
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207.2 0.9 99.1 97.1 1.7 1.1

MONKEY'S AUDIO
A fast and powerful lossless audio compressor

+ Good compression
- Closed source, Windows only
- Player support
- Novel algorithms

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WavPack

+ Fast; decent compression
- Closed source (at the time)
- Lossy mode

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- MUSICompress (1996)
- AudioPaK (1998)
- audiozip (1998)
- MG3 (1999)
- Kexis (2000)
- LTAC,LPAC (~2000)
- RKAU (2000)
- Wavezip (?)

WavPack

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codec

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Range encoding: an algorithm for removing redundancy from a digitised message.

G. N. N. Martin — IBM UK Scientific Center

Presented in March 1979 to the Video & Data Recording Conference, held in Southampton July 24-27 1979.

Redundancy in a message can be thought of as consisting of contextual redundancy and alphabetic redundancy. The first is illustrated by the fact that the letter Q is nearly always followed by the letter U, the second by the fact that the letter E is far more common than the letter X. Range encoding is an algorithm for removing both sorts of redundancy.

Since Huffman [1] published his paper in 1952 there has been a number of papers, e.g. [2], describing techniques for removing alphabetical redundancy, mostly generating prefix codes, and mostly transforming the messages into a bit string. The usual aim of such techniques is to reduce the quantity of storage required to hold a message.

In the last fifteen years the growth of telemetry has increased interest in techniques for removing contextual redundancy. Many of these techniques approximate the message, rather than simply remove redundancy. Such techniques are often analog, and include transmitting the difference

LINEAR PREDICTION FROM SUBBANDS FOR LOSSLESS AUDIO COMPRESSION

Ciprian Doru Giurcăneanu , Ioan Tăbuş and Jaakko Astola

Signal Processing Laboratory, Tampere University of Technology
P.O. Box 553, SF-33101 Tampere, Finland
e-mail: {cipriand,tabus,jta}@cs.tut.fi

ABSTRACT

In this paper we propose the use of *subband predictors* in the prediction stage of adaptive context based lossless audio coding. The original wideband audio signal is first decomposed in subbands, linear prediction is then performed in each subband, possibly using different predictor orders, and finally the predicted values are transformed back to time domain. We take as a pilot signal decomposition the modulated lapped transform (MLT) and introduce several subband prediction algorithms, differing in the way the subband predictions are transferred back to time domain. Experimental results compare the entropy of prediction errors for the cases of subband prediction and fullband prediction.

1. INTRODUCTION

Linear prediction is often used to decorrelate the au-

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MLP LOSSLESS COMPRESSION

JR STUART, PG CRAVEN, MA GERZON, MJ LAW, RJ WILSON

Meridian Audio Ltd, Huntingdon, England
jrs@meridian-audio.com

INTRODUCTION

Meridian Lossless Packing (MLP) is a lossless coding system for use on high-quality digital audio data originally represented as linear PCM. High quality audio these days implies high sample rates, large word sizes and multichannel.
This paper describes the MLP system and is an abbreviated version of [14].

1 OVERVIEW

MLP performs lossless compression of up to 63 audio channels including 24-bit material sampled at rates as high as 192kHz.
Lossless compression has many applications in the recording and distribution of audio. In designing MLP we have paid a lot of attention to the application of lossless compression to data-rate-limited transmission (e.g. storage on DVD), to the option of constant data

Figure 1: An overview of MLP used on disc.

2 LOSSLESS COMPRESSION

Unlike perceptual or lossy data reduction, lossless

Optimization of Digital Audio for Internet Transmission

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GRADIENT-DESCENT BASED WINDOW OPTIMIZATION FOR LINEAR PREDICTION ANALYSIS

Wai C. Chu

DoCoMo USA Labs – Mobile Media Laboratory
181 Metro Drive, Suite 300, San Jose, CA 95110, U.S.A.
wai@docomolabs-usa.com

ABSTRACT

The autocorrelation method of linear prediction (LP) analysis relies on a window for data extraction; we propose an approach to optimize the window based on gradient-descent. It is shown that the optimized window has improved performance with respect to popular windows, such as Hamming. The technique has potential in quality improvement for many LP-based speech coders.

1. INTRODUCTION

The autocorrelation method of LP analysis [1] is widely adopted by many modern speech coding algorithms. The basic procedure consists of signal windowing,

present study an optimization procedure that can lead to windows with better performance. The technique is based on gradient-descent where the gradient is derived from the Levinson-Durbin algorithm.

2. OPTIMAL WINDOW

Input signal samples located within the *analysis interval* are processed to obtain the LP coefficients, these are used for actual synthesis or prediction inside the *synthesis interval*. The two intervals might not be the same in practice. Several metrics are defined to quantify the performance of a given window. The prediction-error energy at the synthesis interval $n \in [n_1, n_2]$ is given by

$$J = \sum_{n=n_1}^{n_2} (e[n])^2 = \sum_{n=n_1}^{n_2} (s[n] - \hat{s}[n])^2$$

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AN INTER-CHANNEL REDUNDANCY REMOVAL APPROACH FOR HIGH-QUALITY MULTICHANNEL AUDIO COMPRESSION

Hongmei Ai, Chris Kyriakakis and C.-C. Jay Kuo

Integrated Media Systems Center
partment of Electrical Engineering-Systems
University of Southern California
Los Angeles, CA 90089-2564, USA
PH: 213-740-8386 FAX: 213-740-4651

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Lossless Audio Coding

Mathieu Hans and Ronald Schafer
Technical Report (CSIP TR-97-07)
Center for Signal & Image Processing (CSIP)
School of Electrical and Computer Engineering
Georgia Institute of Technology, Atlanta, GA
email:{hans,rws}@ece.gatech.edu

November 30, 1997

Abstract

This Technical Report surveys and classifies the currently available and best performing lossless audio codecs. Our study suggests that these codecs reached a limit in compression that is very modest compared to lossy audio coding technology. Assuming this limit to be near the theoretical entropy, we designed a simple, lossless audio codec—AudioPaK—, which uses only a few integer arithmetic operations and performs as well, or better than most state-of-the-art lossless codecs. The main operations of this codec are polynomial prediction and Golomb coding. These operations are done on a frame basis. The complete architecture of AudioPaK is presented.

1 Why Lossless Audio Coding ?

Lossless audio coding of stereo CD quality digital audio signal sampled at 44.1 KHz and quantized on 16 bits will become an essential technology for digital music dis-

IEEE TRANSACTIONS ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING, VOL. ASSP-32, NO. 6, DECEMBER 1984 1213

Two-Dimensional Linear Prediction and Its Application to Adaptive Predictive Coding of Images

PETROS A. MARAGOS, STUDENT MEMBER, IEEE, R
RUSSELL M. MERSEREA

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1 Why Lossle

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SHORTEN: Simple lossless and near-lossless waveform compression

Tony Robinson

Technical report CUED/F-INFENG/TR.156

Cambridge University Engineering Department,
Trumpington Street, Cambridge, CB2 1PZ, UK

December 1994

Abstract

This report describes a program that performs compression of waveform files such as audio data. A simple predictive model of the waveform is used followed by Huffman coding of the prediction residuals. This is both fast and near optimal for many commonly occuring waveform signals. This framework is then extended to lossy coding under the conditions of maximising the segmental signal to noise ratio on a per frame basis and coding to a fixed acceptable signal to noise ratio.

1 Introduction

It is common to store digitised waveforms on computers and the resulting files can often consume significant amounts of storage space. General compression algorithms do not

L REDUNDANCY REMOVAL APPROACH
UALITY MULTICHANNEL AUDIO
COMPRESSION

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ment of Electrical Engineering-Systems
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FROM SUBBANDS FOR LOSSLESS AUDIO
COMPRESSION

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nd predictors
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the subband
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of prediction
and fullband

1. INTRODUCTION

Linear prediction is often used to decorrelate the au-

order entropy of the fullband, for any finite p, whe
p is the size of the block of source samples. Encou
aged by these analytical results, we investigate in th
paper the performance of subband prediction agai
the performance of fullband prediction, with the
tended use of both predictors for lossless wideba
audio coding.

After some experiments with Quadrature Mirr
Filter Banks (QMF)[2] and Modulated Lapped Tra
form (MLT)[3] we decided to analyse only the pred
tion in the subbands built with the Modulated Lapp
Transform, since with QMF the reconstructed sign
may have discontinuities at block boundaries, which
clearly a major drawback for a good prediction. T
modulated lapped transform was extensively used
audio compression schemes (MPEG 1, MPEG 2 a
AC-3)[4][6] to transform the time representation
the signal into a time frequency representation, mo
suitable for perceptually masking and quantization

Work begins...

(mid 2000)

outline (~) - VIM - Shell - Konsole

Session Edit View Bookmarks Settings Help

```
FRONT PAGE:
    - what is @@@
        - lossless
        - approx 2:1 compress on music
        - supports sample-accurate seeking
        - which also means it is suitable for streaming
        - open-source, open-format, patent free
    - technical description on @@@ format
    - download
        - source
        - binaries
        - rpms
        - config file for grip to use @@@ as encoder
    - mailing lists (announce, developer)
TECHNICAL DESCRIPTION:
goals:
    - open format, not patent-infringing (too many patent-protected, non-open formats)
    - lossless (plenty of lossy formats)
    - respectable compression for important streams
    - sample-accurate random access (seeking)
    - faster-than-realtime decoding
    - no DRM!
properties:
    - variable frame size
    - header includes sample number of frame start
    - unique bit streams in header for random access
predictor:
    - polynomial appoximator OR
    - LPC (n-order FIR) OR
    - RLE of constant blocks
    - parameters (huffman?) encoded
    - error signal Golomb/Rice coded
parameters:
    - min/max frame sizes (multiple of 192)
    - min/max polynomial order
    - min/max FIR order
links:
    - shorten
    - ogg squish
"outline" 45L, 1169B
```

Shell

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outline (~) - VIM - Shell

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Shell

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outline (~) - VIM - Shell

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2025-10-25

Work begins...

(mid 2000)

Work begins...

(mid 2000)

Work begins...

(mid 2000)

Work begins...

(mid 2000)

Version 0.1

Work begins...

(mid 2000)

Version 0.1: lost to time

Needs a name!

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Free Lossless Audio Coder

Needs a name!

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Free Lossless Audio ~~Coder~~
Codec

A FLAC is Born

A FLAC is Born

Version 0.2

A FLAC is Born

Version 0.2

CVS repo on Sourceforge

A FLAC is Born

Version 0.2

CVS repo on Sourceforge

December 10, 2000

FLAC

status

FLAC is currently in the alpha stage. The current version is 0.2. The format is well defined but may change enough before becoming beta to break older streams. It also has not gone through enough testing yet to be considered archival quality. You should keep those two things in mind when using any alpha or beta versions of FLAC.

That said, we intend to settle on the format quickly, then start releasing beta versions. If you use FLAC and have suggestions or bugs, please join the mailing list or developers group and help us move to an official 1.0 version.

what is FLAC?

FLAC stands for Free Lossless Audio Coder. The FLAC project consists of:

- the stream format
- libFLAC, which implements a reference encoder, stream decoder, and file decoder
- flac, which is a command-line wrapper around libFLAC to encode and decode .flac files
- input plugins for various music players (Winamp, XMMS, and more in the works)

"Free" means that the specification of the stream format is in the public domain (the FLAC project reserves the right to set the FLAC specification and certify compliance), and that neither the FLAC format nor any of the implemented encoding/decoding methods are covered by any patent. It also means that the source for libFLAC is available under the LGPL and the sources for flac and the plugins are available under the GPL.

See the features page, documentation page, or FLAC format page for more info, the comparison page to see how the reference encoder measures up, or the goals page for what the FLAC project hopes to achieve.

download

Visit the download page for links to the source code or pre-built binaries, or go directly to the source on SourceForge.

documentation

The documentation is available online as well as in the distributions. The general installation and usage documentation for flac and the plugins is here. For a detailed description of the FLAC format and reference encoder see the FLAC format page.

message from the maintainer

I came up with FLAC because no audio compression format I could find did everything I needed. Since I couldn't mash them all together (most are closed-source), I solidified all my requirements (now the FLAC goals) and wrote the first implementation. I intended to open-source it from the beginning for two reasons: 1) so that people who knew more about audio compression than me could help improve it; and 2) I wanted to give something back to the OS community, whose huge body of work I rely on so much.

So I started the FLAC project on SourceForge as soon as I had a relatively complete first implementation. Now I'm the maintainer of the FLAC project. You can get in touch with me about it through the mailing list or directly

--Josh Coalson

news

10-Dec-2000 :
FLAC debuts on SourceForge

links

SourceForge

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FLAC - Free Lossless Audio Coder - Konqueror

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Location: /home/jcoalson/flac-1.1.1.1/doc/index.html

Google Search

FLAC

FLAC

load | features | goals | format | comparison | docu

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FLAC - Free Lossless Au

8:22
2025-10-25

goals

Since FLAC is an open-source project, it's important to have a set of goals that everyone works to. They may change slightly from time to time but they're a good guideline. Changes should be in line with the goals and should not attempt to embrace any of the anti-goals!

Goals

- FLAC should be and stay an open format. The source code is all either LGPL'd or GPL'd.
- FLAC should be lossless. This seems obvious but lossy compression seems to creep into every audio coder. This goal also means that flac should stay archival quality and be truly lossless for all input. Testing of releases should be thorough.
- FLAC should yield respectable compression, on par or better than other lossless coders.
- FLAC should allow at least realtime decoding on even modest hardware.
- FLAC should support fast sample-accurate seeking.
- FLAC should allow gapless playback of consecutive streams. This follows from the lossless goal.
- The FLAC project owes a lot to the many people who have advanced the audio compression field so freely, and aims also to contribute through the open-source development of new ideas.

Anti-goals

- Lossy compression. There are already many suitable lossy format (Ogg Vorbis, MP3, etc.).
- Copy protection of any kind. Don't get me started, just see the features page for the short answer.

METADATA_BLOCK_DATA	
METADATA_BLOCK_ENCODING	Currently, FLAC specifies only one metadata block, the ENCODING block.

METADATA_BLOCK_ENCODING	
<16>	The minimum block size (in samples) used in the stream.
<16>	The maximum block size (in samples) used in the stream. (Minimum blocksize == maximum blocksize) implies a fixed-blocksize stream.
<16>	The minimum frame size (in bytes) used in the stream. May be 0 to imply the value is not known.
<16>	The maximum frame size (in bytes) used in the stream. May be 0 to imply the value is not known.
<20>	Sample rate in Hz. Though 20 bits are available, the maximum sample rate is limited by the structure of frame headers to 1048570Hz. Also, a value of 0 is invalid.
<3>	(number of channels)-1. FLAC supports from 1 to 8 channels
<5>	(bits per sample)-1. FLAC supports from 1 to 32 bits per sample. Currently the reference encoder and decoders only support up to 24 bits per sample.
<36>	Total samples in stream. 'Samples' means channel-wide sample, i.e. one second of 44.1Khz audio will have 44100 samples regardless of the number of channels. A value of zero here means the number of total samples is unknown.
	NOTES • FLAC specifies a minimum block size of 16 and a maximum block size of 65535, meaning the bit patterns corresponding to the numbers 0-15 in the minimum blocksize and maximum blocksize fields are invalid.

FRAME	
FRAME_HEADER	nbsp;
SUBFRAME+	One SUBFRAME per channel.
<?>	Zero-padding to byte alignment.

FRAME_HEADER	
<9>	sync code '111111110'
<3>	block size in channel-wide samples: • `000` : get from ENCODING metadata block • `001` : 192 samples • `010-101` : 576 * (2^(2-n)) samples, i.e. 576/1152/2304/4608 • `110` : get 8 bit (blocksize-1) from end of header • `111` : get 16 bit (blocksize-1) from end of header
<4>	sample rate: • `0000` : get from ENCODING metadata block • `0001-0011` : reserved • `0100` : 8kHz • `0101` : 16kHz • `0110` : 22.05kHz • `0111` : 24kHz • `1000` : 32kHz • `1001` : 44.1kHz • `1010` : 48kHz • `1011` : 96kHz • `1100` : get 8 bit sample rate (in kHz) from end of header • `1101` : get 16 bit sample rate (in Hz) from end of header • `1110` : get 16 bit sample rate (in tens of Hz) from end of header • `1111` : invalid, to prevent sync-fooling string of 1s

FLAC - format - Konqueror

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Location: /home/jcoalson/flac-1.1.1.1/doc/format.html

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FLAC - format - Konque

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2025-10-25

FLAC - comparison - Konqueror

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Location: /home/jcoalson/flac-1.1.1.1/doc/comparison.html

Codec	Source Available?	Plugins Available?	Streamable?	Seekable?	Cost	OS support
flac v0.2	YES	YES (Winamp, XMMS)	YES	YES	FREE	ANY (source)
Shorten v2.3a	YES	YES (Winamp plugin somewhere)	no	no	FREE	ANY (source)
LPAC v1.20 (codec 2.0)	no	YES (Winamp only)	no?	YES	FREE	Windows/Linux/Solaris
Monkey's Audio v3.71	no	YES (Winamp only)	YES	YES	FREE	Windows only
RKAU v1.06	no	YES (Winamp only)	no	YES	FREE	Windows only
WavPack v3.6	no	no	no	no	FREE	Windows only
WaveZIP v2	no	no	no	no	FREE (24-bit costs $)	Windows only
Pegasus-SPS	no	no	no	no	$39 (free trial)	Windows only

The machine I used for encoding the test files is a PII-333 with 256 megs of RAM, running Windows NT 4.0 SP5. Unfortunately, Windows is the lowest-common-denominator platform for all the encoders.

The input corpus currently consists entirely of CD music tracks. In the future it may include more kinds of input (like speech, other sample rates, etc). There are ??? tracks whose genres range from death metal to pop to western classical to Indian classical.

In all tables, the results are sorter by compression ratio, which is compressed size / uncompressed size. The first table is a summary of results on all input tracks. The remaining table shows the results of the encoders on each track.

Some interesting things to note: LPAC quality settings are not too stable with -r (which allows seeking during playback) turned on. In most cases the 'normal' mode makes the smallest file, and much faster. RKAU also has a tendency to get bigger in the 'high' mode. Shorten's method for quantizing and transmitting the LPC coefficients is not very good which is the main reason why the fixed predictors runs are both smaller and faster.

Another ironic fact is that the encoders that are patented or cost money turn out to be the worst by most measures. SPS is so archane and crippled that I gave up trying to put together results for it after one track.

Encoder	Encode time	Compressed size	Compression ratio
Monkey's Audio 3.80 (extra high)	20:24.18	381.85 MB	0.5073
RKAU 1.06 (normal)	52:53.46	383.36 MB	0.5093
RKAU 1.06 (high)	133:25.36	383.75 MB	0.5098
Monkey's Audio 3.80 (high)	7:45.75	387.97 MB	0.5154
LPAC 1.20 (-r, normal)	20:49.34	393.16 MB	0.5223
LPAC 1.20 (-r, high)	88:20.18	393.12 MB	0.5223
LPAC 1.20 (-r, extra high)	109:27.86	393.93 MB	0.5234
flac 0.2 (-6)	41:11.46	399.30 MB	0.5305

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FLAC - comparison - Ko
8:28 2025-10-25


FORM TX
DO NOT WRITE ABOVE THIS LINE
TITLE OF THIS WORK
1
2
a
NOTE
b
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3
4
61
circular
Copyright
Registration
for Computer
Programs
DEFINITION
A "computer program" is a set of statements or instructions to be used directly or indirectly in a computer in order to bring about a certain result.
EXTENT OF COPYRIGHT PROTECTION
Copyright protection extends to all of the copyrightable expression embodied in the computer program. Copyright protection is not available for ideas, program logic, algorithms, systems, methods, concepts, or layouts.
WHAT TO SEND
• A completed application form (typically Form TX)
• A $30* nonrefundable filing fee payable to the Register of Copyrights
• One copy of identifying material (see "Deposit Requirements" below)
Mail all of the above material in the same envelope or package addressed to:
Library of Congress
Copyright Office
101 Independence Avenue, S.E.
Washington, D.C. 20559-6000
*NOTE: Registration filing fees are effective through June 30, 2002. For information on the fee changes, please write the Copyright Office, check the Copyright Office Website at www.loc.gov/copyright, or call (202) 707-3000 for the latest fee information.

- Fixed a bug in **libFLAC** that happened when using an exhaustive LPC coefficient quantization search with 8 bps input.
- Fixed a bug in **libFLAC** where the error estimation in the fixed predictor could overflow.
- Fixed a bug in **libFLAC** where LPC was attempted even when the autocorrelation coefficients implied it wouldn't help.
- Reworked the LPC coefficient quantizer, which also fixed another bug that might occur in rare cases.
- Really fixed the '-V overflow' bug (c.f. bug #231976).
- Fixed a bug in **flac** related to the decode buffer sizing.

FLAC is very close to being ready for an official release. The only known problems left are with the Winamp plugins, which should be fixed soon, and pipes with MSVC.

12-Feb-2001:

- **FLAC 0.7 released.** This is mainly a bug fix release, specifically:
  - Fixed a bug that happened when both -fr and --seek were used at the same time.
  - Fixed a bug with -p (c.f. bug #230992).
  - Fixed a bug that happened when using large (>32K) blocksizes and -V (c.f. bug #231976).
  - Fixed a bug where encoder was double-closing a file.
  - Expanded the test suite.
  - Added more optimization flags for gcc, which should speed up flac.

28-Jan-2001:

- **FLAC 0.6 released.** The encoder is now much faster. The -m option has been sped up by 4x and -r improved, meaning that in the default compression mode (-6), encoding should be at least 3 times faster. Other changes:
  - Some bugs related to **flac** and pipes were fixed (see here for the discussion).
  - A "loose mid-side" (-M) option to the encoder has been added, which adaptively switches between independent and mid-side coding, instead of the exhaustive search that -m does.
  - An analyze mode (-a) has been added to **flac**. This is useful mainly for developers; currently it will dump info about each frame and subframe to a file. It's a text file in a format that can be easily processed by scripts; a separate analysis program is in the works.
  - The source now has an autoconf/libtool-based build system. This should allow the source to build "out-of-the-box" on many more platforms.

15-Jan-2001:

- **FLAC 0.5 released.** This is the first beta version of FLAC. Being beta, there will be no changes to the format that will break older streams, unless a serious bug involving the format is found. What this means is that, barring such a bug, streams created with 0.5 will be decodable by future versions. This version also includes some new features:
  - An MD5 signature of the unencoded audio is computed during encoding, and stored in the Encoding metadata block in the stream header. When decoding, **flac** will now compute the MD5 signature of the decoded data and compare it against the signature in the stream header.
  - A test mode (-t) has been added to **flac**. It works like decode mode but doesn't write an output file.

23-Dec-2000:

- **FLAC 0.4 released.** This version fixes a bug in the constant subframe detection. More importantly, a verify option (-V) has been added to **flac** that verifies the encoding process. With this option turned on, **flac** will create a parallel decoder while encoding to make sure that the encoded output decodes to exactly match the original input. In this way, any unknown bug in the encoder will be caught and **flac** will abort with an error message.

10-Dec-2000:

- **FLAC debuts on SourceForge.** The FLAC project is now being hosted on SourceForge. Visit the FLAC project page to join the mailing list or sign up as a developer.

FLAC - news - Konqueror

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Location: /home/jcoalson/flac-0.8/doc/html/news.html

- Fixed a bug in **libFLAC** that happened when using an exhaustive LPC c
- Fixed a bug in **libFLAC** where the error estimation in the fixed predicto
- Fixed a bug in **libFLAC** where LPC was attempted even when the autoc
- Reworked the LPC coefficient quantizer, which also fixed another bug th
- Really fixed the '-V overflow' bug (c.f. bug #231976).
- Fixed a bug in **flac** related to the decode buffer sizing.

FLAC is very close to being ready for an official release. The only known prob

12-Feb-2001:

- **FLAC 0.7 released.** This is mainly a bug fix release, specifically:
  - Fixed a bug that happened when both -fr and --seek were used at the
  - Fixed a bug with -p (c.f. bug #230992).
  - Fixed a bug that happened when using large (>32K) blocksizes and -V
  - Fixed a bug where encoder was double-closing a file.
  - Expanded the test suite.
  - Added more optimization flags for gcc, which should speed up flac.

28-Jan-2001:

- **FLAC 0.6 released.** The encoder is now much faster. The -m option has be
  - Some bugs related to **flac** and pipes were fixed (see here for the discu
  - A "loose mid-side" (-M) option to the encoder has been added, which a
  - An analyze mode (-a) has been added to **flac**. This is useful mainly fo ... nalysis program is in the works.
  - The source now has an autoconf/libtool-based build system. This shoul

15-Jan-2001:

- **FLAC 0.5 released.** This is the first beta version of FLAC. Being beta, there ... e it streams created with 0.5 will be decodable by future versions. This version a
  - An MD5 signature of the unencoded audio is computed during encodin against the signature in the stream header.
  - A test mode (-t) has been added to **flac**. It works like decode mode b

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- **FLAC 0.4 released.** This version fixes a bug in the constant subframe dete ... ecoder while encoding to make sure that the encoded output decodes to exactly ma

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FLAC - news - Konquero

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2025-10-25

07-Jun-2001:

- **FLAC 0.10 released.** This is probably the final beta. There have been many improvements in the last two months:
  - Both the encoder and decoder have been significantly sped up. Aside from C improvements, the code base now has an assembly infrastructure that allows assembly routines for different architectures to be easily integrated. Many key routines have now have faster IA-32 implementations (thanks to Miroslav).
  - A new metadata block SEEKTABLE has been defined to hold an arbitrary number of seek points, which speeds up seeking within a stream.
  - `flac` now has a command-line usage similar to 'gzip'; make sure to see the latest documentation for the new usage. It also attempts to preserve the input file's timestamp and permissions.
  - The -# options in `flac` have been tweaked to yield the best compression-to-encode-time ratios. The new default is -5.
  - `flac` can now usually autodetect WAVE files when encoding so that -fw is usually not needed when encoding from stdin.
  - The WAVE reader in `flac` now just ignores (with a warning) unsupported sub-chunks instead of aborting with an error.
  - Added an option '--delete-input-file' to `flac` which automatically deletes the input after a successful encode/decode.
  - Added an option '-o' to `flac` to force the output file name (the old usage of "flac - outputfilename" is no longer supported).
  - Changed the XMMS plugin to send smaller chunks of samples (now 512) so that visualization is not slow.
  - Fixed a bug in the stream decoder where the decoded samples counter got corrupted after a seek.

  It should be a short hop to 1.0.

31-Mar-2001:

- **FLAC 0.9 released.** There were some format changes that broke backwards compatibility but these should be the last (see below). Also, there have been several bug fixes and some new features:
  - FLAC's sync code has been lengthened to 14 bits from 9 bits. This should enable a faster and more robust synchronization mechanism.
  - Two reserved bits were added to the frame header.
  - A CRC-16 was added to the FLAC frame footer, and the decoder now does frame integrity checking based on the CRC.
  - The format now includes a new subframe field to indicate when a subblock has one or more 0 LSBs for all samples. This increases compression on some kinds of data.
  - Added two options to the analysis mode, one for including the residual signal in the analysis file, and one for generating gnuplot files of each subframe's residual distribution with some statistics. See the latest documentation.
  - XMMS plugin now supports 8-bit files.
  - Fixed a bug in the Winamp2 plugin where the audio sounded garbled.
  - Fixed a bug in the Winamp2 plugin where Winamp would hang sporadically at the end of a track (c.f. bug #231197).

  FLAC is on track for an official 1.0 release soon.

05-Mar-2001:

- **FLAC 0.8 released.** This release is a result of extensive testing and fixes several bugs encountered when pushing the encoder to the limit. I'm pretty confident in the stability of the encoder/decoder now for all kinds of input. There have also been several features added. Here is a complete list of the changes since 0.7:
  - Created a new utility called `metaflac`. It is a metadata editor for .flac files. Right now it just lists the contents of the metadata blocks but eventually it will allow update/insertion/deletion.
  - Added two new metadata blocks: PADDING which has an obvious function, and APPLICATION, which is meant to be open to third party applications. See the latest format docs for more info, or the new id registration page.
  - Added a -P option to `flac` to reserve a PADDING block when encoding.
  - Added support for 24-bit files to `flac` (the FLAC format always supported it).
  - Started the Winamp3 plugin.
  - Greatly expanded the test suite, adding more streams (24-bit streams, noise streams, non-audio streams, more patterns) and more option combinations to the encoder. The test suite runs about 30 streams and over 5000 encodings now.
  - Fixed a bug in `libFLAC` that happened when using an exhaustive LPC coefficient quantization search with 8 bps input.
  - Fixed a bug in `libFLAC` where the error estimation in the fixed predictor could overflow.
  - Fixed a bug in `libFLAC` where LPC was attempted even when the autocorrelation coefficients implied it wouldn't help.

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FLAC

status

FLAC 1.0 is out! It's been a long time coming but it's finally here. Thanks to everyone who made it possible. FLAC is now clearly the strongest open source lossless audio codec and will continue to improve through the work of the open-source community.

The differences since 0.10 are mostly bug fixes but there is a new feature for aligning encoded files on sector boundaries. The complete list of changes is here.

FLAC 1.0 has been through a huge amount of testing now, including tens-of-thousands of unit and regression tests, and thousands of real-life CD audio tracks. However, if you're paranoid like me you can keep using the verify option (-V) when archiving.

If you use FLAC and have suggestions or patches, please join the mailing list or developers group. Bugs can be filed here.

what is FLAC?

FLAC stands for Free Lossless Audio Codec. Grossly oversimplified, FLAC is similar to MP3, but lossless. The FLAC project consists of:

- the stream format
- **libFLAC**, which implements reference encoders and decoders
- **flac**, a command-line wrapper around libFLAC to encode and decode .flac files
- input plugins for various music players (Winamp, XMMS, and more in the works)

news

20-Jul-2001 :
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06-Jun-2001 :
Version 0.10 released

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05-Mar-2001 :
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12-Feb-2001 :
Version 0.7 released

28-Jan-2001 :
FLAC gets FASTER for version 0.6

15-Jan-2001 :
FLAC goes beta with version 0.5

23-Dec-2000 :
Version 0.4 released

10-Dec-2000 :
FLAC debuts on SourceForge

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status

PhatNoise (makers of the PhatBox, which also plays FLAC) just released their Home Digital Media Player. It includes a DMS cartridge slot so you can pop out your FLAC tunes and pop 'em in your car.

Slim Devices' new Squeezebox, the wireless follow-on to the SliMP3 networked audio player, is available and supports FLAC and Ogg Vorbis.

Primus is offering soundboard recordings from 2003 Tour de Fromage in FLAC and MP3 on primuslive.com. More info here and here.

Independent record label Magnatune is now offering their catalog in FLAC and Vorbis in addition to MP3.

Rio has announced a new portable, the Rio Karma, which supports FLAC and Ogg Vorbis.

livephish.com is now offering recordings in FLAC format in addition to MP3.

last updated 2003-Nov-19

what is FLAC?

FLAC stands for Free Lossless Audio Codec. Grossly oversimplified, FLAC is similar to MP3, but lossless, meaning that audio is compressed in FLAC without any loss in quality. This is similar to how Zip works

news

19-Nov-2003 :
PhatNoise's new Home Digital Media Player supports FLAC

18-Nov-2003 :
Slim's new 'Squeezebox' supports FLAC

11-Nov-2003 :
Primus offers live shows in FLAC

13-Oct-2003 :
Magnatune catalog available in FLAC

11-Aug-2003 :
New Rio Karma supports FLAC

23-Jun-2003 :
livephish.com offers FLAC shows

09-Feb-2003 :
ReQuest adds FLAC support

29-Jan-2003 :
FLAC joins Xiph.org!

26-Jan-2003 :
Version 1.1.0 released

(all news)

flac

free lossless audio codec

facebook

2002 - Mike Wren

2005 Rebrand

FLAC - Free Lossless Audio Codec - Konqueror

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jcoalson@acheron:~/flac-1

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2025-10-26

Challenges…

Challenges...

MontaVista Software

SCO v. IBM: Lawsuit claims and counterclaims

- March 2003: SCO sues IBM
  - Main allegations: misappropriation of trade secrets (IBM's AIX product includes proprietary SCO code); breach of contract
  - Amended in July 2003 to include more specific claims of contract breach (IBM agreements and Sequent agreement)
- August 2003 and September 2003: IBM countersues SCO
  - Main allegations: breach of contract, Lanham Act and unfair competition; unfair/deceptive trade practices; patent infringement; copyright infringement; breach of GPL
    - GPL claim: "Hey, you distributed this same code under the GPL! How can you now say it's proprietary?"
- October 2003: SCO claims the GPL is unenforceable

Challenges...

MONTAVISTA SOFTWARE

Red Hat v. SCO

- August 2003: Red Hat
  - Declaratory judgment t
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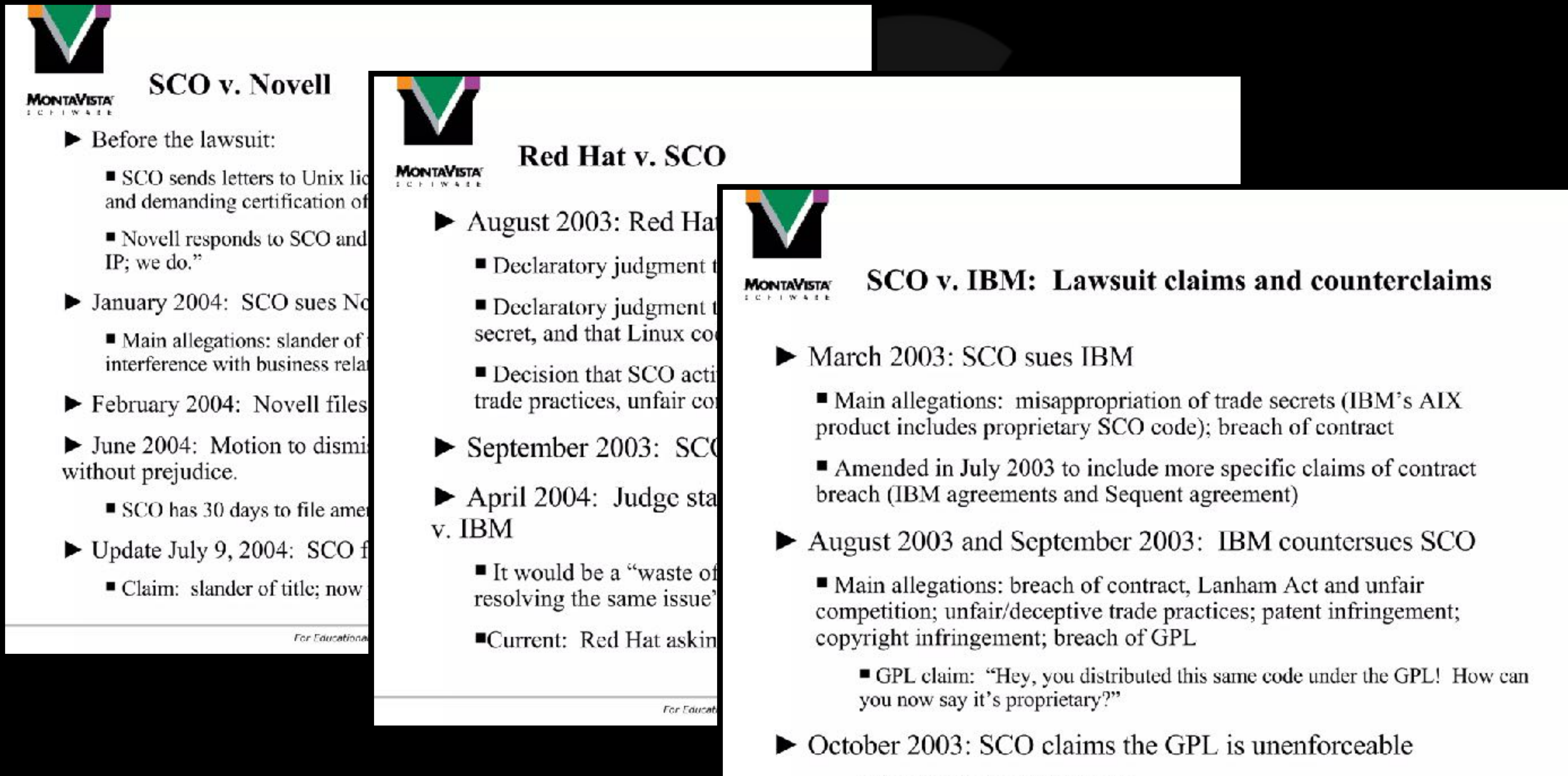

MONTAVISTA

SCO v. Novell

- Before the lawsuit:
  - SCO sends letters to Unix lic
    and demanding certification of
  - Novell responds to SCO and
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- January 2004: SCO sues No
  - Main allegations: slander of
    interference with business rela
- February 2004: Novell files
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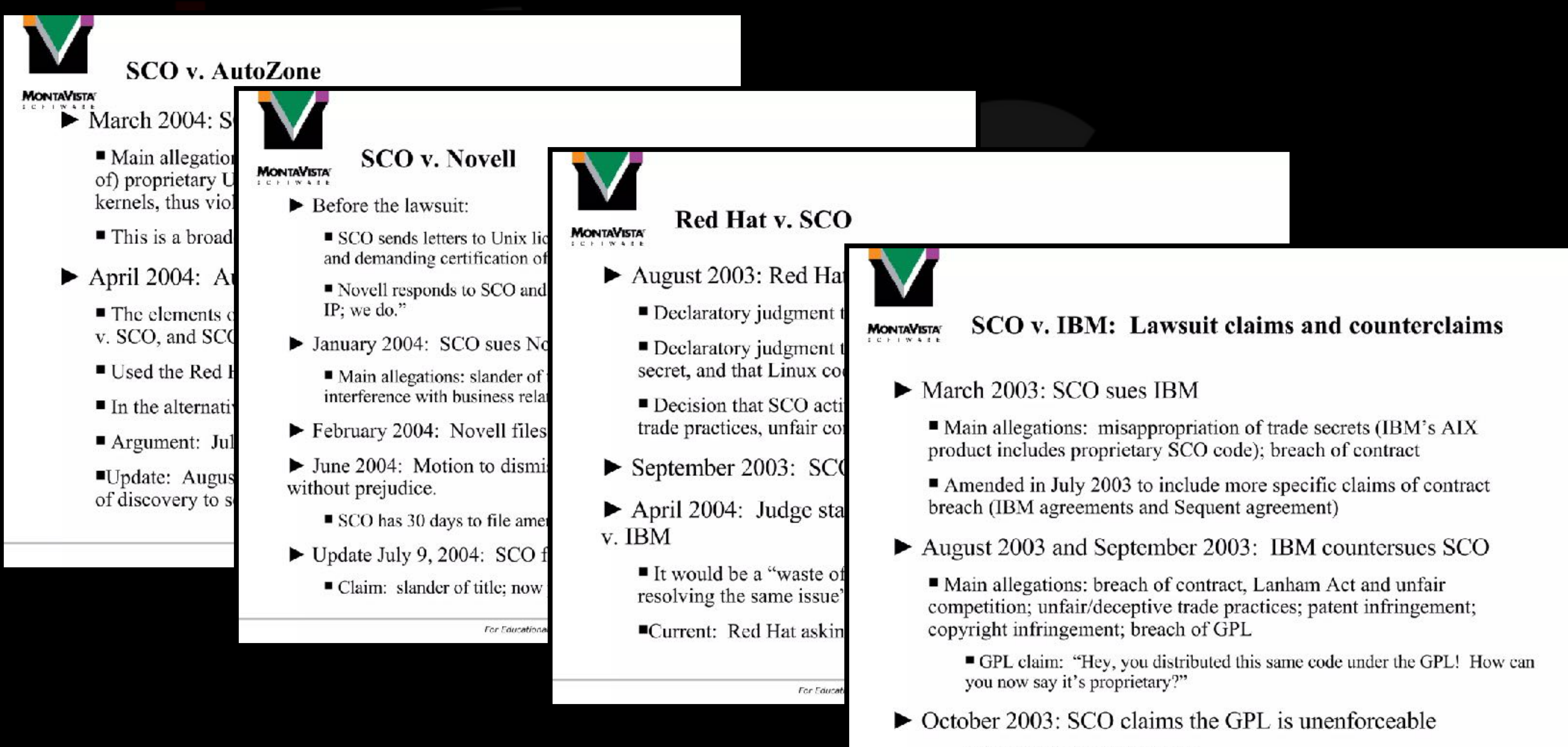
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WMA Lossless

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WMA Lossless

Apple Lossless

Good advice

"Switch license to BSD!"

Good advice

"Switch license to BSD!"

Bad advice

"FLAC should compress more!"

Advocacy

Advocacy

hydrogenaudio

the audio technology enthusiast's resource

flac

free lossless audio codec

Advocacy

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Advocacy


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Making Me Nervous
Brad Sucks
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Rio

Advocacy

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Home stereo:


Squeezebox
(our review)


Transporter
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Sonos
(our review)


Escient


Hifidelio


Olive


Arcus DAR300


Blackbird


McIntosh MS300


Helios X5000


Netgear EVA8000


ReQuest


MediaREADY MC


Zensonic Z500


Ziova


Sooloos


HD MediaBox


TViX 4/5000 Series

Other home stereo:

- Avega Systems' wireless Oyster loudspeakers
- Denon's SE-32 and SE-52 tabletop players

- PhatBox
- URAL ConceRt CDD

Portable/Handheld:


Cowon iAUDIO


i-Station mini DX


Iwod G10


KNC HR-2800


Meizu M6 Miniplayer


Onda VX737


Rio Karma


Teclast TL-29


TrekStor Vibez


Bluedot BMP-1430

Other Portable/Handheld:

- Gemei X-750 and X-760
- Hyundai NH-260
- iPod via the Rockbox firmware replacement
- iRiver iHP-120/iHP-140/H320/H340 via the Rockbox firmware replacement
- Maxian D900
- OPPO Blast
- Portable Media Player
- Shearer V2000
- Zarva MV209

Other:

- Numark's DJ equipment like the HDX and CDX turntables with integrated hard drive and CD player, and the HDMIX mix

Bands!

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Primus offers live shows in FLAC

11 Nov 2003

Primus is offering soundboard recordings from 2003 Tour de Fromage in FLAC and MP3 on primuslive.com

Next (Slim's new 'Squeezebox' supports FLAC) »

« Previous (Magnatune catalog available in FLAC)

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03 Mar 2004

Metallica is offering soundboard recordings of live shows in FLAC format

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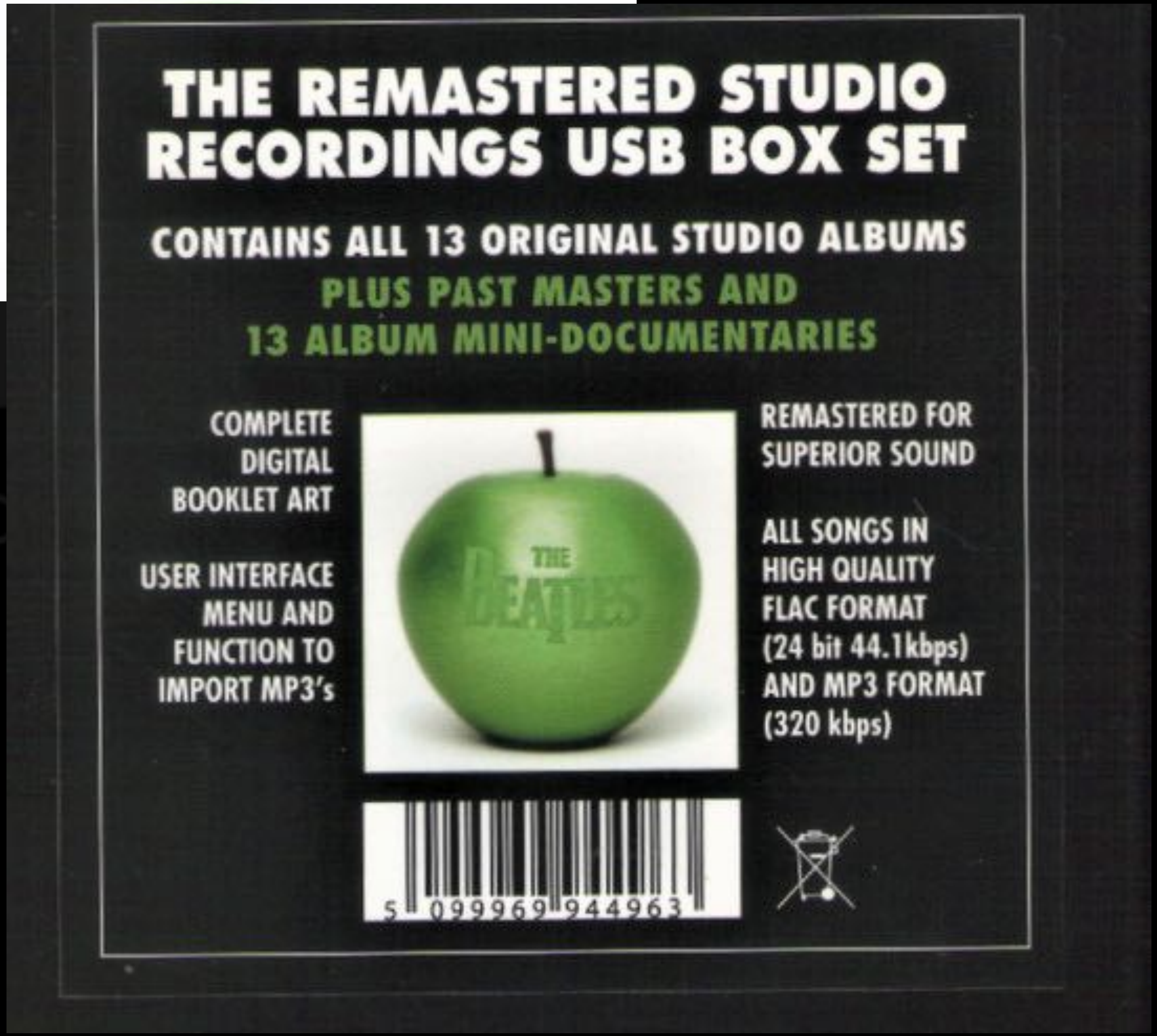


Topic: Deutsche Grammophon selling albums in FLAC

previous topic - next topic

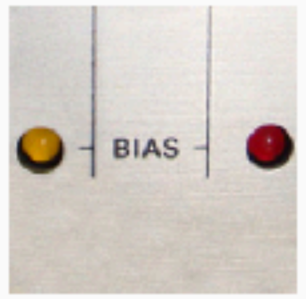
Highlander

2008-12-21 22:58:28 Deutsche Grammophon sellin...

I just received a newsletter from Deutsche Grammophon and among the news I noticed that DG is starting to sell some albums of his catalogue in FLAC. For now, there is only a selection of 50 albums avalaible in FLAC, but more should follow.

The selection available in FLAC is here

Kees de Visser



Reply #1 – 2008-12-21 23:38:37 Deutsche Grammophon sellin...

Interesting development. FLAC is a bit more expensive compared to MP3 but that sounds fair.
With a bit of effort it's probably possible to find better CD offers, but downloads can be fast and convenient.

€ 19.99 Mail Order
€ 12.99 Download FLAC Lossless
€ 10.99 Download MP3 320
€ 0.99 Stream (7 Days)

Mainstream

flac

free lossless audio codec

Mainstream

WILL WORK FOR NOTHING

HERE'S A PROBLEM FOR BUSINESS SCHOOL PROFESSORS: How do you prosper in a marketplace where the only way your product will be a success is if you don't charge any money for it? For David Bryant and Josh Coalson, the answer is simple: Make sure you enjoy what you are doing.

Bryant, 49, and Coalson, 39, are both experienced computer programmers living in Silicon Valley. They pay the rent by laboring during the day in the tech world's digital salt mines. Their nights and weekends are also spent hunched over computer

Mainstream

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Coalson says that in the early days of Flac, it was like having a second job. Now the program's success draws in other volunteers offering to help.

When the megawealthy freely appropriate your open-source code to get even richer, don't you feel like a sap?

The situation is not without its cruel ironies. Bryant, for example, says he started his work in part because of his interest in music; he collects vintage audio equipment from the 1950s and 1960s, the sort made out of tubes rather than transistors. But with no time for serious listening, he leaves much of the gear packed in boxes in his house.

People unfamiliar with the world of open-source software often presume its adherents to be hostile to property, profits and the other mainstays of capitalism. Not true. Bryant, for example, has plans for a for-profit program unrelated to WavPack, and both men say they would consider consulting engagements from companies trying to get the most out of their software.

Apple refuses to support either man's file system in the iPod; presumably, it thinks that its own Apple Lossless format is good enough. But Flac files are everywhere on the Internet now. And WavPack automatically creates a smaller, compressed version of a file along with the main one. Wouldn't it be nice if Apple hired these guys to make their files work on iPods and iPhones? F

Forbes.com Senior Editor **Lee Gomes**, in our Silicon Valley bureau, can be reached at **lgomes@forbes.com**. Visit him at **www.forbes.com/gomes**.

Wrapping up

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Network effects take over

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2013 - Xiph adopts FLAC project

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The moral of the story?