The Digital Audio Workstation

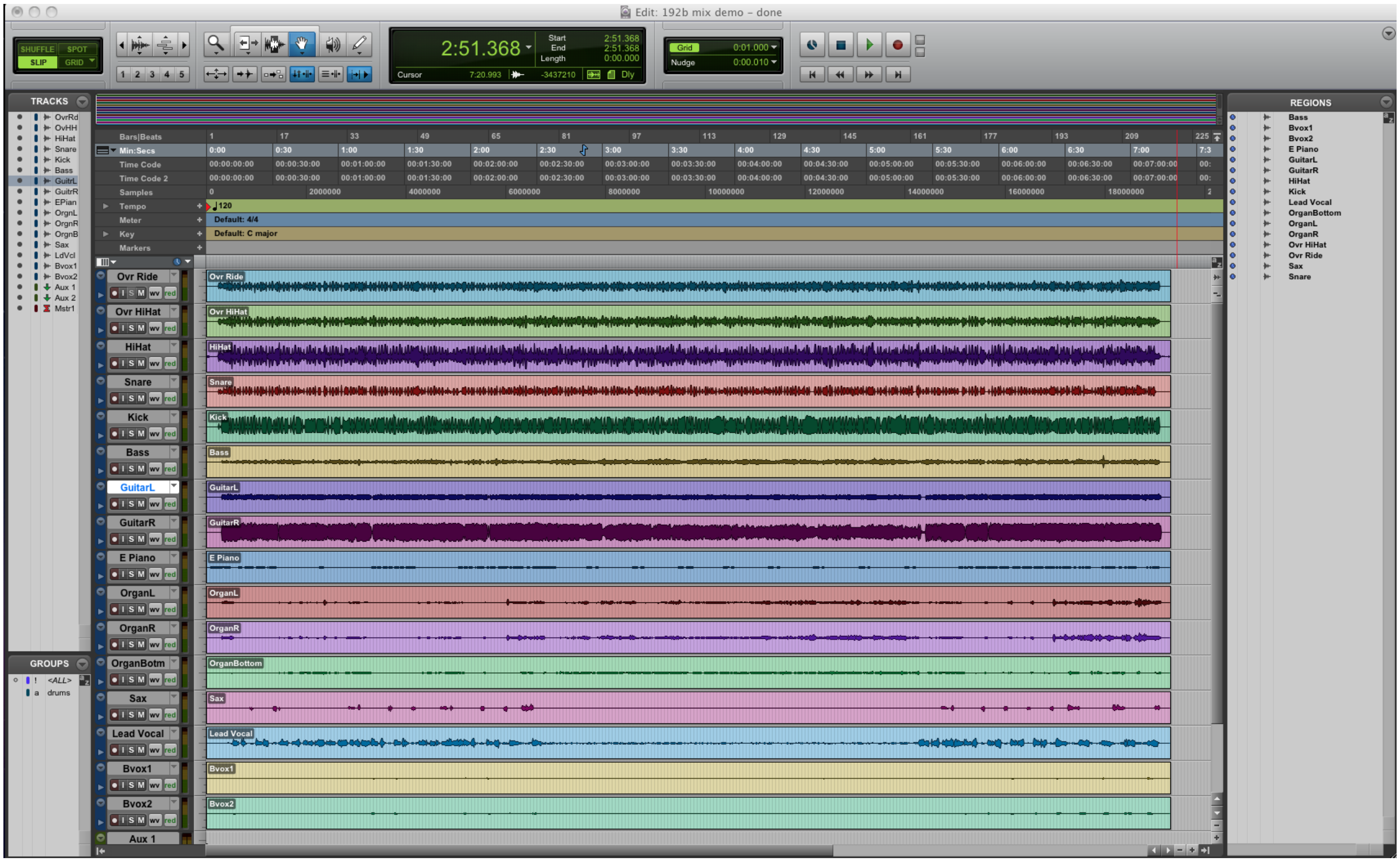
The recording studio traditionally consisted of a large collection of hardware devices that were necessary to record, mix and process audio. That paradigm persisted until the computer began to take over the individual tasks one by one until we are now able to recreate the studio equipment exclusively inside the computer. This has resulted in many changes in how music is recorded, but at its core the process is essentially similar to the original methods. The beginning and end of the process still involves analog systems, the microphones and loudspeakers, neither of which can be accomplished by a computer. Between those events, however, we now employ software running on general-purpose computers to perform the actions formerly requiring specialized hardware. It is not surprising, then, that the graphical interface presented by the software resembles the hardware formerly required. Cynical old-timers have referred to the process as "mixing with pictures" of the equipment.

There is no doubt that the transition from hardware to software has greatly augmented the availability of the tools necessary to produce high-quality audio recordings. At first, the sound quality was compromised because of limitations caused by the limited computing power available from the computer. External processing was necessary, with audio signal processing accomplished by add-on computer hardware using digital signal processing (DSP) chips. Eventually, the average personal computer improved to the point that the DSP hardware became unnecessary, at which time the transition to digital audio workstations really accelerated. Today, there are many choices of software systems that all accomplish the same basic tasks, recording, playing and mixing multiple tracks of audio. Choosing one involves personal preference to some extent, but if one intends to share work with other studios there are some platforms that are more common than others. We will examine the various choices but all of these programs do pretty much the same thing.

One part of the process of creating music that was limited in the era of tape-based recording is editing. Sure, analog tapes can be edited by physical cut-and-paste techniques and engineers got very good at doing so, but there is a limit on this because all tracks are stuck together in time and one could only edit them all together. With the transition to digital recording, each individual track was now separate and could be edited independently. This might be one of the most significant differences between the old and new recording methods. Most DAWs use graphical representations of the audio data and you can look at the data while you cut and paste. Since the image used to find edit points is simply a map telling the computer where in the audio file you wish to edit, the original data is not permanently altered and the edit can be undone, something impossible when cutting tape. The non-destructive edit is another of the great improvements allowed by the digital audio workstation.

In addition to non-destructive editing, DAWs allow non-destructive overdubbing and punching-in as the tracks are recorded. With analog tape, the number of tracks was limited and adding corrections to previously recorded audio meant over-writing the original data and any mistakes were permanent. With the digital system, one could record as many tracks as allowed by the software and new takes no longer required deleting old ones. Punching-in could be done over and over without eliminating the original sound recordings. Little wonder the DAW so rapidly became the preferred recording system.

Since the DAW performs the same basic functions as the traditional recording studio, they tend to model their user interface after the hardware model. The functions of recording/editing and mixing are generally represented on two separate windows. The recording/editing window displays audio waveforms as they are recorded and allows graphical editing of the waveforms after they are acquired. The mixer window display resembles a hardware mixer with faders, panning, routing and signal processing. Most DAWs use this representation and the are in many ways similar. We will look at several of the DAW windows to get an idea how they compare.

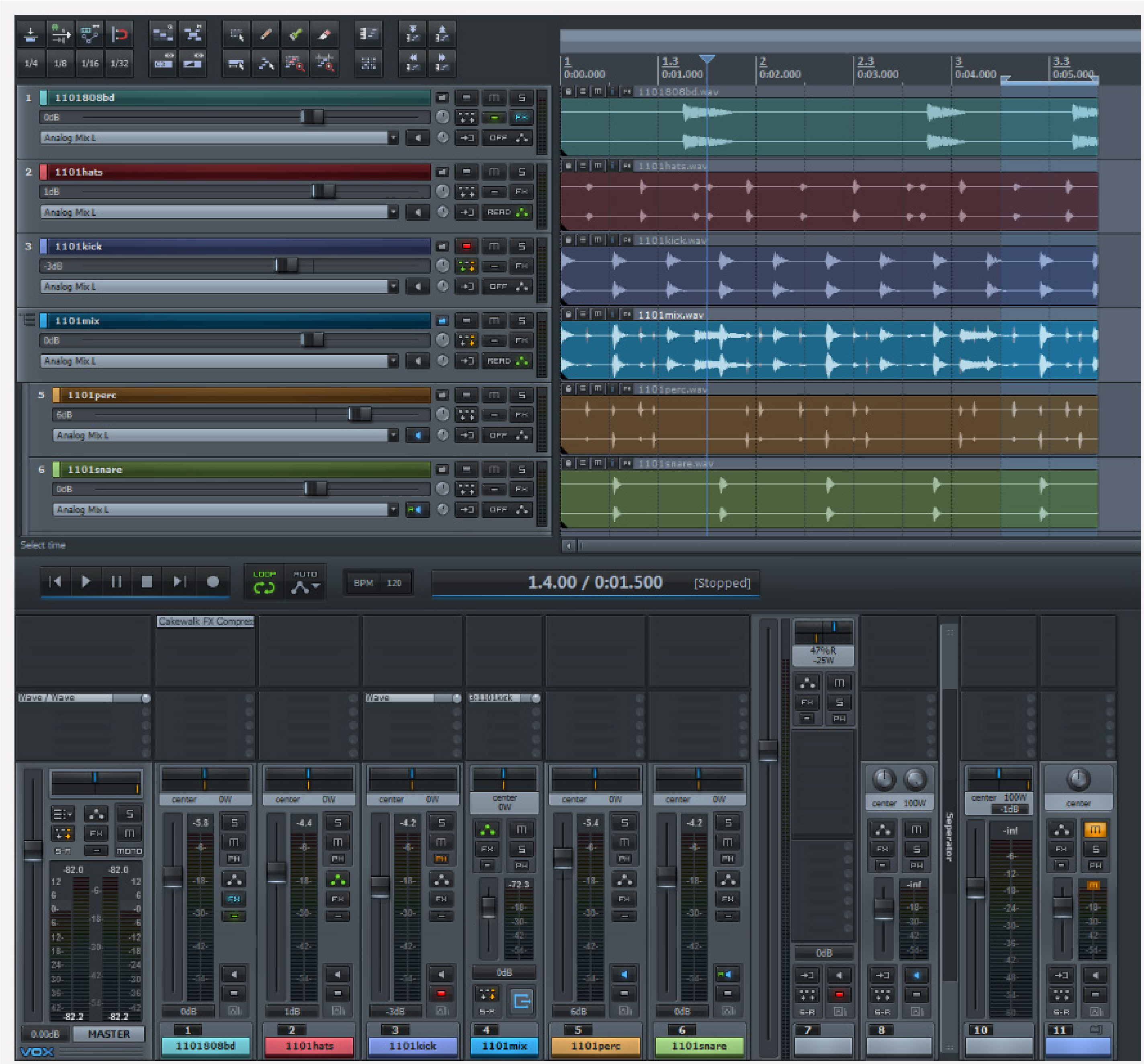


Pro Tools Edit Window

Pro Tools Mix Window

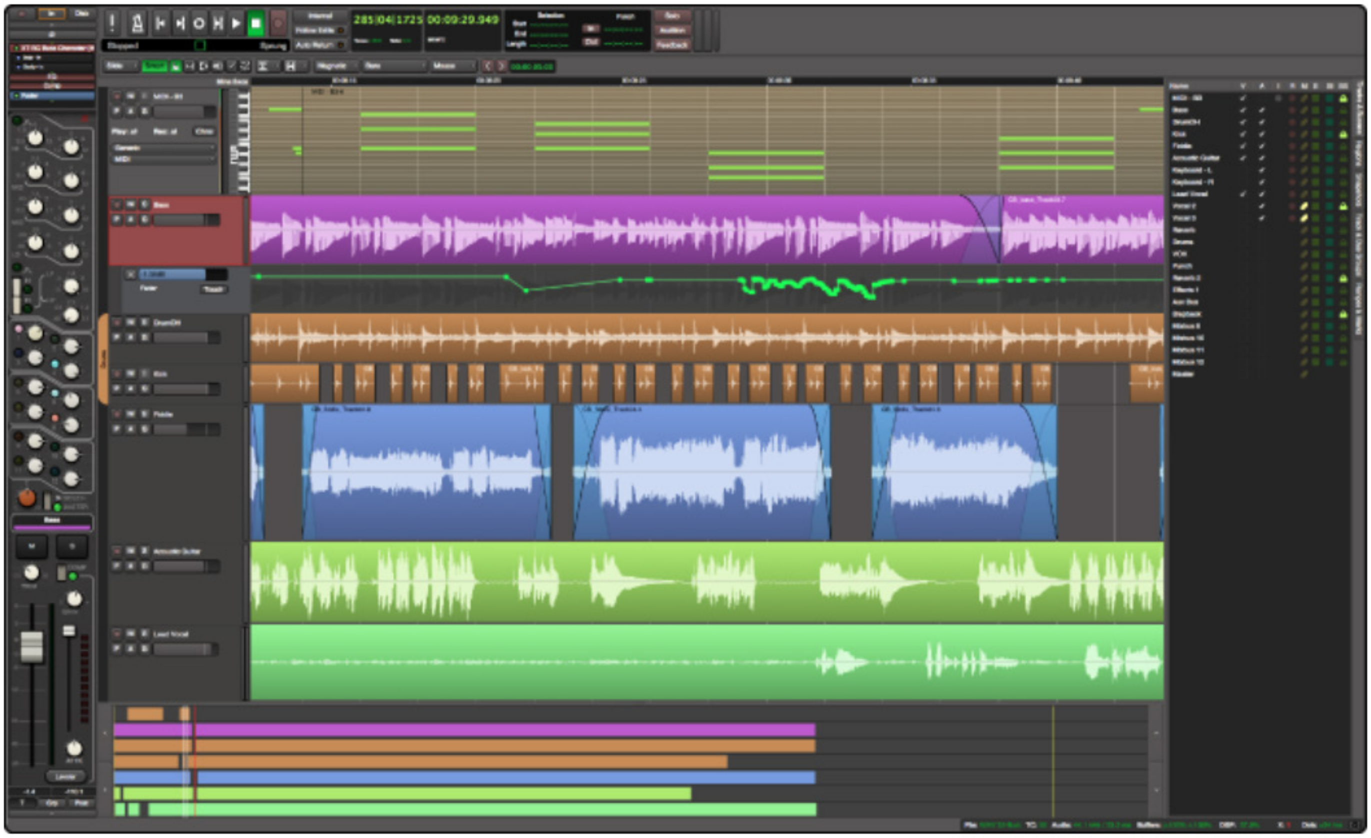
Logic Arrange Window

Logic displays two sets of windows in the Arrange window, the lower of which can be selected between functions including mixer and destructive sample processing. Windows can be separated into free-floating windows if there is sufficient monitor space. The Arrange display can be customized.



Reaper window

Reaper is heavily customizable, with downloadable “themes” that change the look of the interface. You can make your own!



MixBus Edit window

MixBus Mixer window

MixBus is unique in that it is essentially a "front end" for Ardour, an open-source recording program. (Ardour is used at CCRMA for Ambisonic playback as it supports multi-channel busses and panning via Ambisonics through plug-ins.) MixBus is a product of Harrison Systems, a manufacturer of large, high-end mixing consoles. MixBus and the more advanced MixBus32C bring the sound of their hardware consoles to the DAW. As with the other programs, third-party plug-in processing is possible but their built-in and add-on plug-ins are designed to sound like their hardware consoles, which are known from famous recordings such as Michael Jackson's "Thriller". MixBus is one of the only DAWs that runs on all three of the most popular computer operating systems, Windows, MacOS and Linux.

General Characteristics of DAWs

It is apparent that most DAWs depend on the same basic paradigm - the analog recording studio. While the particulars vary, having famiarity with one system generally prepares an engineer to use any of the platforms without entirely learning a new workflow.

One central element of these programs is the edit decision list: the mapping between the graphical display and the underlying sound files. This allows non-destructive editing using a graphic representation of the recorded sounds. Individual tracks can be moved in time and cut up and re-assembled without changing the underlying sound files since the edit list simply determines how the data is played back. This also allows edits to be cross-faded since the files still exist before and after selected edit points. The mixer function allows control of the incoming sounds from the A/D converters. It also allows us to select what we hear in the process. Common mixer functions like mute, solo, pan and gain settings are clearly represented in all of the programs.

A common limitation in software designed for recording is caused by the use of the mouse to control the system. Only one operation at a time can be conducted with a mouse while hardware mixers can theoretically change every parameter simultaneously. There are hardware control surface devices available to control software through MIDI, but there are sometimes issues with communication between these devices and the software, due in part to MIDI time delays and in part to incomplete implementation of the protocols used for communication. These control surfaces can make the recording and mixing processes easier when everything is working.

Integrating Audio With The Personal Computer

Using a personal computer as the core of a recording system brings inexpensive power to the table, but these machines are not optimized nor are they intended to be recording studios. The external ports available are intended to connect devices like printers and scanners, not audio interfaces. This often requires specialized software called "drivers" be added to the operating system to allow access to the amount of audio data provided by external audio interfaces. These interfaces must perform analog-to-digital conversion on numerous separate channels and deliver that data quickly to the operating system. Since each operating system handles input from the standard interface ports differently, drivers must often be provided by the interface manufacturer to optimize the performance of the overall system. In the past, certain faster computer interfaces (SCSI, FireWire) were required for audio as the standard ones were not fast enough. Now, the ubiquitous USB interface is fast enough to handle significant data throughput and many audio interfaces use these ports. They are now often standardized through what's know as "class compliance" so that devices can operate using drivers that are included in the operating systems.

The latest development regarding computer/interface connection makes us of the Ethernet protocol that is in general use for computer networking. By tailoring the higher-level software associated with general computer network comunication, it is possible to send large amounts of audio data over existing networks. There are both proprietary and open-source approaches to this. Dante is an example of a proprietary protocol implementation while AVB (Audio Video Bridging) is a standard protocol available to everyone. The advantage of these systems is the ability to connect over existing computer networks. Routers must be AVB compliant in the case of AVB while standard Ethernet networks and switches work with Dante. Ethernet-based audio allows a single CAT-6 cable to replace heavy copper-wire snakes to connect locations like the stage and a studio or front-of-house mixing console.

When personal computers were first used for audio, they needed additional processing power because the CPU was not fast enough to handle the demands. Dedicated hardware was added to the computer using DSP chips on circuit boards that were added to the main bus inside the computer. Pro Tools pioneered the development of these boards and optimized the processing by adding their own bus between cards to reduce the demands on the computer's main bus, allowing faster interchange of audio data. As computer hardware has developed, it is now more than fast enough to handle not only the transfer of data but also the computation necessary to perform signal processing and graphic display. This has reduced both the complexity of necessary audio hardware and the cost.