24 bit 96 kHz Digital Audio Workstation using high performance Be Operating System on a multiprocessor Intel machine

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ABSTRACT

Digital Audio Workstation (DAW) has quickly established itself as the most important digital audio production tool. With the advent of high-resolution multi-channel audio formats and Internet audio exchange, the high performance and flawless operations of the desktop DAW have become a necessity. The authors explaining why current popular computer architectures are not suited to these new tasks, explore the possibilities of the new Be Operating System (BeOS)- specifically designed and optimized to handle digital audio and video. New features, unattainable with current operating systems, are discussed.

1. Introduction

Ever increasing performance of computers has caused a gradual migration of the key audio production tools from hardware embodiments to the virtual world of computers. This trend will continue, propelled by bottom line economics and the appearance of new features such as network audio exchange. Although current computer hardware is up to the task, existing general purpose operating systems are the actual performance bottleneck . Designed over 10 years ago for general purpose computing, they fail to meet increasing demands for speed and file size. The new BeOS has been designed from ground up to handle high bandwidth digital audio and video in a modern multiprocessing and multitasking environment.

The paper focuses *solely* on the use of commodity personal computers (IBM Compatibles and Apple) and their operating systems (Windows, MacOS, BeOS and Linux). Although number of specialized platforms such as SGI provides superior performance, they were omitted, as their presence in the current professional audio environment is minimal.

2. Personal Computer based DAW

The last decade witnessed a popularity surge of personal computer (PC) based DAW. Low cost, availability of commodity computer hardware and great flexibility are the main reasons for their popularity. As the processing power of PC's increase, they are bound to conquer larger areas of the music production process, gradually replacing dedicated audio hardware.

A designer of a PC based DAW must weigh hundreds of factors that will affect the quality and performance of the complete system.

All the elements-- signal integrity, quality and speed of signal processing, intended features, performance of the hardware and operating system have to be carefully balanced. It is not one, but all of the elements together that make a professional DAW.

These factors are analyzed further.

3. Most common desktop computer DAW architectures

The architecture of desktop computer based DAW can be generalized into two categories: DSP-Based DAW and Host-Based DAW.

3.1 DSP-based digital audio workstation

DSP-based digital audio workstation uses a dedicated DSP (digital signal processing) hardware for digital audio processing. Older designs used to handle all tasks related to digital audio data, including storage and networking **[1] [fig3.1]**. The computer host serves as a user interface and can also handle tasks related to digital audio data storage. The host software communicates with the dedicated DSP hardware through an API (Application Programming Interface). With a fast PC controlling multi-processor DSP hardware, fully functional multi-track recording and processing system have become a commercial reality **[2]**. Such systems are capable of providing functionality similar to that of a medium size 48 track recording studio. Current popular operating systems, Windows and MacOS are usually sufficient for this task. They are not involved in digital audio processing, although their shortcomings become apparent when the amount of tracks required and thus the data throughput, increase. BeOS data handling capacity and media oriented file system poses an advantage. **[fig3.2]**

3.2 Host-based digital audio workstation

Host based digital audio workstation uses the host computer for all the tasks, including audio processing. Its main advantage is the relatively low price of hardware. Processing large number of audio tracks is computationally intensive and is not handled well by either Windows or MacOS. Currently, it is not possible for a Host-based DAW to achieve the performance equivalent of a 48 track DSP-based system. However, it is possible to assemble a fully capable system for tasks requiring smaller number of tracks such as DAW's used for CD and DVD-Audio mastering. Significant improvements of the Host-based DAW can be achieved by employing the BeOS operating system. The Host-based DAW is the subject of further discussion. A high resolution, 8 channel 24 bit, 96 kHz digital audio workstation system discussed further in the paper is used as an example. **[fig3.3]**

4. Digital Signal Processing

4.1 – Host-based vs. Dedicated DSP systems

Use of digital signal processing chips (DSP), specifically designed for these tasks, is the most efficient way to implement audio processing within dedicated hardware. DSP should be used for large multitrack DAW systems, where numerous audio tracks require simultaneous processing. Smaller systems, like the ones required for CD and DVD mastering or multitrack editing, do not need dedicated hardware. Today's personal computer processors can do the job. How well it is done depends on the operating system used and efficiency of the code itself. The processing power of a microprocessor should be compared with a dedicated DSP chip. Although a microprocessor is substantially more powerful, its signal processing performance is compromised by non-optimal instruction set, operating system overhead and the fact that software is almost always compiled--hence the code is not optimal. DSP chips are usually efficiently programmed using assembly language and their architecture is specifically designed for signal processing. Vaguely estimated real life comparisons were made by the authors and similar findings were reported elsewhere **[3] [4]**. A single 400MHz Pentium II Processor and an 80 MHz Motorola DSP56301 can perform a similar amount of signal processing. A 400MHz Pentium III Processor with optimized code, can do up to 40% more processing. A 400MHz G3 PPC Processor can perform up to 80% better. These figures should be taken with caution as they may vary as much as three-fold - depending on the particular implementation of the microprocessor based solution. The choice of operating system and efficiency of the code become essential.
This comparison underlines the weaknesses of Host-based processing on uniprocessor machines used by Windows 98 and MacOS. BeOS, Windows 2000, Windows NT, Linux and MacOS X are multiprocessor capable, and can thus compete with smaller and medium DSP-based systems. A multiprocessor machine can provide sufficient processing power for a capable Host-based DAW, as long as the OS used supports symmetric multiprocessing.

4.2 – Digital signal processing using host microprocessors

A professional DAW must provide a sufficient degree of sound quality. 16 bit PCM audio is not adequate. 24 bit, 96kHz sampling PCM audio is currently considered a professional standard. Signal integrity requires a higher resolution processing than that of the source. A 32 bit floating point processing and storage is integral to most modern OSes and can provide excellent results **[5]**. A 24E8 (24 bit mantissa, 8 bit exponent) IEEE-754 format simplifies programming, and provides superior results to a single precision fixed 24 bit format used in some popular DSP chips. Constant dynamic range of floating point signal result in a better overall precision, and better handling of low level signals and multiple gain changes **[3]**. The floating point format is also less prone to programming errors, commonly found in bad sounding audio software packages.
The authors have proposed target resolution for each component of the 24 bit, 96 kHz Host-based DAW that guarantees signal integrity **[fig 4.1]**. A/D converters maintain 120dB dynamic range. Their 24 bit fixed point output signal is fed to the host computer using 24 bit capable digital I/O card. The processing is performed in 32 bit floating

format and disk storage is also 32 bit. The processed signal is then dithered at a level corresponding to 24 bit fixed word dither, converted to 24 bit fixed PCM and output using the DI/O card. Certain applications such as mastering, may need double precision processing. This is a standard function of Pentium processor and requires approximately twice the processing power after the overhead is counted in. **[6]**

In summary, the Host-based digital signal processing is easier, faster to implement than that of DSP-based systems, which almost always require hand written assembly codes. The Host-based code, often written in languages such as C and C++ is highly portable and can maintain very high signal quality if the 32 bit floating format is used **[7] [8]**.

5. The choice of operating system- the BeOS, a fresh approach

The currently proliferating operating systems-- Windows 95/98, Windows NT and Mac OS were originally designed over a decade ago, when the idea of personal computers handling real-time audio and video was pure science fiction. Windows 98 is a descendant of DOS; Windows NT of the VAX systems originally designed by DEC; Mac OS originates from Xerox Parc and Mach. The outdated design premise of these OSes is becoming an obstacle to new advances in real-time requirements. With these OSes, the only options for increased performance is to use faster hardware while patching around problems.

BeOS unlocks computer hardware power. Speed, multiprocessing, multitasking, multithreading and media oriented architecture make the BeOS the preferred platform for digital audio applications. Low latency, adequate clocking, media kit API, file system and hierarchy further assist the programmer in quickly developing high performance audio software. **[9] [10]**

5.1 - Real time and low latency processing requirement

Analog equipment processes sound close to the negligible speed of light, instantaneously. Real time or low latency is the basic requirement of almost any piece of recording equipment. Digital audio is a real time contiguous stream of data and should be processed as such. Latency is inherent to all digital audio systems and can be described as time that takes for the signal to travel from input to output. DSP-hardware based audio systems are almost always designed as “hard real time” systems. The latency is fixed and always guaranteed regardless of the system load. Most computer OSes don’t provide “hard real time” capability, although this may be attempted using Linux RTkernel **[11]**. A reasonable low latency, below 20 milliseconds, is required. The long latency is the major reason why Windows98 and NT are inadequate for audio processing. Disabling of interrupts by the OS causes latencies to often exceed 100ms in Windows98 and MacOS. Windows NT and MacOS X perform even worse. BeOS is specifically designed for low latency processing and currently achieves sub 10ms latencies by quickly handling 2.7 ms data buffers. The data buffers are adjustable down to 32 frames and latencies as low as 6ms are possible. It is possible to achieve low latencies in Windows or MacOS, but only

through special drivers for the particular audio I/O hardware, that can “hack” the OS. Microsoft proposed WDM architecture to improve Windows latency designed for consumer multimedia **[12].**

Typical latencies reported (standard/special drivers) **[3] [4]** :

BeOS – 10ms
Linux – 15ms
MacOS – 50ms/15ms
Windows 98 – 100ms/25ms
Windows NT – 100ms
MacOSX – 150ms
Yamaha 02R digital console “hard real time” – 20ms

5.2 – Multiprocessing, multitasking, multithreading and protected memory in BeOS

The advantages of using a multiprocessor machine as a Host-based DAW platform were explained in (*4.1*). Increased acceptance of multiprocessor machines is the current trend in computer hardware design. BeOS features *symmetric multiprocessing (SMP)* which spreads the processing load equally amongst all available processors. Up to 8 processors can be handled. Some operating systems such as Windows NT, Windows2000, MacOS X and Linux are also SMP capable.

BeOS *multitasking* allows for performing multiple tasks at once. They are prioritized along the media focus of the OS. Audio having the highest priority, next video, and then come other tasks.

Pervasive multithreading takes large tasks, such as applications, and breaks them down into a myriad of smaller tasks which can be performed in parallel. The user sees significant performance and capability increases as a result. The BeOS enables the programmer to write multithreaded applications, and thus complex software packages such as DAW can be highly optimized.

Protected memory in BeOS keeps the system and other applications running in case of an application crash.

5.3 – Media Kit – standardized Application Programming Interface in BeOS

A standardized comprehensive media API is built into BeOS **[13]**. It is specifically designed to handle real time media such as digital audio, MIDI and digital video. It provides for *low latency* data streaming, *computing of latencies* to synchronize multiple data streams, and *timing*. Timing can be based on any source such as DI/O, MIDI or system The data is timed in microseconds, thus allowing for sample accurate audio editing at sampling rates higher than 192 kHz **[14]**. A *“node”* based Media Kit

architecture allows for access of multiple audio applications to the same hardware or data. BeOS Media Kit is the most advanced audio API amongst all operating systems.

5.4 – 64 bit multithreaded journaling file system in BeOS

The 32 bit file system used in most operating systems limits the file size to 4 Gigabytes. A 15 min. long, 16 channel, 32 bit, 96 kHz audio take, occupies 5.5 GB easily exceeding this limit. The BeOS features a 64 bit file system capable of handling 18 billion GB files. In addition there are *database features* built in for quick retrievals, useful for applications such as sound libraries and samplers. The file system is capable of *journaling* interrupted file transfers, eliminating the need to restart from the beginning. There is also a *plug-in architecture* allowing for mounting other file systems such as Windows FAT and MacOS HFS. Any file system can be plugged-in to BeOS, allowing for easy data interchange. Popular file formats and format translation services are supported by BeOS. BeOS has *native TCP/IP networking* protocol.

5.5 – No legacy fresh OS foundation. Other features of BeOS

BeOS is a modern operating system designed from the ground up, specifically as a Media OS. Other operating systems especially MacOS and Windows carry layers of legacy code crippling the OS performance and substantially complicating programmer's job. They help cultivate bad programming habits and compromise performance of the system drivers which must support legacy hardware. BeOS has no inherited legacy, and is designed for maximum performance. Its system drivers are designed for modern hardware, resulting in highly improved overall performance. BeOS is programmed in ubiquitous object oriented C++, in a modern multithreaded object oriented style. Applications of similar functionality result in code 2 or 3 times smaller in BeOS when compared to Windows. As a result BeOS applications perform up to 4 times faster on the same hardware when compared to Windows 98 **[10]** .

6. An 8 Channel 96 kHz DAW using BeOS on a Four Processor Pentium III Machine.

An efficient, high performance Host-based DAW was assembled using a set of high quality 8 channel 24 bit; 96 kHz A/D and D/A converters; a PCI digital input/output card capable of carrying 16 channels of 24 bit 48 kHz using two ADAT lightpipe interfaces; and an "IBM compatible" computer **[15] [fig. 6.1]**. The computer hardware consisting of a four-processor Pentium III 400 MHz motherboard, a fast disk with an ultra-wide PCI SCSI interface and the necessary standard items. The system is capable of triple boot of Windows 98, Windows NT and BeOS. With the appropriate software 8 channels of 24 bit, 96kHz audio can be input and processed in real time. A significantly larger number of tracks can be edited and played back.
A sample multiplexing method **[16]** is used to carry 8 channels of 24 bit 96kHz audio over two ADAT lightpipe interfaces. The audio converter boxes provide format translation to and from standard AES/EBU format and low jitter system clocking.

SUMMARY

A professional, high performance, high resolution DAW can be built using multiprocessor Intel machine and Be Operating System. This solution is a robust, cost effective alternative to DSP-based medium size systems used for CD and DVD mastering and multitrack audio editing. The BeOS unlocks the hardware power and removes a number of performance problems associated with current DAW's such as instability, hardware and file format incompatibility, and poor system responsiveness. The BeOS based DAW is capable of challenges of the future Internet based audio production and distribution **[17]**.

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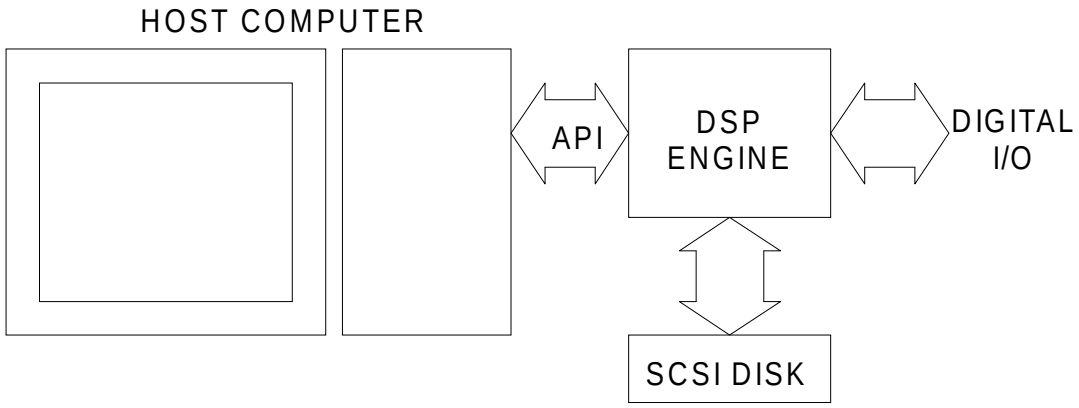
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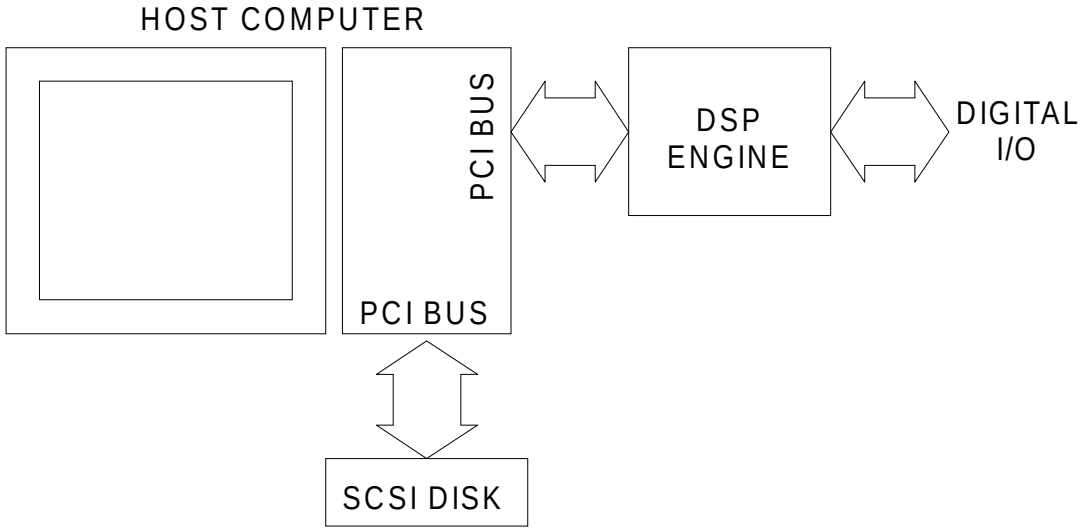
Fig. 3.1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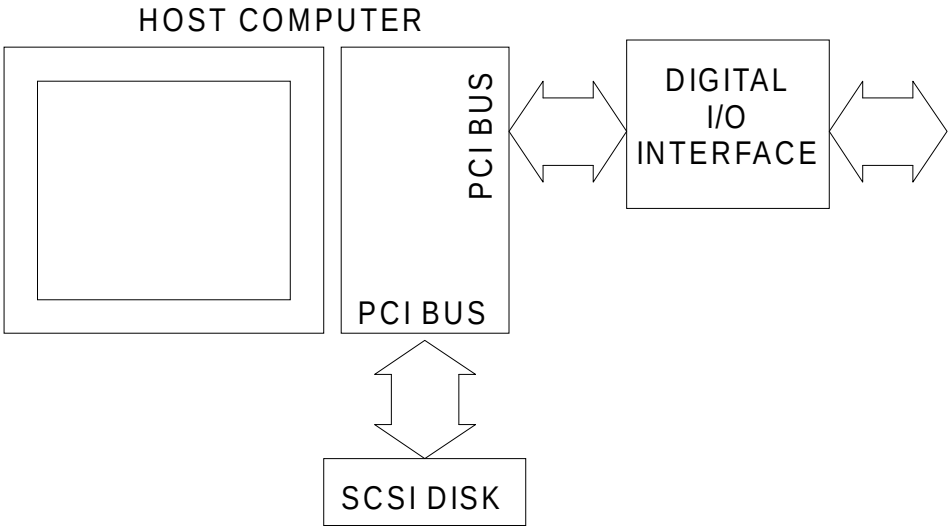
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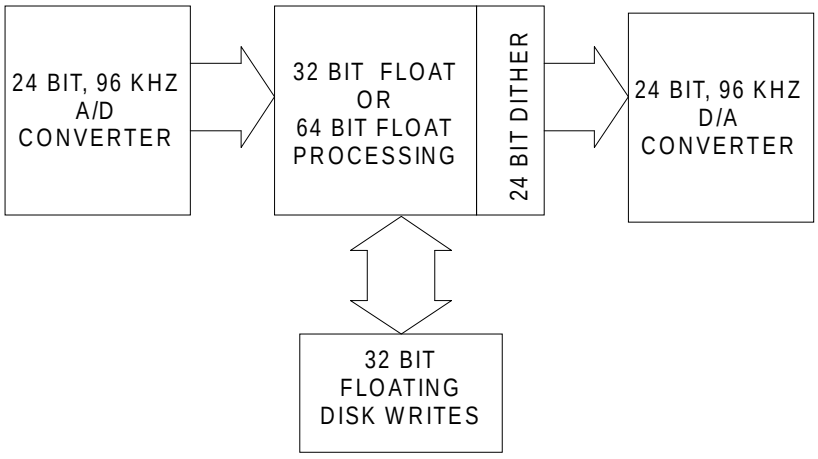
Fig. 3.3


24 BIT, 96 KHZ
A/D
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32 BIT FLOAT
OR
64 BIT FLOAT
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24 BIT DITHER
24 BIT, 96 KHZ
D/A
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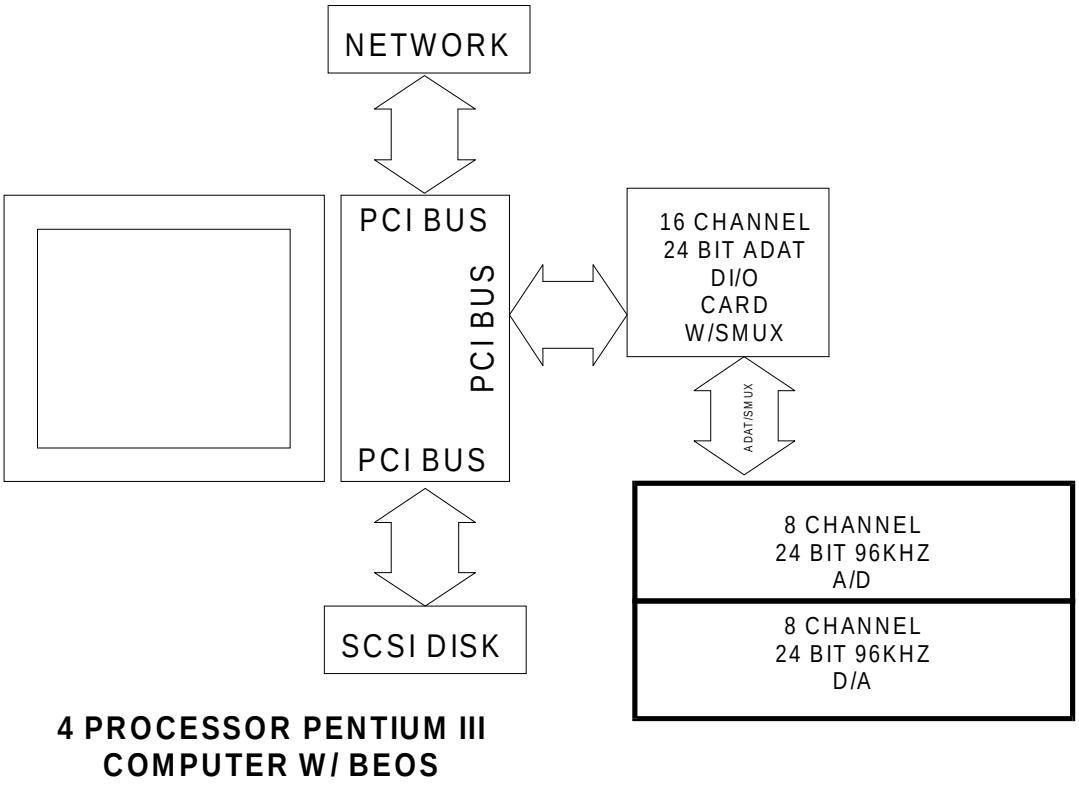
Fig. 4.1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8 CHANNEL
24 BIT 96KHZ
D/A
SCSI DISK
4 PROCESSOR PENTIUM III
COMPUTER W/ BEOS

Fig. 6.1