FH MMA SALZBURG – MUSIC PRODUCTION, MIX & MASTERING

DIGITAL AUDIO FORMATS, AUDIO DRIVERS AND PLUGINS

Figure 1: RME Fireface UFX rear panel, featuring MIDI, Firewire, USB2, Wordclock, 2x ADAT, AES/EBU and analogue I/O

1. DIGITAL AUDIO FORMATS

WORDCLOCK

Figure 2: Wordclock BNC connectors

It is used to synchronize digital audio devices together, such as audio interfaces, AD/DA converter units, preamps with digital output, digital effects, etc.

The device that maintains the clock is called the *Master* (set to *internal clock*), all others must be configured as *Slaves* (set to *external clock*).

Many of the audio formats described in the following pages can transport Wordclock together with the digital audio signal, and do not require a separate Wordclock connector. However, when trying to maintain sync across several different devices, it is advisable to use a high precision central Wordclock generator.

If additional slave devices should be connected, but the master device has no more free outputs, a Wordclock generator or distribution amplifier should be used.

Figure 3: Wordclock BNC cable

AES/EBU (2-CHANNEL INTERFACE)

AES (Audio Engineering Society)
EBU (European Broadcasting Union)

This 2-Channel Interface was already specified back in 1985 and later updated in 1992. It allows the transfer of PCM audio with up to 24-bit precision, up to 96 kHz sampling rate (originally only up to 48 kHz). Recently it has been updated to support also 176,4 and 192 kHz sampling rates.

Figure 4 - AES/EBU XLR connectors

AES/EBU uses balanced cables with XLR connectors (AES3) for the signal transfer. This ensures highest reliability also in critical conditions. Sometimes also coaxial BNC cables are used to connect with video recorders (AES3-ID).


Figure 5 - AES/EBU XLR cable

One cable transfers the information for two channels (each frame contains 2 interleaved subframes, right + left channel). One subframe data-word is 32-bit long, of which 20 are reserved for audio. Using the additional 4 user-bits it is possible to transfer also 24-bit samples. The AES/EBU interface usually allows synchronization directly from the input signal (no additional wordclock sync cable is required).

ADAT (ALESIS DIGITAL AUDIO TAPE)

Multichannel Optical Interface (TOSLINK)

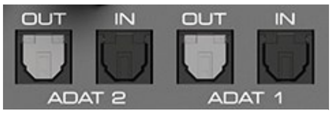

Figure 6: ADAT TOSLINK connector

This format was originally developed for Alesis Digital Audio Tape 8-track recorders. It is nowadays used by a large number of audio cards, AD/DA interface, musical instruments, etc. The standard connector is the TOSLINK (TOShiba-LINK) fibre-optical connector (also called *lightpipe*), through which it is possible to transfer 8-channels of 24-bit audio at up to 48 kHz sampling rate.


Figure 7: ADAT TOSLINK cable

To support higher sampling rates, S-MUX was developed, which splits the datawords across 2 audio channels (so 1 TOSLINK can support 4 Ch. at 96 kHz, and 2 TOSLINKs 8 Ch. at 96 kHz).

TDIF (TASCAM DIGITAL INTERFACE FORMAT)

This format was developed for Tascam's series of digital 8-channel recorders. It can transfer up to 8 tracks at up to 24-bit / 96 kHz precision. The standard connector is a DB-25 (similar to a PC printer cable). The channels are transferred in pairs, and they are also written in pairs on the digital multitrack recorders.

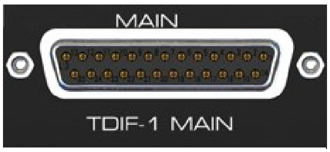

Figure 8: TDIF DB-25 connector

In addition to the audio data, LRCK (Left Right ClocK) is transferred in both directions, which allows synchronization across devices.


Figure 9: TDIF DB-25 cable

MADI (MULTICHANNEL AUDIO DIGITAL INTERFACE)

Figure 10: MADI fibre-optical and coaxial connectors

This is a modern multichannel digital audio format, developed by AMS Neve, Solid State Logic, Sony and Mitsubishi. It was introduced in 1989 and finally specified in 1991 (AES10-1991). The norm was later updated in AES10-2003 and AES10-2008. It supports serial digital transmission of 28, 56 or 64 channels and sampling rates up to 96 kHz.

As specified in AES10-2003:

- 32 kHz to 48 kHz ± 12.5%, 56 channels
- 32 kHz to 48 kHz nominal, 64 channels
- 64 kHz to 96 kHz ± 12.5%, 28 channels

Figure 11: MADI fibre-optical cable

MADI uses coaxial cables with BNC connectors for the signal transfer (similar to those used for Wordclock). Also fiber- optical connectors are supported, they are however different and incompatible with ADAT TOSLINK. The data format is very similar to AES/EBU, however the information for all channels is interleaved.

Among the advantages of fiber-optical connections, they can be run on relatively long distances (100m or more) without signal transmission problems.

S/P-DIF (SONY/PHILIPS DIGITAL INTERFACE FORMAT)

Figure 12: S/P-DIF Toslink and Coaxial connectors

This format is very similar to AES/EBU, only the standard connectors are coaxial (unbalanced RCA-type chinch) or optical (TOSLINK). With the exception of some status-bits, the data stream is identical with AES/EBU and it is usually possible to connect AES/EBU with S/P-DIF successfully.

Figure 13: Antelope Audio OCX HD Master Wordclock, featuring WC, Video, S/P-DIF and AES I/O

2. AUDIO DRIVER ARCHITECTURES

WINDOWS 7 / 8 / 10

- **WDM Audio** *Windows Driver Model*, previously known as MME (*Multi Media Extensions*)
  standard Windows native audio driver
  used for CD/DVD playback and basic stereo I/O audio applications

  not to be used for professional multitrack audio recording
  (very high latency, no multitrack capability)

- **DirectX Audio** part of the *DirectX* APIs, that include Direct-Input, Direct-Sound, and Video drivers
  generally used for Windows Games only, bad for low latency audio

- **ASIO** *Audio Stream Input/Output*

  Professional multichannel I/O, low latency driver (down to 1,5 ms latency and less)
  supported by Steinberg Cubase, Nuendo, Wavelab, Sony Vegas, Presonus Studio One, Ableton Live, iZotope Studio RX, etc.

- **DAE** *Digidesign Audio Engine*

  Professional multichannel I/O, very low latency driver
  used by Avid Pro-Tools TDM/HD/HDX (DSP based systems)

MAC OS X

- **Core Audio** MacOS native audio driver
  multichannel I/O and low latency capability
  supported by Logic Audio and most MacOS audio applications

- **ASIO** *Audio Stream Input/Output*
  Professional multichannel I/O, low latency driver (down to 1,5 ms and less)
  supported by Steinberg Cubase, Nuendo, Wavelab, etc.

- **DAE** *Digidesign Audio Engine*
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LINUX / IOS

- **Class Compliant** Some sound interfaces, like the RME Fireface UCX, support in addition to *USB* or *Thunderbolt* and *stand-alone* mode, also *Class Compliant* mode. This is a standard that is natively supported by operating systems like Windows, Mac OSX, iOS and Linux.

  No proprietary drivers are required; the device will be directly recognized when the CC firmware is loaded. Usually, native features will be limited in comparison to those provided with a specific USB or Thunderbolt driver.

3. SOFTWARE PLUGIN FORMATS

WINDOWS NATIVE

- **Direct-X** native Windows standard – nowadays supported only by a few applications (like Sony CD Architect 5)
- **VST** supported by Steinberg Cubase, Nuendo, Wavelab, Sony Sound Forge, Sony Vegas, PreSonus Studio One, Ableton Live, etc.
- **RTAS** older format: still supported by Avid Pro Tools MP 9, Pro Tools 10 and HD 10
- **AAX Native** new format: requires Avid Pro Tools 10, HD 10 or higher

MAC OS X NATIVE

- **AU** native MacOS standard – supported by Logic Audio, Final Cut Pro, Garage Band, and most MacOS X Audio applications
- **VST** supported by Steinberg Cubase, Nuendo, Wavelab
- **RTAS** older format: still supported by Avid Pro Tools MP 9, Pro Tools 10 and HD 10
- **AAX Native** new format: requires Avid Pro Tools 10, HD 10 or higher

WINDOWS DSP-BASED

- **TDM** older format: still supported by Avid Pro Tools HD 10
- **AAX DSP** new format: requires Avid Pro Tools HD 10 and a Pro Tools HDX DSP card
- **UAD** requires a Universal Audio UAD-1 or UAD-2 DSP card (PCIe, USB or Thunderbolt)
UAD plugins are available to the host DAW in VST/RTAS/AAX format

MAC OS X DSP-BASED

- **TDM** older format: still supported by Avid Pro Tools HD 10
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UAD plugins are available to the host DAW in VST/AU/RTAS/AAX format

Legend:

- AU Audio Units
- AAX Avid Audio Extensions (Avid, Pro Tools Native and DSP)
- RTAS Real Time Audio Suite (Avid, Pro Tools Native)
- VST Virtual Studio Technology (Steinberg)
- TDM Time Division Multiplexing (Pro Tools, DSP)
- UAD Universal Audio

VST2 VS VST3

VST3 plugins offer several advantages compared to VST2 versions, for example:

- support for a "side-chain" input (which is particularly useful in combination with dynamic processors)
- audio input bus for VST instruments (for example for a vocoder synthesizer)
- dynamic I/O configuration (they can adapt to the channel they are inserted in, whether it is mono, stereo, surround, etc.)
- resizable GUI
- improved accuracy for parameter automation
- improved stability and reliability

You can read more about the VST3 standards here: http://www.steinberg.net/de/company/technologies/vst3.html

Installed Plugin Versions

In Cubase, you can check the VST version of the loaded VST Effects and VST Instruments in the Plugin Manager.
If a project is loaded, you will also see here how many instances of each plugin are running.

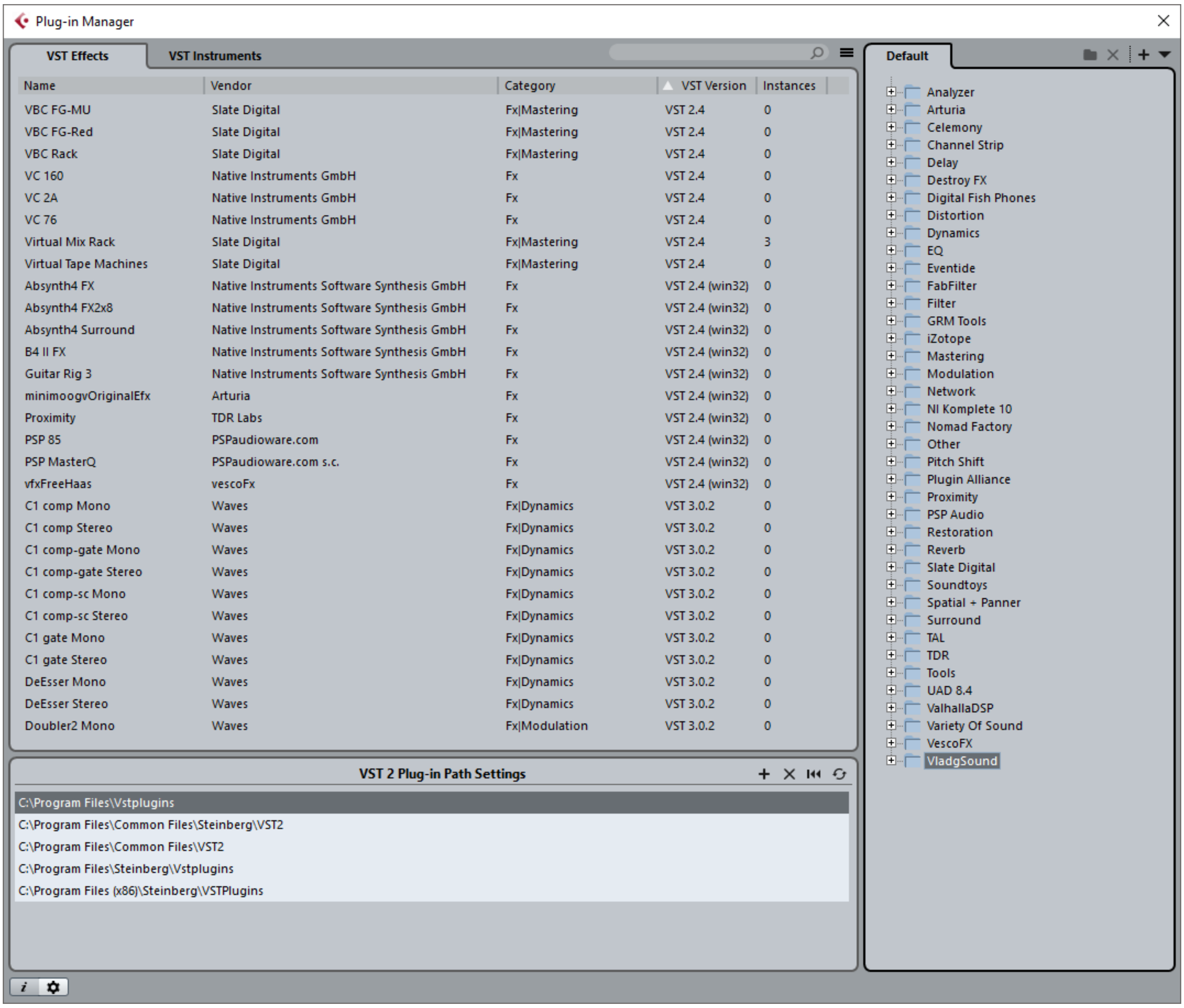

Figure 14: Cubase Plug-in Manager > VST Effects and VST Instruments, displaying the installed VST plugins, category, and version

32 VS 64 BIT OPERATING SYSTEMS

Some operating systems, such as Windows 7, 8 and 10, are available both as 32-bit and 64-bit versions.

The main issue with a 32-bit OS is that the max addressable RAM is limited to 4 GB (minus the RAM used for the video card). You can install more RAM, but it will not be recognized by the system and cannot be used.

A 64-bit OS can overcome this limit. This is particularly useful when working on a DAW project many multiple software sampler instruments and high quality sample libraries, that might require several GB of memory.

Physical Memory Limit on 32-bit & 64-bit OS			
Operating System	**Version**	**bit**	**Physical Memory Limit**
Windows	10 Home, Pro, Edu, Enterprise	32	About 3 GB
	10 Home	64	128 GB
	10 Pro, Edu, Enterprise		2 TB (2 000 GB)
	10 Server 2016 Standard, Datacenter		24 TB (24 000 GB)
Mac OS X	Lion	32	4 GB
	Lion	64	18 Exabytes (Hardware: 96 GB)
	Maverick		18 Exabytes (Hardware: 128 GB)
	Yosemite		18 Exabytes (Hardware: ??? GB)

32 VS 64 BIT PROGRAMS AND PLUGINS

Some audio applications, such as Cubase, are both available as 32-bit and 64-bit versions. The same goes for most VST2 and VST3 audio plugins.

<u>For maximum compatibility and performance, use a 64-bit OS with a 64-bit DAW and 64-bit versions of all plugins</u>.

32-bit & 64-bit OS, Cubase and Plugin Compatibility Chart					
Operating System	**Cubase version**	**VST2 32-bit**	**VST2 64-bit**	**VST3 32-bit**	**VST3 64-bit**
Windows 32-bit	Cubase 32-bit	✓		✓	
Windows 64-bit	Cubase 32-bit	✓		✓	
	Cubase 64-bit	✓ (through VST Bridge)	✓		✓
Mac OS X 32-bit	Cubase 32-bit	✓		✓	
Max OS X 64-bit	Cubase 32-bit	✓		✓	
	Cubase 64-bit	✓ (through VST Bridge)	✓		✓

RUNNING CUBASE UNDER WINDOWS

If you use a 32-bit versions of Windows (7, 8, 10), you can only install the 32-bit version of Cubase.

If you use a 64-bit versions of Windows (7, 8, 10), you have the choice to install either the 32-bit or the 64-bit (native) version of Cubase (version 6 or higher). You can also install both versions and choose which one to use depending on the project requirements.

The 32-bit version of Cubase only supports 32-bit VST2 and VST3 plugins and is limited to 4GB max addressable RAM (regardless if the host OS is Windows 32-bit or 64-bit).

The 64-bit of Cubase supports 64-bit VST2 and VST3, and 32-bit VST2 through the “VST Bridge”. This provides compatibility up to VST 2.4 plugins; however, several plugins do not run very reliably and might cause the VST Bridge to crash. It is therefore recommended to upgrade all 32-bit plugins to 64-bit versions.

RUNNING CUBASE UNDER MAC OS X

On Mac OS X Cubase is available since version 6 as hybrid 32/64 bit executable. Depending on the OS version, the 32 or the 64-bit version will be used. All current Macs (as for 2016) run Mac OS X with a 64-bit kernel.

On a 64-bit OS it is possible to force the 32-bit mode opening the “Info” window of the Cubase executable and selecting the option “Open in 32-bit mode”. This might be useful if you need to run the program with older 32-but plugins that are either unsupported or not available in 64-bit mode.

Also on Mac OS X, Cubase 64-bit offers compatibility with 32-bit VST 2.4 plugins through the VST Bridge. For best performance, it is recommended also under Mac OS X to upgrade all plugins to 64-bit versions.

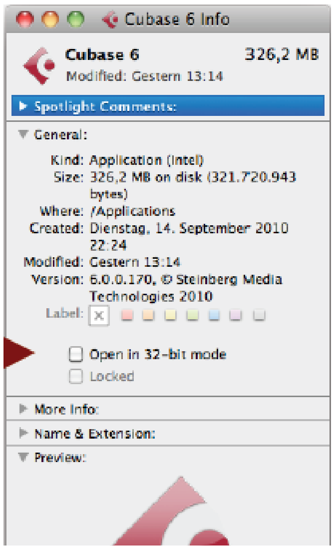

Figure 15: Cubase Info on Mac OS X (you can select to open the program in 32-bit mode)

CHOSING WHAT PLUGINS TO INSTALL

Most installation programs let you choose the plugin format (AAX, RTAS, VST2, VST3) and bit version you want to install. You can also set the location of your default VST plugin folders (separately for 32-bit and 64-bit versions).

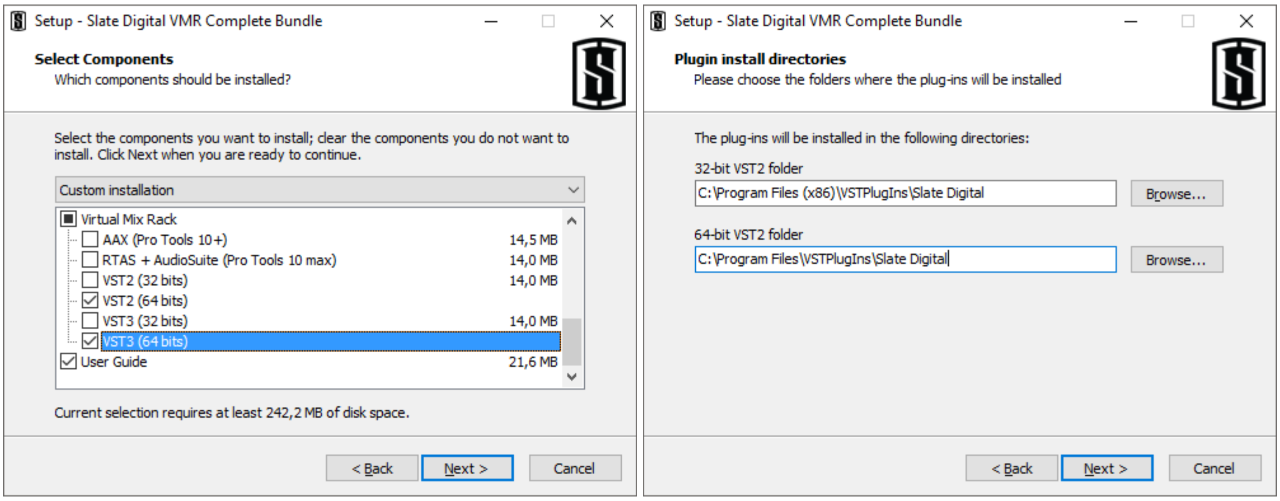

Figure 16: Slate Digital VMR installation. Left: selection of the plugin format (AAX, RTAS, VST2, VST3) and bit version (32 and 64 bit). Right: setting of the VST folder for VST2 plugins. VST3 plugins are installed in a separate (hidden) folder.

4. AUDIO FILE FORMATS

UNCOMPRESSED FILE FORMATS

- **PCM** — *Pulse Code Modulation*
  Standard uncompressed audio format, supporting a wide range of quantization depths (16, 24-bit, 32-bit float) and sampling rates (44,1 / 48 / 88,2 / 96 / 176,4 / 192 / 384 kHz)
  This is the standard format of any audio stream transferred through AES/EBU, ADAT, TDIF, MADI and S/P-DIF

- **WAV** — *Waveform Audio File Format*
  Standard container format for uncompressed LPCM (Linear PCM) audio under Windows; the WAV file is an instance of a RIFF (Resource Inter change File Format) defined by IBM and Microsoft; originally limited to 4 GB file size;
  WAV audio files can be mono, stereo or multichannel interleaved audio

- **BWF** — *Broadcasting Wave Format*
  Same as WAV, but it includes metadata with timecode information (source time)
  Format of choice when exchanging project files across different DAWs

- **WAV 64 / RF 64** — Based on the WAV format, but supports file sizes beyond 4 GB; BWF compatible
  Cubase can be set to switch automatically to RF 64 when the file size exceeds 4 GB

- **AIFF** — *Audio Interchange File Format*
  Standard container format for uncompressed LPCM (Linear PCM) audio under Mac OS X
  The format was developed by Apple in 1988 and is based on IFF (previously used on Amiga systems)
  AIFF audio files can be mono, stereo or multichannel interleaved audio

- **DSD** — *Direct Stream Digital*
  An audio format substantially different from PCM, as it uses pulse-density modulation encoding; the signal is stored as delta-sigma modulated digital audio; a sequence of single-bit values at a sampling rate of 2,8224 MHz (64 times the CD Audio sampling rate of 44,1 kHz)

  DSD is a recording and delivery format (Super Audio CD), but cannot be edited/mixed natively by almost any DAW, with only a few exceptions

LOSSLESS COMPRESSION FILE FORMATS

- **FLAC** — *Free Lossless Audio Codec*
  Audio coding format for lossless compression of digital audio. Digital audio compressed by FLAC can typically be reduced to 50-60% of the original size.
  It only supports PCM audio with fixed-point sample resolution, from 4 to 32 bits per sample, but no floating-point (so also not 32-bit float)
  Supports 1 to 8 channels of audio (including mono, stereo and 5.1 surround)
  FLAX uses linear prediction to convert the audio samples

- **ALAC** — *Apple Lossless Audio Codec*
  Similar to FLAC, developed by Apple for lossless compression of audio data.
  As iTunes does not natively support FLAC, users that want to use a format that supports metadata have use ALAC instead.

LOSSY COMPRESSION FILE FORMATS

- **MP3** *MPEG-1 and /or MPEG-2 Audio Layer III*
  Commonly referred as “MP3”
  The most widely used audio coding standard for lossy digital audio compression
  Used both for streaming and storage, it has become a de-facto standard on most digital audio players and computing devices
  It was designed by the Moving Picture Export Group as part of the MPEG-1 standard.
  The first subgroup for audio was mainly formed by several teams of engineers at Fraunho fer IIS, University of Hannover, AT&T / Bell Labs, etc. It was finalized in 1993-1995.
  Compared to CD quality uncompressed audio, MP3 usually achieves 75 to 95% reduction in size.
  Compression is achieved by reducing or approximating the accuracy of certain parts of the audio stream that are considered to be beyond the auditory resolution ability of most listeners. This method is often referred to as *perceptual coding*.
  Unfortunately the MP3 artefacts (loss in spatial perception, smearing of the transients, high frequency distortion, inter-sample clipping etc,) are quite noticeable at lower bi rates.
  At 320 kbps the quality is however almost comparable to CD standard

- **AAC** *Advanced Audio Coding*
  Audio coding standard for lossy digital audio compression
  Designed to be the successor of MP3, it generally achieves better sound quality than MP3 at similar bit rates
  Default audio format for YouTUbe, iPhone, iPod, iPad, Nintendo DSi, Nintendo 3DS, DivX Plus Web Player and PlayStation 3
  Compression is achieved discarding signal components that are perceptually irrelevant and eliminating redundancies in the coded audio signal (in a fashion similar to MP3, but with more advanced algorithms)

- **Dolby AC-3** *Audio Codec 3 / Advanced Codec 3*
  Also known as “Dolby Digital”, or “DD”
  Supports mono, stereo and multichannel audio formats (for example 5.1 Surround)
  Supports sample-rates up to 48 kHz and a maximum coded bit rate of 640 kbps.
  35mm film prints use a fixed rate of 320 kbps (same as the maximum rate for 2-channel MP3). DVD-Video disks are limited to 448 kbps, while Blu-Ray Disc, the PlayStation 3 and Xbox can output AC-3 at full 640 kbps.

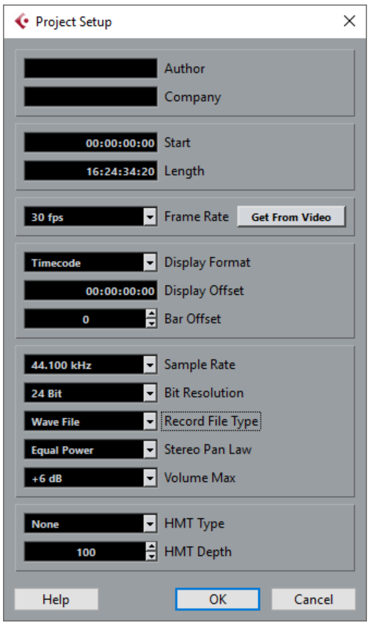

Figure 17F: Project settings in Cubase, including PCM audio file format

RECOMMENDED LITERATURE

- HENLE, Hubert: *das Tonstudio Handbuch* – GC Carstensen 2001 (ISBN 3-910098-19-3)

WEBSITE

- http://www.digitalnaturalsound.com/fh--multimediaart/music-production.html
- www.digitalnaturalsound.com or www.dns-studios.com > FH | MultiMediaArt > Music Production