Free Software Audio Applications for Audio Playback, Recording, Editing, Production and Radio Broadcast Management and Automation

An Overview of Functionality and Usability

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Acknowledgements:

I am grateful to free software community and numerous developers who dedicated a lot of their time for the development of free audio software. Their patience, resilience, knowledge and dedication to freedom kept me working and evaluating their work in order to suggest features, bug fixing and writing documentation. I am grateful to Stephanie Hankey from the Tactical Technology Collective for encouraging me to write this scan in order to contribute to strategical involvement in helping media organisations, advocacy groups and various professionals to enjoy and practise their freedom by using free software. She patiently edited my text and added clarity where needed.

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Introduction

Audio applications are well developed in the world of non-free software. Development of those applications have been fostered by market requirements and increasing popularity of multimedia activities. As it happens in the market some vendors are defunct despite that fact that they have played pioneering role in the world of audio free software. On the other hand there is rapid increase of development of free audio software. In the world of free software there are dozens of .mp3, .ogg and CD players. Some of them outperform equivalent non-free packages. As far as software for audio recording is concerned some free-software packages are still lagging behind non-free counterparts. In addition, our culture is characterized by visual interaction, while free software is in majority of cases characterized by various forms of texts that require structured and systemic approach to interaction with software itself. Actually, free software still does have that mark on its skin: it is hard to manage. We can nowadays say that that attitude is in majority of cases false. But, consequences of that attitude caused skepticism about its functionality in the world of multimedia and professional audio and video applications. Those who dedicated their time to accept textual interface, adopt themselves to format of so-called man pages are keen to declare their sympathies to free software like funny donkey to famous ogre Shrek: “He is ugly 24/7.” Truly, it will be unjust to say that free-software is exclusively text-only based and that it is not always user-friendly and eye-candy. In many cases it is much nicer that non-free counterparts. But, on the other hand there are few packages that are functionally completed so that they may be used fully functionally and replace non-free software. Thus, proper overview of those packages may be useful to those who do not have enough time, knowledge, skills and courage to check dozens of packages and look for the right one that suits needs of that user. This, document is therefore aimed to those who consider themselves as audio enthusiasts being involved in various artistic, cultural or civil society development projects. However, professionals working for radio stations, news agencies or in organizations where audio editing and processing is an important activity may find this document useful in order to plan migration to free software. Since there are attempts to provide users with integrated free software audio solutions, system integrators may find this document useful to consider what target groups they may address to. In addition, various development agencies and donors may find this document useful when planning their program strategies to aid various organizations and individuals on the territory of their operation.

Indeed, computer audio applications are omnipresent in many fields of everyday work. Advocacy work of NGOs, media, arts and culture, ecology, health, work with children and youth, education are just some application fields where audio applications are being used often. Hence, people from many fields may need free audio applications and consult this document in order to undertake training efforts and become equipped with powerful audio applications spiced up with community communication, support and development. However, contribution to community work requires patience and respect towards community members, curiosity for the matter concerned, skills to present clearly one's needs and requests, wisdom of handling various temperaments and cultural characteristics.

Audio Tools and Free Software

Although, audio applications look very intuitive and simple to use, professional work requires more knowledge about technical areas, such as: digital signal, digital signal processing, digitization of analog signal, basics of audio mixing and audio production and post-production. Often people use or try to produce their audio recordings and create playlists, but they find that the level of sound they captured may not be always as they would like. Sometimes, various glitches, clicks and pops decrease the natural color of sound and therefore don't meet the expectations of the application users.

We have learned from the expectations of professionals and audio enthusiasts that audio applications should be precise, stable, easy to work with, capable to visualize data, inter-connectable, capable to handle various uncompressed and compressed audio data formats. Thus, the development of audio applications is not an easy task. Audio applications should meet one specific requirement: to be able to communicate with various hardware based on standards, i.e. PCI, USB, IEEE1394, MIDI and other standards and protocols. Although recommendations for audio protocols, standards and application guidelines are recommended by AES (Audio Engineering Society) many commercial manufacturers of audio hardware do not strictly follow those recommendations and guidelines. Some manufacturers manufacture their studio equipment and software by using their own proprietary formats such as ADAT by Alesis Inc, T/DIF by Tascam, YGDAI by Yamaha etc. Many of them use proprietary formats since they are trying to be unique by their technical specifications in a highly lucrative market. In addition, audio industry is in some ways strange one. It is hard to define list of technical standards that are exactly defined, evidenced and measured by some certified equipment. A field of audio technical specifications is rather bewildering forest with sooty paths. Consequently, commercial hardware and software manufacturers often communicate with their users by communication policies established rather by their marketing departments than by research and development labs. Democratization of innovation is not their field of excellence. User protection organizations are confronted with the fact that some manufacturers print on their packages false technical specifications or that they do advocate truthful communication with customers until their sales staff overtake their hardware and software designers and start to manipulate users. Thus, those who are more familiar with non-free audio software should not worry too much. A field of uncertainty is already under their feet. Entering into the world of free software increase chances that technical merit will become predominant since software source is open for critique and discrepancies may be discovered easily. However, the world of audio hardware is still dominated by manufacturers that do not pay much intention to the education of their users. However, a rare exceptions are companies Rane and D & R which are getting closer to the world of freedom of innovation. Still, the world of audio equipment is characterized by self-indulgent production concepts. Consequently, the issue of interconnectivity with existing equipment is still rather problematic. However, interconnectivity with devices of other manufacturers is not always of the highest concern. Due to the fact that the audio field is a very commercial business many manufacturers have developed their own applications based on the current entertainment or music genre trends. In contrast, the increase in the number of manufacturers as well as the knowledge available to people has increased the competition in a very limited market. The increased availability of software packages and the appearance of free software audio packages has started to shape a new scene on the field of audio software. Commercial/non-free software manufacturers are facing financial difficulties and the major software manufacturers are being acquired by larger corporations such as Apple, Yamaha and Sony. On the other hand, the fast development of hardware with semi-professional and professional technical specifications is an ongoing process and more and more manufacturers are opening up specifications of their chipsets to driver developers in order to make their products available to a larger number of customers. The fast development of personal computers has fostered the development of audio applications and increased need for control from publishing and producing associations and corporations. A need to impose absolute control on audio file formats, broadcast, file sharing is still a threat for the freedom of software in the audio field.

Some manufacturers such as Cirrus Logic Inc. have invented new ways of transmitting audio signal by using Ethernet. That technology called CobraNet is currently the most promising concept; instead of traditional audio input and output connectors it is possible to use an Ethernet interface, thus enabling users to deploy various forms of networking including multiplexing, broadcast, multicast, blogging etc. That technology may be a challenge to developers of open hardware to create their own audio/Ethernet devices. A good example of manufacturers that manufacture audio/Ethernet computer audio cards is AudioScience Inc. they are manufacturing PCI audio cards and other audio interfaces with Ethernet interface providing users with GNU/Linux drivers.

Audio card equipped
with Ethernet connector

As far as the development of free software is concerned, we can say that there is a plenty of free software audio packages free to download from various software repositories. Only a few of them are capable of performing complex audio tasks. Some of them require prior knowledge of sound synthesis, scripting and basics of programming whereas others have a full graphic user interface (GUI). A presence of GUI is not the exclusive criteria, but in a complex world of audio it is highly desirable, since some complex operations require simultaneous control over many important elements such as cueing in the playlist, audio signal level, transfer of files over local area network etc. Thus there a few that may be used easily by the average or beginner user. However, this does not mean that other applications that are much more complex and not easy to grasp at the first moment are not useful or powerful. Even contrary, they may be very powerful, but not quite useful for the beginner that is not familiar the application specific way of using it. For example, CSound is powerful sound synthesis application used in some very demanding environments such as in movie and advertisement industries. Ecasound is powerful multi-track sound recording software that may function nicely on computers with weaker technical specifications. But, average user is not expected to discover all their potentials so easily. However, some work and study or guidance may be needed in the beginning.

The focus of this scan will be on audio applications that are relatively easy to interact with for users who are not necessarily familiar with intensive scripting, complex parametric definitions and controls or programming.

Audio applications are often made of two layers: the *core layer* and the *presentation layer*. The core layer is made of an *audio engine;* the audio engine is the part of the application that is responsible for audio processing) and sometimes the relational database management system (*RDBMS). The RDBMS is the* part of the software that retrieves stores audio files from and to the relational database (such as *MySQL*) and there is also an *audio storage core*. This audio storage core takes care control of various audio storage systems The *Presentation layer* is actually the part of the application that enables us to interact with it. The majority of audio applications have an audio engine and presentation layer only. However, uses may combine several applications for demanding tasks, for example, interaction with a relational database and versioning system may be desirable in complex environments. Professional artists, scientists, radio program producers and audio engineers often require databases (ie. *MySQL* or *Postgres*) and/or versioning software (i.e. *CVS* or *Subversion*) for conducting their work professionally. Some software packages for radio station automation require *MySQL* installed and configured prior to installation of playback automation software. Such an approach requires that users have basic skills in database management and versioning systems. This is not so much hard to do as it is time and energy consuming to handle a vast amount of data without those basic skills.

Types of free software audio applications

The workflow of media institutions can be divided in to various stages. Firstly, it is needed to record certain audio files, edit them and in some cases do mixing with other audio files. Those files may be stored in a database of songs, interviews, sound effects, completed mixdowns. Such mixdowns may be broadcasted or recorded on CDs, DVDs, directly from the application itself or by using other applications specially designed for that purpose. Although for example, an application *Rezound* is equipped with GUI access and management of CD burning programs, there are many other CD burning frontends that can be used for easier CD and DVD burning for distribution or archiving purposes.

Despite the fact that there has been a rush by free software developers to increase the number of features in their applications, free software audio applications may be divided in the following categories:

Audio recording and editing
Multichannel mixing and recording
Inter-application connectivity
Playback, streaming and broadcast automation (playlists, radio cart software, interaction with telephone hybrids)

In truth the functionality of the majority of free software audio applications overlap. But, some of them are specifically designed for certain purposes. For example, radio software packages do have recording capability but recording and editing may be better done in an applications that are specifically designed for that purpose i.e *Rezound* or *Audacity*. Despite the fact that *Rezound* and *Audacity* have the capability to record multichannel sound files, multichannel recording and editing is more often done in *Ardour or Rosegarden.*

Choosing an application for audio recording and editing

Although there are plenty of audio recording and editing software packages such as *Snd, Rezound, Audacity* and others, many average users choose their favorite application from how obvious, comprehensive and manageable the application's interface is, how stable the application is, if the application produces unwanted sound artifacts (clicks, pops, glitches), how precisely it performs and which audio file formats it imports and exports to.

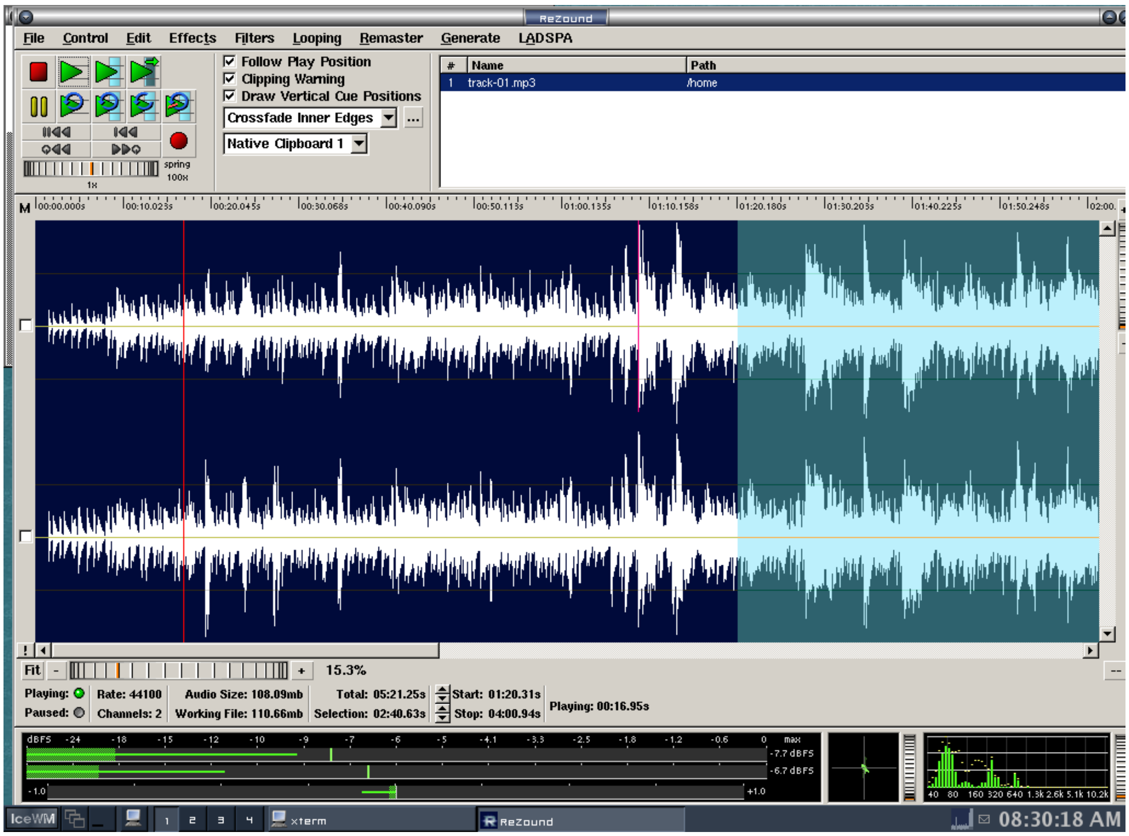
Some audio recording and editing applications such as *Snd* are more sophisticated. *Snd* does have a built in listener that is capable of listening to commands from various programming languages that can give unique capabilities, however the average user is unfortunately not capable of using its capabilities to the maximum . Thus, due to ease of use, simplicity and sufficient features, the two most frequently used audio recording and editing free software audio applications are *Rezound* and *Audacity*. Experienced sound engineers will always remind any audio enthusiast that the basic rule of digital audio recording is :"Trash in – trash out". This rule says that the quality of recording is the essential point in the management of audio recordings. There is no software which may fully recover badly recorded sound. The role of *Rezound, Audacity, Snd* and similar software packages is not to successfully repair very bad recordings, but rather to edit properly recorded audio files. Despite the fact that some applications are equipped with tools that may remove various noises from audio recordings such as vinyl recording crackles, audio cassette hissing, some pops and clicks and some sound artifacts caused by keeping inappropriate distance of speaker from the microphone, the basic purpose of audio recording and editing applications is not to create productions from poor quality original recordings. Hence, when doing advocacy work, or preparing audio material for broadcast we have to keep in mind that the person who is recording the audio material should acquire at least basic skills in audio recording or the original audio used should be of good quality. Numerous manuals on the Internet and nearby bookstores may be helpful in providing you with information necessary to acquire knowledge and skills for properly choosing a microphone, sound card supported in GNU/Linux, and other audio hardware that may be needed for audio recording in broadcast or various outdoor or indoor environments. Some suggestions can be found at the end of this paper in the resources section.

Gaining more professional results may be achieved by selecting computer audio card with professional technical specifications that is supported in GNU/Linux, using proper microphones, microphone preamplifiers, cables, connectors. Indeed, it is needed to acquire certain recording skills and knowledge on physics of sound, functions of audio recording and editing equipment. Many publications and training centers may be found on the Internet. However, the Internet site of the Rane corporation is the real source of no-compromise approach to the users. Their technical notes are the real source of knowledge that anyone involved in audio recording and editing should be familiar with. A level of professionalism is always gained by experience and keeping intact your hearing capabilities. “Louder is not always better”, the first lesson of audio recording. “User is recording sound, not graphical representation of file on user's computer monitor”, the second lesson. “Chain is strong as it is strong its weakest link”, the third lesson.

Audio recording requires, not only technical knowledge, but some talent and feeling for subtle differences. Thus, skills for professional recording of interviews, speech in some cases are not too hard to acquire. Professional recording of music is different thing and it requires a lot of skills including knowledge of music, musical instruments, music composing styles, acoustics etc. High quality recording for teaching, training, reporting may be done with modest budget equipment and some skills of audio editing and recording. Use of audio free software may be quite helpful in gaining very good results on the user's personal computer. Its freedom is promising concept from the perspective of technical merit since there is no marketing department to hide some features and innovation is democratized so everyone can contribute with his/her own knowledge and skills. That is phenomena that is unfortunately not often present in commercial companies. Thus, it is expected that the future of innovation and technical merit is on the side of free software.

Rezound

Rezound is a free software application that can be frequently used by professionals and amateurs alike. The latest versions are stable and equipped with powerful sound editing tools suitable for successful audio editing.

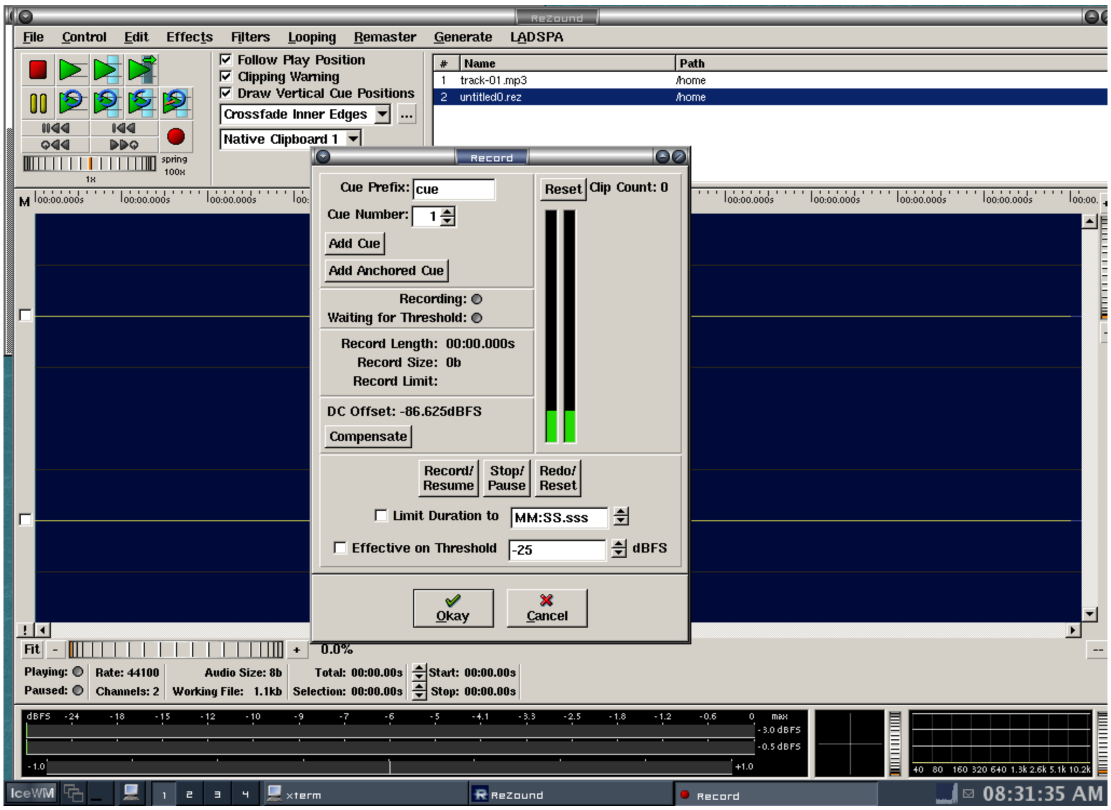


Rezound is full featured, easy to use and stable free software audio application

Rezound is an application that MS Windows users would most easily compare with the popular non-free *Sound Forge* package. The interface is intuitive and easy to use, but the most important thing about *Rezound* is its fully featured audio recording package capable of performing professional recording in a way that meets professional resolution, sample rate and audio file format standards. The display of audio files is done in an user-friendly manner so that the user may select part of the file, cut it, copy or apply various operations to it with ease. *Rezound* was tested for this overview[1] on weaker and powerful desktop computers as well as on laptop. It is possible to start it via desktop panels and shortcuts as well as from console simply by typing: *rezound* and pressing Enter.

1For example, I tried it also in *IceWM* environment on Compaq Armada E500, 600MHz CPU laptop with 128MB RAM, Red Hat 8.0 GNU/Linux distribution and it run smoothly.

It is very much a standalone application and it runs smoothly over graphical environments such as, *GNOME, KDE* and *Window Maker.* When capturing and editing audio 450 or 500MHZ with 128M RAM and 4GB HD is the bottom line configuration that may perform reliable sound recording. Although sound recording is possible on weaker configurations it is strongly recommended to use it on the bottom line or better configurations. *Rezound* is easy to install via using apt-get tool that is implemented in Debian, RedHat, Fedora and some other GNU/Linux distributions too. It is enough to open your console connect to



Audio recording in rezound is easy procedure

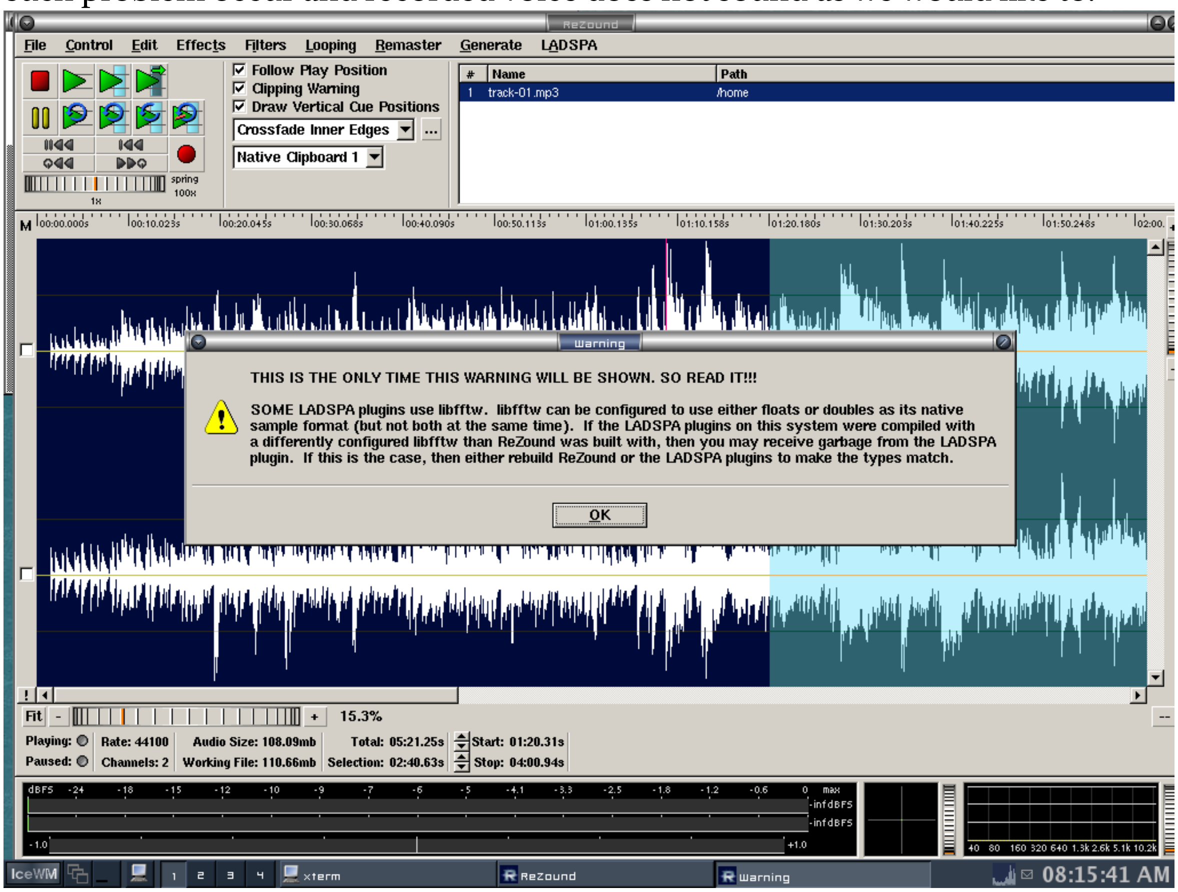
the Internet and type: *apt-get install rezound* and press Enter. Voila! After that *Rezound* will be downloaded and installed on user's system. User does not need to worry on other technical preconditions since apt-get will take care on them. It is considered to be easy to install it on all major GNU/Linux distributions Slackware, Debian, Mandriva, Red Hat, Fedora etc. There is not yet version for Windows and Mac OS. However, it is ported to FreeBSD and Sun's Solaris so those who are more in favor of BSD and Solaris concept may use it too. Free BSD is an free operating system that emerged from BSD operating system originated in early seventies. It is very stable, ready for heavy duty environments such as Yahoo. It comes with KDE, Gnome, Windowmaker and other graphical environments as in GNU/Linux. Since it is equipped with GNU/Linux compatibility features many GNU/Linux software packages may operate in FreeBSD. Solaris is non-free operating system manufactured by the Sun

Microsystems. It is widely used in server and heavy duty enterprise environments. Although it is not free as other free software it is important to say that Sun Microsystems is a company that is very much friendly towards the free software community.
Rezound can open, display and save to majority of audio file formats including wav , aiff, au formats often used on PCs/Windows and it can open compressed files too. On the bottom it shows a peak meter, small but handy frequency presence and phase display. Recording itself may be started by pressing the red recording button on the top-left part of the display. The user is prompted with the ability to define type of file, resolution, sample rate and other important parameters for recording. The process of recording is easy to manage and display, the recorded file may be zoomed in or out in order to do precise editing. The recorded file may be easily manipulated and cueing may be done too which is quite a useful capability. A variety of controls and filters are present that are more than sufficient in using *Rezound* in media environments, as long as the user has the basic skills of audio recording, microphone positioning and digital signal processing practise. Even, some filters and other controls being provided in *Rezound* may be suitable for scientific work rather than work in media organisations.

Rezound and LADSPA plug-ins

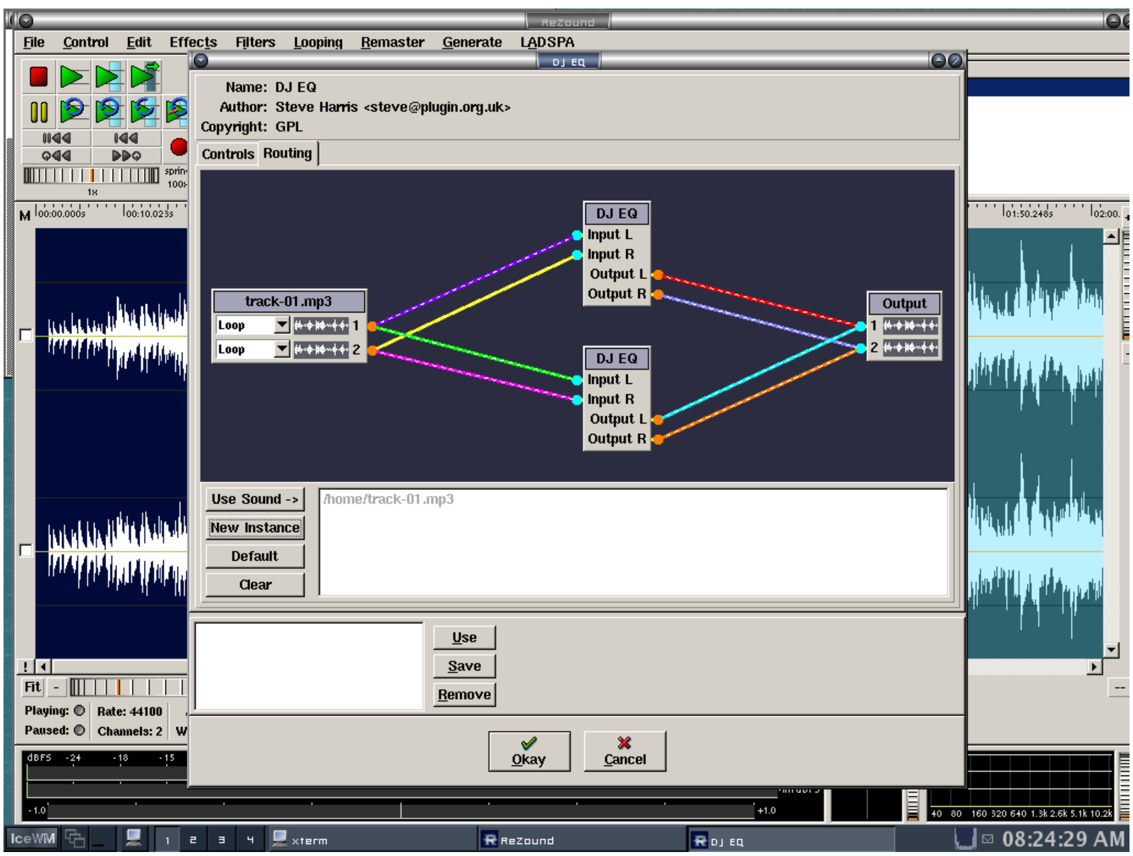
A very important part of the design of *Rezound* is that it is capable to interact directly with *LADSPA* plug-ins. *LADSPA* stands for Linux Audio Digital Signal Plugin Architecture. This, makes dozens of audio processing plug-ins available. Though, some plug-ins are not useful for media organisations, some such as *Noise Gate* and other dynamic processors are quite useful when journalists or civic advocacy activists are preparing their interviews and recordings for broadcast and various ways of reporting. For example, someone may record an interview and notice that there are some environmental sounds in the background. Noise Gate may be defined in a way that gate is open for sounds louder above certain threshold. Thus, sounds below that threshold will not be heard and recording will sound cleaner.
Though *LADSPA* is a very important project it is not standardized and some plug-ins work better than others. *LADSPA* is actually a huge collection of tools that enable user to rectify, modify and adjust his/her recordings. For example, if user does have recording where the main voice is covered by louder voices it may be necessary to compress voices that are louder above the loudness of voice that is more needed to listen. It is needed to set threshold above which louder voice will be compressed by let say ratio of 3:1 thus enabling the less loud voice to surface. Or, user may often change spaces in which recordings may be done. Since each room does have its own form, it is made of various materials ad therefore it does absorb or reflect low, middle or high frequencies differently. It is hard to do proper recording in such environments without unwanted sound artifacts. One of *LADSPA* tools is equalizer which enable user to increase or

decrease presence of certain frequencies so the characteristics of absorption and reflection of sound in certain space may be corrected. That is also important when users record sound in environments which does have change temperature and humidity such as in open space. Due to changes of temperature and humidity the speed and absorption of sound is being changed thus changing its characteristic. Those who record in various environments such as villages, private houses etc. will often need to use this tool. Use of *LADSPA* equalizer tool may correct those sound variations. In various occasions people involved in sound recording do not have time and other technical conditions to organize careful and precise microphone positioning. However, they may not need studio quality recording, but it is much easier and pleasant to listen voice that is recorded properly. However, some of *LADSPA* plug-ins may be helpful when such problem occur and recorded voice does not sound as we would like to.



use of LADSPA does have some limitations

For example, recorded voice may sound somehow dry and it is possible to make it more pleasant by adding so called acoustical effects so the voice sounds much more natural and pleasant. Despite the limitations of working with *LADSPA* plug-ins, audio processing tools provided are very powerful and easy to control. Although some plug-ins need more precision and stability the majority of them are quite useful and may foster the creative use of software. It is important to say that some plug-ins for other platforms are very expensive, not sufficiently precise and can even produce disturbing artifacts. Some proprietary plug-ins are priced much higher than sound recording applications themselves. Although there is possibility to apply some proprietary plug-ins, such as, *VST* plug-ins within GNU/Linux environment, *LADSPA* plug-ins are sufficient quality and easy enough to use. In reality, people do need to use three or four *LADSPA* plug-ins, but it may be worth for education, training and scientific work to use some others too. Compressor, noise gate, equalizer, limiter, echo and reverb will be the most often used. Their functions are in a way self-explainable and user with a couple of hours of exercise may be accustomed to their functions. However, it is important that user is concentrated on sound and listening rather than on "intuitive visual guessing" that is often preventing users to think about what they really need to achieve. Making users interaction with computer systematic and highly conscious is very important prerequisite for successful and creative usage.



LADSPA DJEQ plug-in routing tab in action

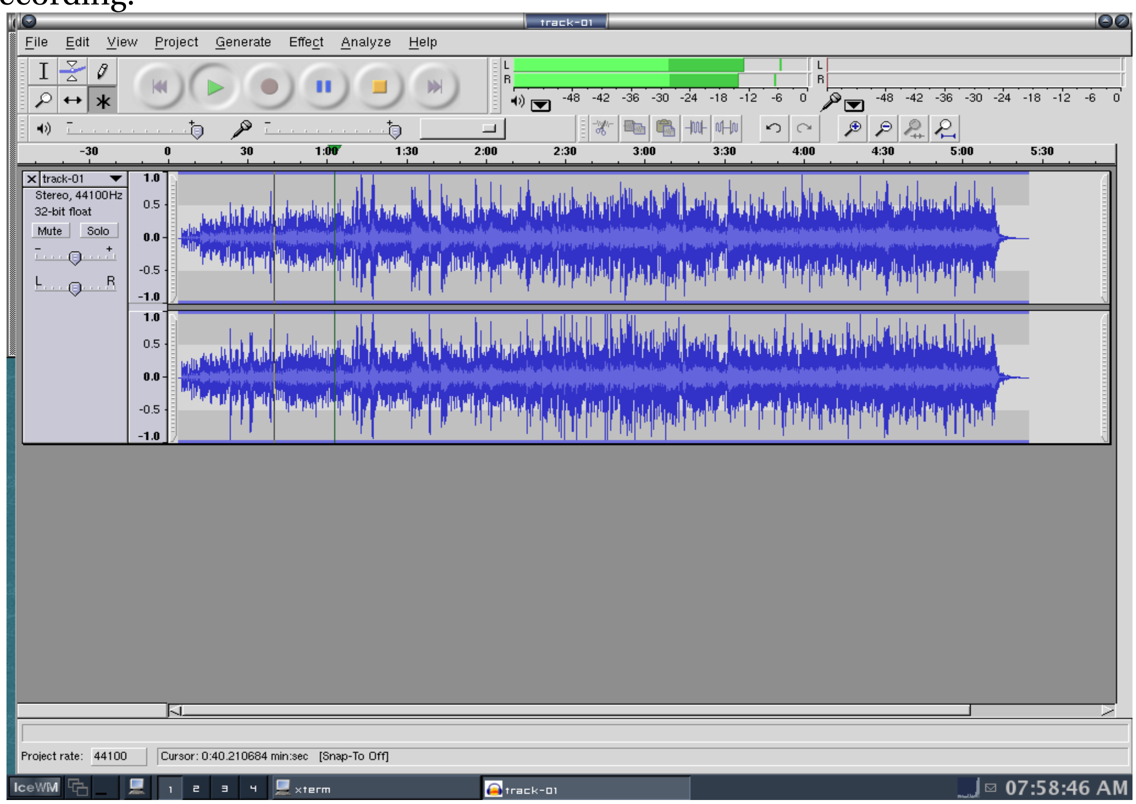
Graphical representation of *LADSPA* plug-ins is usually made of two parts. Firstly, there is a tabbed display which enables the user to change parameters by typing them in required fields or moving appropriate faders, checking checkboxes etc. and secondly, a separate tab to define routing of the signal as it is presented on the above image. The use of some filters require a wider knowledge of digital signal processing and related disciplines, but the majority of plug-ins may be used in a straightforward manner efficiently in audio editing activities. Since *LADSPA* is an ongoing project, it is expected that the number and quality of plug-ins will rapidly increase in the future so the users will be equipped with powerful tools for audio processing on a professional level. Although many experienced sound engineers use to say that the best effect is the one which is not easy to notice in the recording, it may be quite useful to have handy a variety of effects and other processing tools that may assist the user in achieving the desired results. Additional inter-application connectivity via a tool called *Jack* increases the number of possibilities for the user in polishing up or transforming audio recording in a creative way. Thus, the freedom of the software may serve directly for the implementation of freedom of speech and expression in some media organizations. Efficiency of the use of *Rezound* would be significantly increased when the proper documentation will be written including documentation for *LADSPA* plug-ins.

Audacity

Audacity is a relatively young application, but we can say that it is a fully featured audio recording and editing application. Some developers and contributors such as Phil Burk and Roger Dannenberg are experienced programmers for digital audio processing and with Unix . Thanks to this and the contributions of other developers *Audacity* has become a very well developed and stable application in a very short period of time. Due to its freedom there is a modified version with the possibility to use drivers for Audio Science Inc. audio cards that are equipped with an Ethernet connector. Although the interface is not so much eye-candy as many non-free applications, its layout is obvious and organization of the controls is well done, so the user migrating from non-free audio applications should not have problems interacting with *Audacity*. *Audacity* is a cross-platform digital audio application, so it may be used on various platforms enabling the user to migrate easier to free operating systems afterwards. It is available for Windows, Mac OSX. Those who prefer FreeBSD operating system may use port of *Audacity* for FreeBSD, since FreeBSD is equipped with GNU/Linux software binary compatibility feature so GNU/Linux software packages may operate on FreeBSD that runs on user's personal computer.

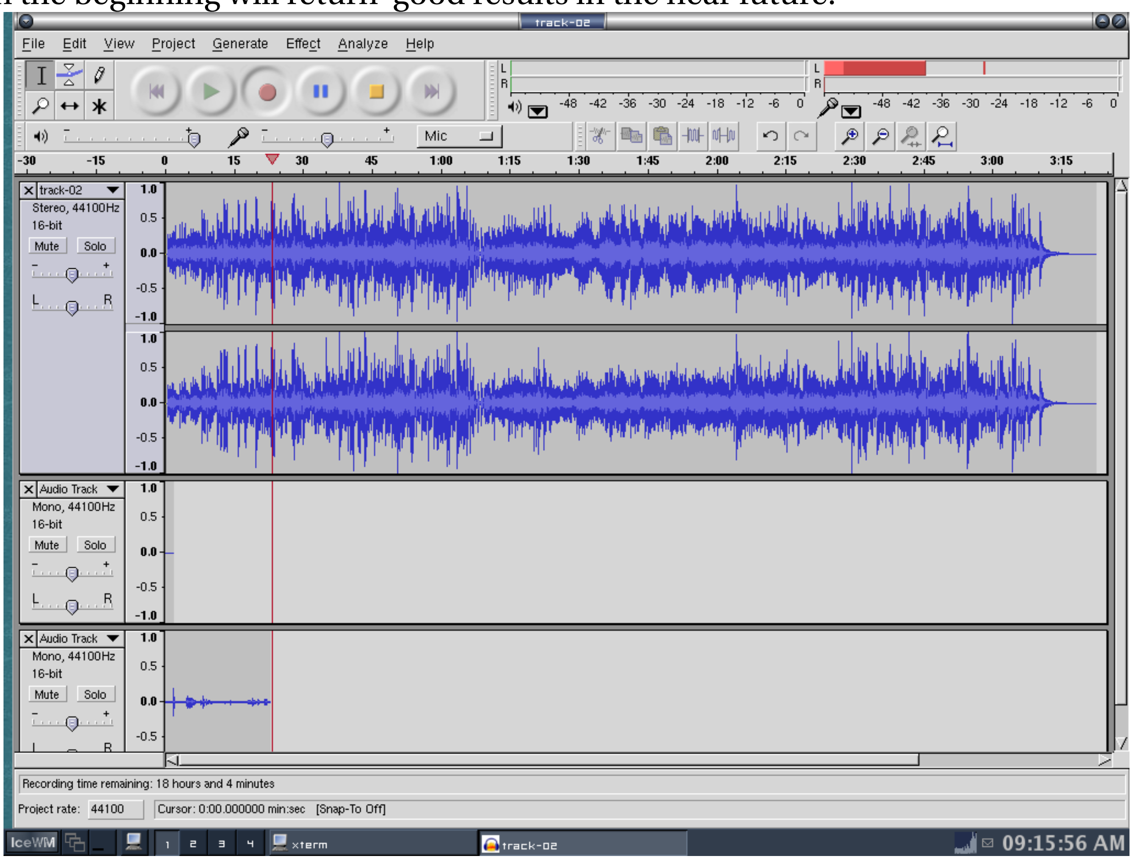
Despite the fact that there are some areas where Audacity is behind proprietary software, such as with the interface, it is important to emphasize that this free software application has some very important characteristics that outperform many commercial packages. Users can request for a feature and be in touch with developers who are willing to include that feature in future releases. The community of developers is open to listening to users for their comments and experience in using the software package. The development is fast, releases are relatively often and indeed, upgrades are free. Users who do have some knowledge of programming are free to modify the package and free to distribute the package so the community may have benefit using it. For that matter, use of *Audacity* is worth considering in short-term and long-term plans.

Audacity can perform well on weaker configurations though professional environments with demanding tasks require 800MHz CPU with 128 or 256MB of RAM and fast HD as the bottom line. Even that configuration may satisfy a modest radio station. I have been involved in a project where the radio broadcast machine with modest sound editing requirements is done on IBM personal computer with 800MHz CPU and 128MB RAM. This is very important due to the fact that many people and organizations still use personal computers with modest technical specifications. Even more, modest laptops may serve for the recordings, such as fact finding missions in localities where there is no access to electricity and no possibility to bring expensive recording equipment and their demanding power supplies. Power supplies are sometimes very expensive equipment. Power conditioners are often enormously expensive and often not affordable to advocacy groups or independent media. However, laptops with good quality batteries may be very helpful and if laptop is equipped with good quality PCMCIA audio card it may be sufficient for professional recording.



Audacity does have easy to manage GUI that is similar to many user friendly non-free audio applications

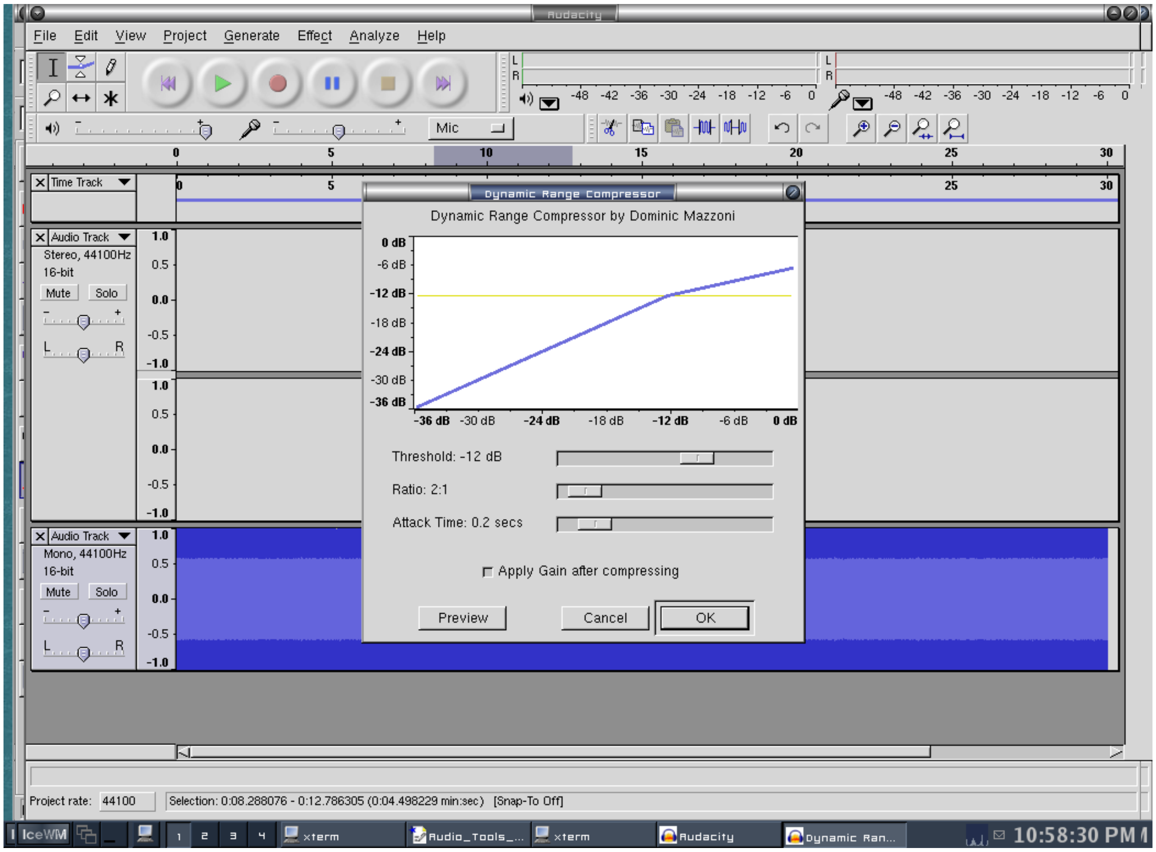
In our case a personal computer with 800MHz CPU and 128MB RAM is working flawlessly for more than 10 months without single reboot continuously playing back audio files, yet having enough CPU for modest audio editing. Indeed, *Audacity* is capable of multichannel audio recording for a beginner audio enthusiast as well as a straightforward process for a professional . However, the lack of a fully featured multichannel mixing console still keeps *Audacity* behind professional multichannel recording and mastering audio applications. To be honest, radio broadcasting does not have the technical requirements of mastering that professional mastering of audio recordings of music have. The features provided with *Audacity* are sufficient enough for the realization of work required by radio stations.   Actually, the quality of the recording is guaranteed by the quality of the audio card and the quality of the recording process itself. The operating systems on which *Audacity* can work are more than efficient for audio recording. Mac OS X is an operating system that is widely used in audio applications and Audacity may be good choice for audio recording on Mac OS X in order to replace very expensive audio applications. However, Audacity for Mac OS X may satisfy audio enthusiasts, radio program manager including those involved in education, training, advocacy groups etc.   Those, who are more technically skilled may write their own plug-ins for Audacity or exchange with others existing plug-ins written by other users. However, time that user dedicate in the beginning will return good results in the near future.



Audacity is capable for multichannel recording that is suitable for various projects

Although there is a rare need in media studio environment to record more than four channels at once, *Audacity* offers an unlimited number of audio channels for handling. It can record at once 16 audio channels, but multi-channel recording audio card is required for that. In real work, it is more needed to have capability of handling many files than being capable of recording many channels. This is useful, since often audio editing for a radio or TV program means keeping available various versions of four channel recordings, this requires presence of eight or twelve audio channels for instant playback. It is very important to stress that commercial vendors price such software packages very high and that any additional effect processing plug-ins need to be purchased separately. However, *Audacity* architecture is made in a way that enables *LADSPA* plug-ins to be used. Thus more than 150 varieties of plug-ins are available for use. An unstable plug-in can crash the application, it is therefore advised to test a few that one needs the most often and to use them. Other plug-ins may be tested with work that is not especially important just in case the unstable plug-in crashes the application and possibly damage user's recordings.
Audacity can handle *LADSPA, VST* and *Nyquist* plug-ins. *VST* stands for Virtual Studio Technology and it is introduced by the Steinberg Corporation. *VST* plug-ins are primarily developed for Microsoft Windows and Apple MacOS platforms. However, they are not free (free as in freedom) and there are successful replacements in the world of free software. Since *Audacity* is open to Nyquist programming language its limits are high up there in the sky. *Nyquist* is a powerful language named after Harrold Nyquist who was very important theorian of audio recording. *Nyquist* is aimed at sound processing and synthesis. Users who do have basic programming skills and a good knowledge of sound may experiment with it and gain astonishing results that can be applied in science as well as in contemporary arts. Indeed, those who are audio enthusiasts or professionals without programming skills should not expect from the *Nyquist* part of *Audacity* immediate satisfying results. The project leader Dominic Mazzoni has developed a dynamic range compressor plug-in that may be often used in media organizations. However, compressor, limiter and echo processor are often the most frequently used computer assisted tools. Although, many media organizations use specially designed hardware for such purposes, software plug-ins may be quite handy, these can be successful replacements from the point of functionality, cost saving and upgrade .
Although, *Audacity* DSP effects do not function in a real-time regime they do have a *Preview* button which enable the user to preview how audio files will sound after the effect has been applied. Real-time regime assume that change of sound is audible instantly and immediately so the user does not feel and delay while processing of audio is being done. Recordings made in broadcast environments are usually intact and they just need some editing touches. But, recordings made by advocacy and civic groups outside a broadcast environment often need editing by using various noise gate filters and other dynamic processors. *Audacity* is equipped with such audio editing tools, but it should be

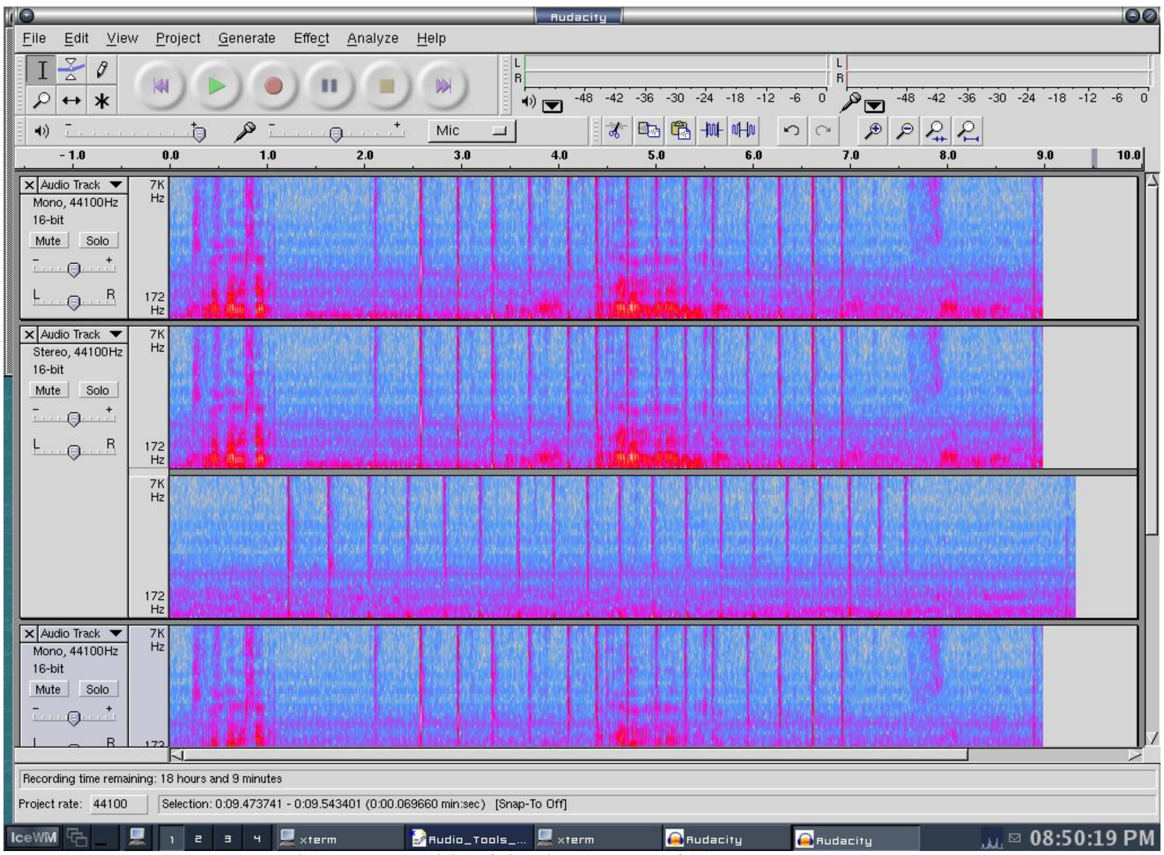
kept in mind that the quality of recording is always the central point of an audio recording. Thus, training of advocacy groups in audio recording may significantly improve their capacity to make successful reports from fact-finding missions. Implementation of DSP effects indeed, may increase the creativity of users and successfully improve their artistic and educational efforts.



Envelope tool is useful for visual control of gain of audio recordings

Traditional audio recording studios are equipped with expensive mixing consoles that serve for multichannel recording and mixing of audio recordings. Thus, in traditional studio there are various microphones or devices plugged in the mixing console. Mixing console then receives audio signal through its connectors and mix it with others according to the needs and taste of a person that manage audio recording process. Personal computer audio recording use a virtual software mixing console in order to interact with a multichannel audio card and handle multiple audio channels. Thus, instead of the real mixing console, software with graphical representation of mixing console interact with audio card that is connected with external microphones, tape recorders and perform audio recording process like in traditional studio.
The envelope tool is a tool that enables the user to modify the gain of a recorded audio signal by using the mouse to increase and decrease the gain by a simple

click and drag procedure as is shown on the above image. This tool is quite handy when fast editing is needed. Those who are more accustomed to traditionally set up audio studios would prefer virtual mixing consoles instead of an envelope tool. Journalists, members of advocacy groups and various audio enthusiasts may find the envelope tool quite useful for fast editing purposes. Final recordings may be exported in compressed and uncompressed audio file formats for broadcast, recording on CDROMs, USB flash or other media for transport and archiving on audio storage facilities. The free compressed format (*.ogg),* is sufficient enough to replace proprietary *.mp3* format when there is a need for high compression of audio files. But since many professional sound engineers prefer uncompressed audio files *.wav* and *.au* formats, they are sufficient for preserving a professional criteria of a recorded audio signal.



Audacity is capable of displaying waveforms as spectrograms

The analysis of audio recordings may be important in various situations. For example, detailed analysis of audio files may be very important in educational, scientific, medical and even forensic applications. *Audacity* is equipped with capability to display audio recordings as spectrograms that show the amount of energy in various frequency bands. The amounts of energy are represented by various colors so the user may easily recognize differences and variations according to dynamics in change of sound. The user may define a frequency range and parameter that define how wide or narrow the band is that can be analyzed by the *Audacity* spectrogram feature. Thus, users may focus their attention on some frequencies and phenomena happening within that frequency range. This tool may also be useful in understanding sound nature of human speech, music and natural sounds. Such an understanding is essential to acquire skills for successful recording of an audio signal that is increasingly important in the world characterized by intensive communications in very fast information highways.

Applications for inter-application connectivity
Jack Audio Connection Kit

Free software packages are not always envisioned and designed as primarily commodity products and therefore tend to work on a different set of principles. At times they can look like some 'user friendly' applications from the world of non-free applications. They usually have plenty of features that may be not equally as stable and are not always well integrated. But, overall we can say that the current status of the development of free software audio applications is in a pretty solid state. Actually, free-software audio applications rather reflect the creativity of the members of the development community of each software package. The concept of features in the world of free-software is sometimes rather envisioned as an organic form which grows its branches and therefore features sometimes in an unexpected way. Because of this, users often use applications that are equipped with features that are very rare or non-existing in the world of commodified software. The increase of features and the appearance of new software packages decreases the need for expensive hardware equipment. This puts the user in a situation to create his/her own working environment by combining applications and combining features of certain applications by ensuring that inter-application exchange of data is going well. It is not easy to pursue that task in the world of non-free software. The proprietary character of some applications and the lack of design of some operating systems, prevent users from solving that problem. There are attempts by various companies to solve that problem, but due to different programming styles and the non-free character of software development solving of this problem is still very slow. In addition, the inter-application connectivity of interfaces increases the price of applications.

In the recording environment users often capture data from equipment placed outside the computer. Users may use microphones, cassette decks, dictaphones, CD players and other recording equipment which they use for recording activities in their work. However, in traditional studio it is needed to have various devices ant to connect them with various cables and connectors. Since, audio equipment manufacturers often define their own proprietary formats in order to force the customer to buy only their products one studio should be equipped with various interface devices and format converters so the studio may function as a system. Commercial non-free software applications are conceptualized as commodities targeted at specific markets and their features are designed to satisfy customers from a certain segment of market. In some ways they do have their own file formats and behave like their hardware counterparts. If user's activities are wider than one target group defined by the marketing department of the software vendor, the user is forced to purchase a highly priced software package in order to perform what are relatively simple tasks. Such discrepancies still pose very important question to the users. It is very important for the user to know does he/she really need such and such software, how it is possible to combine features of several software packages and how those software packages may exchange data in order to enable user to be more productive by using several software packages. Jack is a software package that is developed with aim to give answers to these questions.

Why you use Jack?

Jack is an inter-application connectivity kit that enables users to untie that knot and make connections according to his/her preferences. The role of *Jack* is to ensure that synchronized data can move between audio applications with no audible unwanted delays, jumps or stuttering. When user use personal computer in order to do audio recording he/she interacts with its operating system which enable user to start audio application, audio application itself, audio card that through its drivers communicate with an operating system and audio application itself. In the world of GNU/Linux audio card drivers are inserted in the kernel of operating system itself. If we look in more detail we will see that seamless functioning should be ensured by efficient communication between the operating system kernel, audio card, *ALSA* drivers, *Jack* and audio applications themselves. The package management and installation system that is built-in to the GNU/Linux operating system ensures that all those settings are done automatically. Associations and companies that build GNU/Linux distributions made necessary scripts that prepare operating system kernel, *ALSA* drivers, audio applications and *Jack* to communicate seamlessly. Usually, there is no or little tweaking needed so the user may use his/her new digital audio workstation immediately. Radio stations and advocacy groups that dedicate a lot of time to audio applications should dedicate one or more computers to a GNU/Linux distribution that is dedicated to audio. This is very important since often experiments with installation and de-installation of software packages may lead to confusions in the operating system. In addition, since audio (and video) activities are often very demanding tasks the system should be optimized for that purpose. If user performs simple audio recording tasks then usual system configuration may be sufficient. But, if users do practice multichannel recording activities they should keep one computer exclusively for that purpose. For example, ecologists who record certain animals from shorter distances do not necessarily need multichannel recording capable computer. NGO activist involved in interviewing citizens on certain social issues does not need multichannel recording also. But, local cultural worker involved in audio recording of local orchestra may need computer dedicated for audio applications. Radio program editor may satisfy his/her needs with modest configuration personal computer, but movie or TV program audio recording may require dedicated computer for audio applications.

Qjackctl is graphic frontend for controlling Jack

By using *Jack* the user can easily move data from one application to another as well as between certain applications and the port of the user's audio hardware. *Jack* itself does not have a GUI, but there is an applications called *qjackctl* which acts as a frontend for Jack enabling the user to control it seamlessly. It is highly recommended to study man pages for *Jack* and to subscribe to discussion lists about *Jack* in order to do fine tuning of a *Jack* server. *Jack* tuning can be a time consuming process, but its features really pay off for efforts undertaken by the users. The setup of *Jack* is usually an easy and straightforward process, but some audio cards with poor technical specifications need some additional tweaking. The best results are gained if the users personal computer is equipped with an audio card that is fully supported by *ALSA* (Advanced Linux Sound Architecture) package.

Since semi-professional and professional audio cards are not expensive, it is highly recommended to have as best as possible audio card. Manufacturers such as MAudio, Terratec, RME, Audio Science, Creative Labs, Cirrus Logic, Avance Logic and many others have given specifications of their chipsets to developers so that the proper drivers are written and exist within GNU/Linux and other free software operating systems. Thus, users can find on the market affordable, yet technically sufficient audio cards that can perform audio recording on a professional level. Some of those audio cards are equipped with 4, 6 or 8 audio inputs and outputs for more complex audio applications and software inter-connections. Those who are involved in planning and design of their audio system should seriously take into consideration rule saying that each chain is strong as it is strong its weakest link. Thus, technical specifications of hardware purchased should be prepared in a way that it is not expensive and that it sufficiently meets requirements of audio recording process. However, a brief consultations with someone who is familiar with audio recording techniques may be very helpful and sufficient for affordable yet efficient audio recording system. Careful planning may provide users with high quality audio recordings and efficient mission realization. Hardware manufacturers such as Rane and MAudio wrote brochures that may be very helpful in planning stage of the users audio projects so various inter-connection equipment should not be needed.

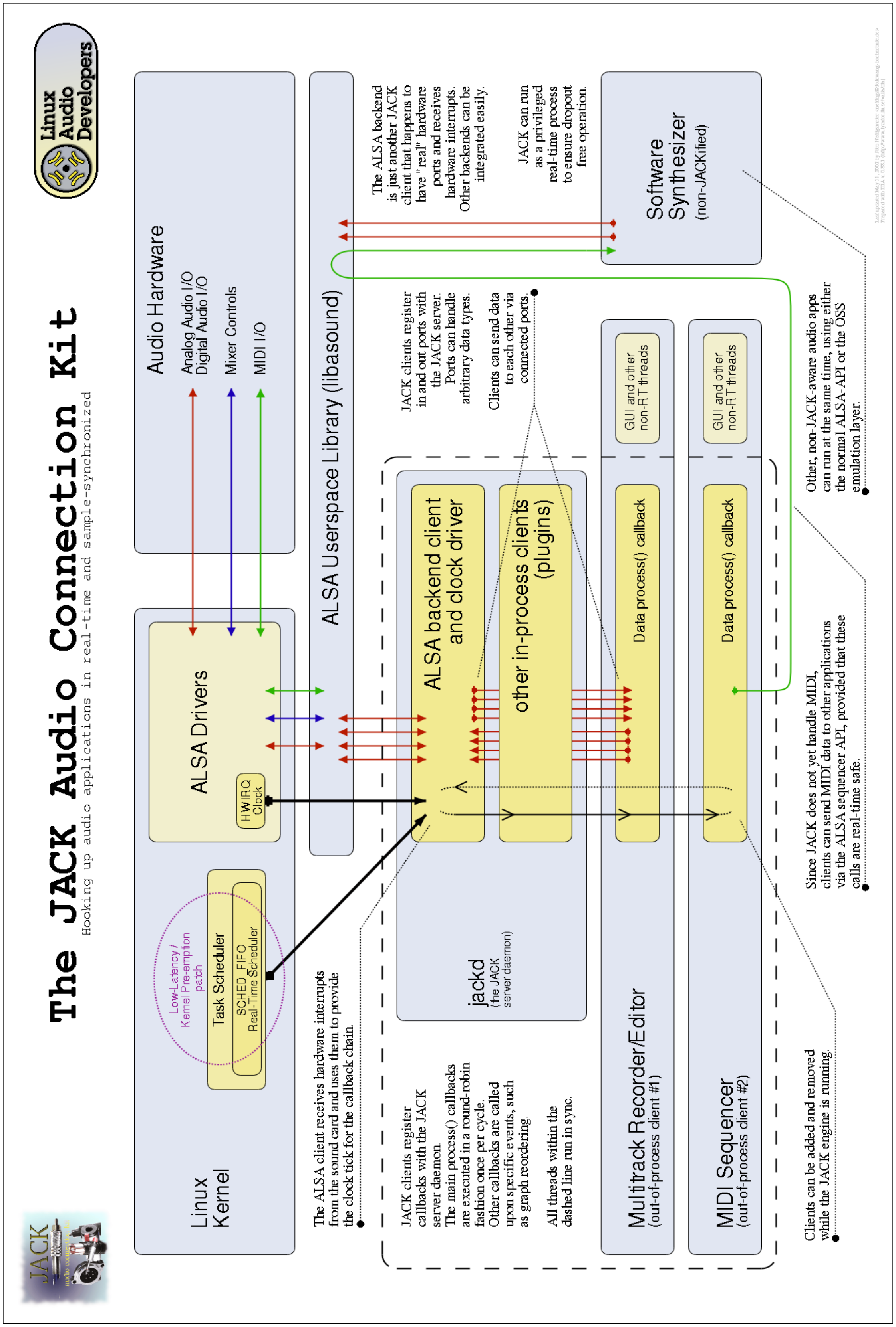
Jack Patchbay is powerful inter-connection tool

One of the most important features of *Jack* is its patchbay. *Jack's* patchbay is a powerful tool that creates all the 'virtual cables' and links between various audio applications and the audio ports on the audio card so that highly demanding and complex audio mixing and processing techniques may be implemented seamlessly. The easiest control of *Jack's* patchbay is by using *qjackctl* so all inter-connections are at the user's fingertips. (Seamless operation is possible with personal computers that do have CPU stronger than 1GHz and at least 256MB of RAM. But, nowadays such personal computers are commonly present in offices and studios. Beside those applications there are more and more applications that are compatible with Jack.). Implementation of patchbays is known to people that do have experience working in radio, music or TV studios. But, advantage of personal computer and free software is that expensive equipment and patchbay inter-connection devices from studios are free and placed in personal computer that may be used by all. For example, radio program editor may need background music to be covered by telephone calls from the audience and at the same time audio signal from telephone call should be enriched so it may sound properly. Radio program editor may inter-connect audio recording application, plug-ins and music playback application that are controlled by software audio mixing console through which radio program editor may control loudness of each part of sound to be broadcasted. NGO activist that records meeting of local minority group may position two or three microphones in the meeting venue in order to record discussion between various participants. However, each microphone may need different compressor, equalizer and noise gate that are interconnected with audio recording application that in addition broadcast that recording via Ethernet to the nearby radio station. Jack is very helpful tool in such situations and it replace expensive equipment and numerous and lengthy inter-connection cables and devices that are not affordable by many civic groups, independent media or local culture and arts organizations. Hence, its numerous possibilities may be used in a creative way.

Jack is a project with fast development and with fast development of applications that are compatible with *Jack*. Bearing in mind that some very professional audio cards are supported by *ALSA* drivers, using *Jack* along with audio applications could satisfy some very demanding professionals working in radio broadcast, music composition and audio recording. The real success of the jack functionality is in fact that it is based on so-called low-latency principles. Low-latency principle assume that there are human-unnoticeable delays between an input being processed and the corresponding output. Due to possible inter-connection with hardware with Ethernet connector it may serve well for instant broadcast, editing for media organizations, arts and culture projects, as well as Internet audio blogging, Internet radio, Internet news agency projects etc. Since some audio cards supported in GNU/Linux are manufactured with rigor technological criteria they may serve for high school and university education and activities in applied sciences such as ecology, various measurements, seismic research, geographical study, physics, mathematics, technology development. It is very hard to say what the limits of use are since free software is sometimes developed quickly due to the free availability of source code for software. The freedom of use and distribution of the source code and compiled software is encouraged, so it is less often necessary to reinvent the wheel. Thus, various development projects may be introduced where digitization and data acquisition is an important part. It is hoped that this kind of philosophy can help even poorer areas using audio to their production of audio material and access to information and knowledge.

Diagram on the next page shows principle of interconnection with jack[2]

2 Image is taken from the jack site, copyright belongs to author of image

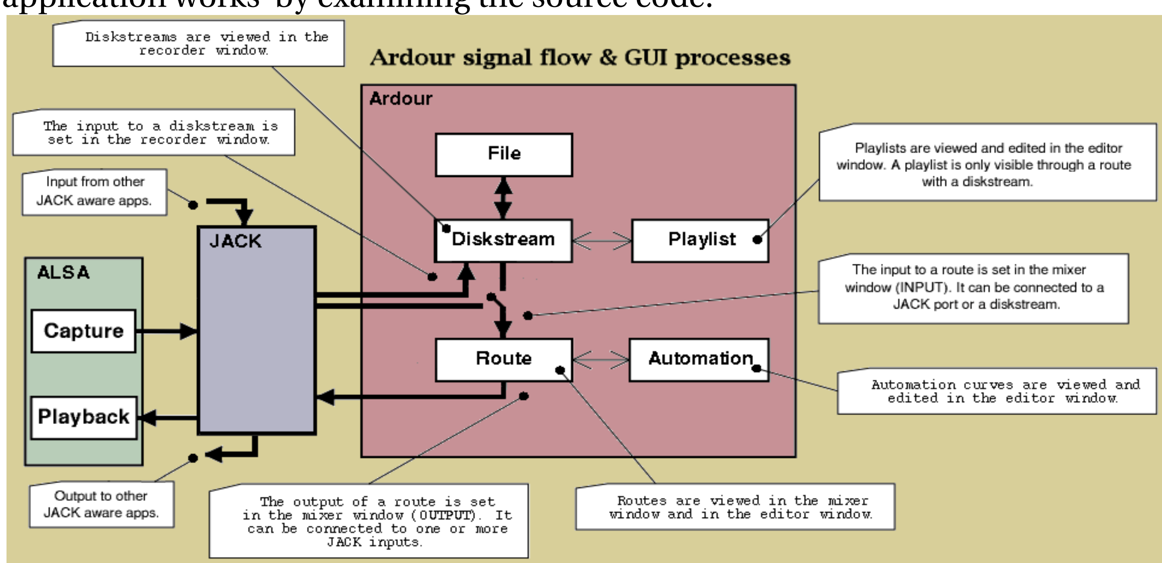

The JACK Audio Connection Kit
Hooking up audio applications in real-time and sample-synchronized
Linux Audio Developers
Linux Kernel
Low-Latency / Kernel Pre-emption patch
Task Scheduler
SCHED_FIFO Real-Time Scheduler
ALSA Drivers
HWIRQ Clock
Audio Hardware
Analog Audio I/O
Digital Audio I/O
Mixer Controls
MIDI I/O
ALSA Userspace Library (libasound)
The ALSA client receives hardware interrupts from the sound card and uses them to provide the clock tick for the callback chain.
JACK clients register callbacks with the JACK server daemon.
The main process() callbacks are executed in a round-robin fashion once per cycle.
Other callbacks are called upon specific events, such as graph reordering.
All threads within the dashed line run in sync.
jackd
(the JACK server daemon)
ALSA backend client and clock driver
other in-process clients (plugins)
JACK clients register in and out ports with the JACK server. Ports can handle arbitrary data types.
Clients can send data to each other via connected ports.
The ALSA backend is just another JACK client that happens to have "real" hardware ports and receives hardware interrupts. Other backends can be integrated easily.
JACK can run as a privileged real-time process to ensure dropout free operation.
Multitrack Recorder/Editor
(out-of-process client #1)
Data process() callback
GUI and other non-RT threads
MIDI Sequencer
(out-of-process client #2)
Data process() callback
GUI and other non-RT threads
Software Synthesizer
(non-JACKified)
Clients can be added and removed while the JACK engine is running.
Since JACK does not yet handle MIDI, clients can send MIDI data to other applications via the ALSA sequencer API, provided that these calls are real-time safe.
Other, non-JACK-aware audio apps can run at the same time, using either the normal ALSA-API or the OSS emulation layer.

Multichannel mixing and recording

Ardour

Although *Rezound* and *Audacity* are audio recording applications capable of multichannel recording of audio they lack some features that are needed for professional multichannel recording on digital audio workstations. Jack itself has solved some of those problems, but *Ardour* is a multichannel recording audio software that is designed to operate as a professional multichannel audio recording workstation.

Its native file format is EBU (European Broadcast Union) broadcast wave file format that is standard for broadcast environments. *Ardour* supports major professional file formats such as *.wav, .aiff, .riff, .au, .snd* etc. *Ardour* is capable of interacting with SCSI and firewire (IEE1394) hard disks. It is capable of communicating with software packages that store large size files in computer configurations with several hard disks. This is very important when logging of radio program is required due to legal obligations for radio stations that broadcast news and live-talk program. The mixing and editing controls for *Ardour* may be associated with MIDI control change parameters, so MIDI control surfaces and keyboards may be used to control *Ardour*. A virtual multichannel mixing console and more than 80 plug-ins that maybe applied on an unlimited numbers of tracks or more than 24 channels of recording, make *Ardour* a really professional looking application. It has a well developed graphical user interface and convenient positioning of controls, these make *Ardour* easy to work with and increasingly sound engineers will find *Ardour* among their favorite applications. Most of the controls are accessible in a small number of steps, so users do not need to remember complicated procedures to achieve desired results. Commercial non-free applications with features like *Ardour* would be very expensive. Despite the argument of price, those who are familiar with programming may learn how audio multichannel recording application works by examining the source code.



Ardour signal flow

Thus, even limited investments in free software modification may pay off when someone wants to build applications for scientific, training or other purposes.

Ardour is equipped with tools that are needed for professional audio recording

Ardour is an application that is aimed at professionals involved in multichannel audio recording and it is equipped with the main tools needed for professional audio work. Multichannel recording and mixing are easy to perform. *Ardour* requires *Jack* to be set up and started correctly prior to starting of *Ardour*. A detailed explanation of *Ardour* functionality would need a lot of space, since *Ardour* is a powerful application with numerous features. It does have graphical interface similar to professional hard disk recorders. It can communicate with various software synthesizers and other audio applications. It is not designed for writing musical notation as is *Rosegarden*, but due to its possibilities of being inter-connected with musical notation software packages it can be a part of the *Jack* inter-connected system that is capable for audio and music notation performance.

Ecasound

Ecasound is special application in many ways. In fact, although there are graphical frontends for *Ecasound* the purpose of mentioning *Ecasound* here is that it basically runs without graphic user interface. However, one can question mentioning application that is designed for multichannel recording and mixing without graphic user interface. Although, it may look hard to grasp how to use console based application for multichannel audio recording there are some very important advantages why it may be good sometimes to use such an application without graphic user interface. Graphic user interfaces are often memory and CPU consuming and usable mainly by the users who do not have any sight-impairments. Indeed, we should be aware of the fact that software should be as much as possible useful to those who do not have computers with the cutting edge specifications and that sometimes software users are people that do have various disabilities including sight-impairments. Command line in console by definition provides us with an opportunities for better interaction for sight-impaired users, lower resource requirements and better performance, off-line and realtime. Due to text to speech conversion software applications command line typing may be easily converted to speech so user with sight-impairments may hear what is typed and what software responded to the user as a result of the user's commands. It is possible to do all basic controls of soundcard in console by using command line. LADSPA plugins, *Jack* inter-connectivity toolkit, basic soundcard control mixers, ALSA drivers, various players are fully controllable from the command line shell. Due to its complexity *Ecasound* is capable of multichannel recording, applying various effects with different parameters and mixing of incoming audio signal. Actually, it is fully featured multichannel recording software without native graphic user interface. All parameters that we set by using graphic user interface in other sound recording applications we can can set by using *Ecasound*. When we look carefully we apply a chain of parameters to each track that we record by using multichannel recording software. The same applies for *Ecasound*. It is necessary to write so-called chain commands. *Ecasound* chain command may look like example below:

```
ecasound -b:512 -r -f:16,4,44100 \
-a:1 -i jack -el:reverb,1500,0,0,0,1,1,1 \
-a:2 -i jack -el:reverb,1000,0,0,1,0,1,0 \
-a:3 -i jack -el:reverb,2500,0,0,1,1,0,1 \
-a:4 -i jack -ef3:800,1.5,0.9 -km:1,400,4200,74,1
-km:2,0.1,1.5,71,1 \
-a:all -o /home/your-directory/your-4channel-recording.wav
```

In this example we set up four channels for audio input via the *Jack, a*dded a LADSPA reverb effect to the first three channels, *Ecasound's* resonant lowpass filter to the last track, set up a MIDI controller for the *Ecasound* filter and recorded incoming audio streams into a four-channel WAV file. With some practising chain commands may be written with ease and users can achieve satisfactory results in their audio recording activities.

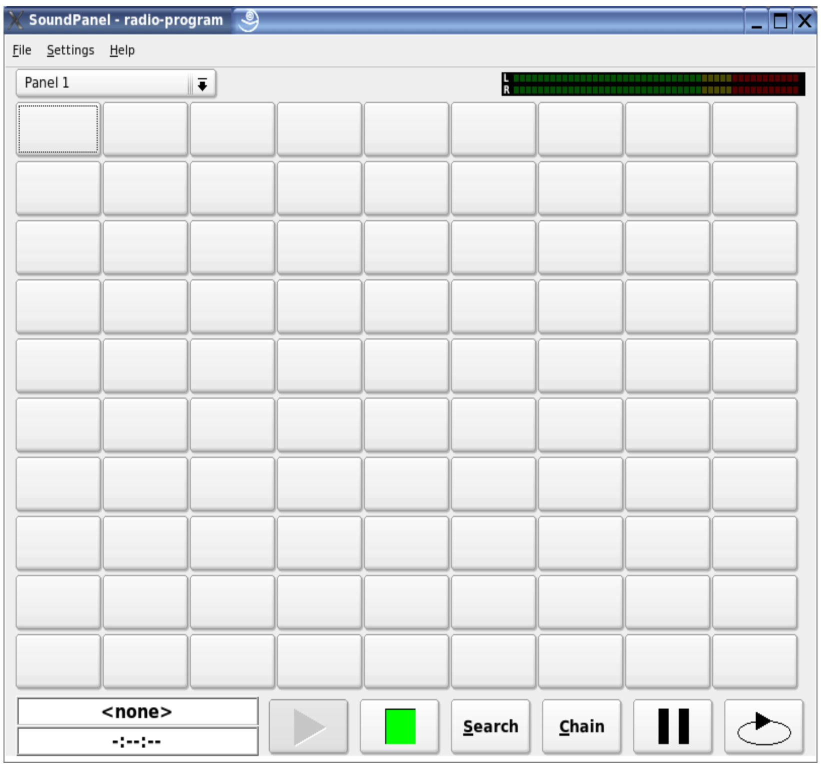
Playback, streaming and broadcast automation

In many cases we can say that playback and streaming are in some ways similar terms, but in audio applications jargon streaming is more understood as process of delivery of media content by using TCP/IP protocol over the Internet. Usually, streaming is focused on compressed audio content. It may be very useful in situations when there are no resources for building the real radio stations having provided that there is sufficient Internet access. In addition, it may be useful to use Internet streaming when legal regulation of spectrum may prevent people from broadcasting the content they would like to make available to the audiences. But, still in many cases sufficient access to the Internet is not provided in many developing areas, so streaming is still unfortunately a sort of luxury. Truly, *Icecast* is one of the best streaming software packages available. There is version for the Microsoft Windows and GNU/Linux. In addition, multimedia players such as *Mplayer, Xine* and some others are capable of receiving streams so they can serve as players of streamed content. Although, there are a lot of Internet based radio stations, it became easy to establish a real radio station operated and automated by people using free software. One good practice case study is Salem Radio Labs which is a division of Salem Communications Corporation. They host a www site with a collection of free software applications for radio station operation and automation. Actually, the everyday operation of the radio station is based on the use of various hardware for professional radio broadcast. Thus, precise clocks, synchronization devices, audio processing tools, telephone hybrides are part of the everyday practise of each radio station. Sound engineers and technicians who want to operate their radio station by using free software should be very skillful and have a systemic approach to their work. In fact, the operation of a radio station is highly demanding task that needs a simple interface for operation that is not easy to accomplish. But, using GPIO devices (General Purpose Input Output) is one of the ways that enable technicians and sound engineers to connect various devices with their computers and manage their operations. Technicians should know how to change the position of their satellite dish, turn on remote CD players, amplifiers or other devices etc.

It looks complicated, but if an approach to the radio station management is well thought out and done management of a radio station may become a quite simple and automated task. Due to a variety of legal and technical conditions that any radio station has to meet, the radio station should have a range of equipment, such as logging devices, telephone lines and well designed management tools. Since free software audio applications are well developed, a radio station does not need to spend a lot of money on expensive equipment. Sound recording, editing, creation of a playlist, remote control of the station, audio logging, connection with telephone lines within an on-air program may be done with free software and personal computers. As storage space, memory and processing power becomes cheaper and more efficient, it is not too expensive to purchase equipment that was not possible to dream of a couple of years go. Nowadays personal computers are more than fast enough and efficient to perform complex audio tasks. Thus, storage, automation, management, inter-connection is no longer an expensive and technically very complicated task. Personal computers, free software and and some other small core equipment are sufficient to manage a radio station for much less money than was needed a couple of years ago. This fact has opened a field of opportunities for independent journalists, civic advocacy groups and other organizations to achieve much better results in informing the public on their work in the community. Software developers at Salem Radio Labs are communicating with developers and end-users worldwide through their discussion list that is free to subscribe to. Those who are interested to install and compile files on their own, may download sources and install or modify software if needed.

SoundPanel

Soundpanel is an application that is designed by radio professionals for

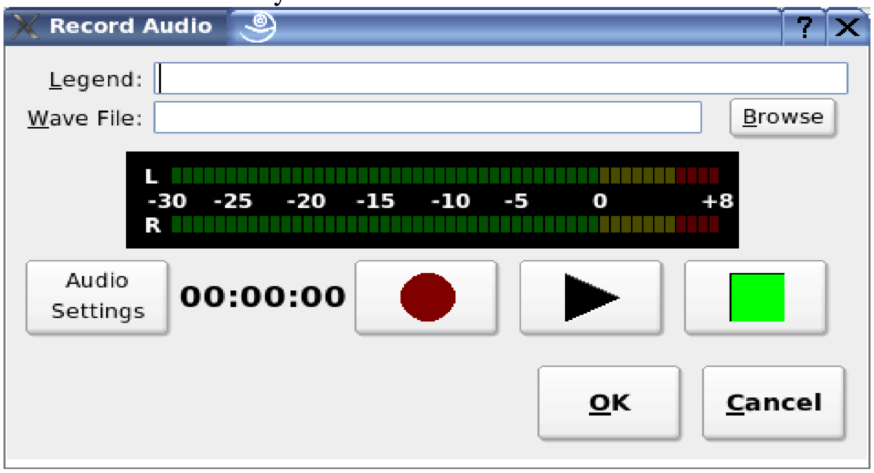


radio professionals that use GNU/Linux operating system. It's graphical user interface is simple, all features are straightforward and it is easy to use. This is very important since radio speakers, DJs, or other people who conduct the program usually have very busy hands; writing notes, preparing CDs, moving microphones, holding the telephone handset, communicating with sound engineers etc. Hence, the application they use should be very simple. *Soundpanel* consists of three parts. The first part is the main panel with blank buttons where user can write titles of recordings that will be played. On the bottom part, the user can find buttons that will open possibilities to record, search, start or stop playback.

Recording of news, readings, songs, various announcements must be an easy task. so the news, additions, announcements may be added to the program easily and instantly, with no interruption of program and in a timely manner. *Soundpanel* stores audio recordings in user defined directories. It can run on personal computers with modest technical specifications.[3] *Soundpanel* run

3It is tested on Siemens personal computer with 450MHz CPU and 160MB of RAM on SuSe 9.2 GNU/Linux distribution.

flawlessly with a cheap *ALSA* supported audio card. The main panel of the *Soundpanel* that is shown on above image is user oriented so the user may name the songs and control the broadcast program according to collection of songs – recordings that are stored in the database and if needed instantly jump from one recording to another that are named on the main panel of the *Soundpanel*. Such a flexible and configurable control interface is what each radio sound engineer dreams about. Since sometimes radio stations are not rich institutions it is important that software may be used under modest circumstances. Personal

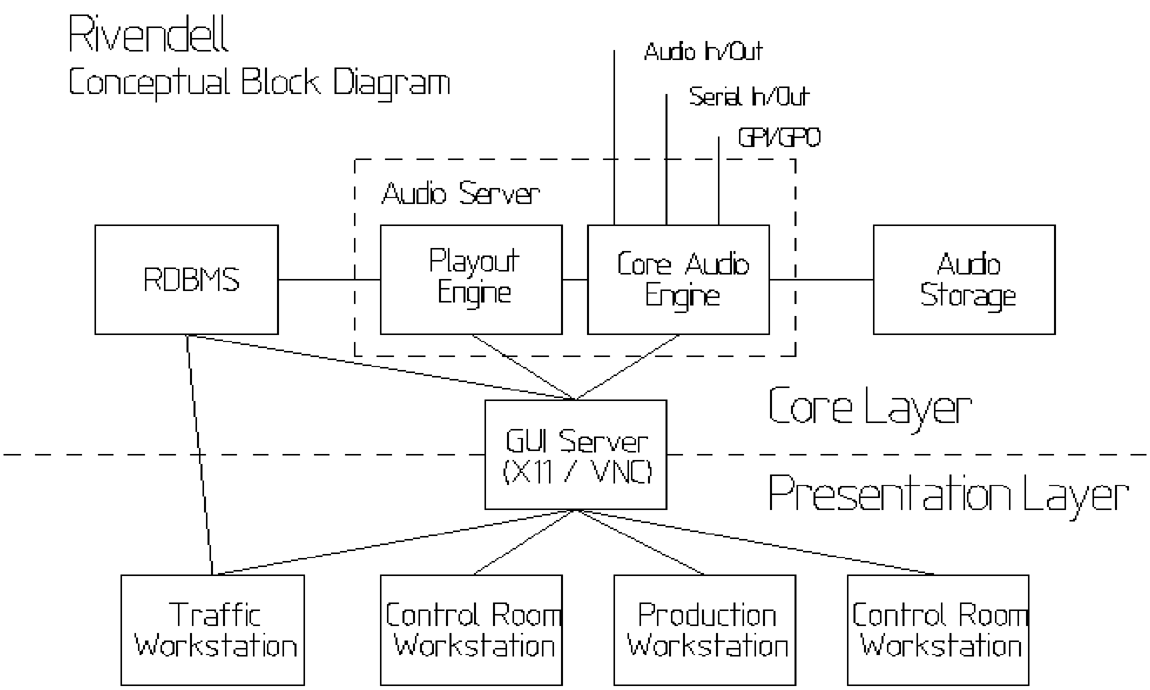


Easy andstraightforward controls of recording

computers with much better technical specifications are widespread, so we can say that *Soundpanel* is sufficient enough for modest organizations. If the users audio card is supported with *ALSA* drivers it is easy to use it. However, it is necessary to set up the recording level on the sound mixing applet so that the recording level may satisfy the user with its clarity and comfortable volume level. The recording level may be controlled via its graphic representation or recording activity with metering and the location where the recording will be stored ensuring that the audio settings are at the user's fingertips. The ability to queue recordings and name them on the sound panel makes possible an easy and smooth management of the radio program. The use of audio CDs with pre-recorded audio material is also easy, since there is a ripping capability of *Soundpanel*. It is possible to name tracks according to the needs of the user. A very important control is the ability to normalize ripped audio material. Actually, often audio CDs are recorded differently which is not suitable for a radio program, in particular when it should have approximately the same sound level all the times so the listeners will not notice significant variations of the sound volume level that can make listening uncomfortable for them. A comfortable level of normalization can be defined by the sound engineer who is familiar with other equipment too. *Soundpanel* obviously does have features that are easy to manage and control which is necessary precondition for successful radio program management. Radio program speakers will enjoy the way *Soundpanel* is designed, since it enables them to create dynamic and vivid radio program.

Rivendell

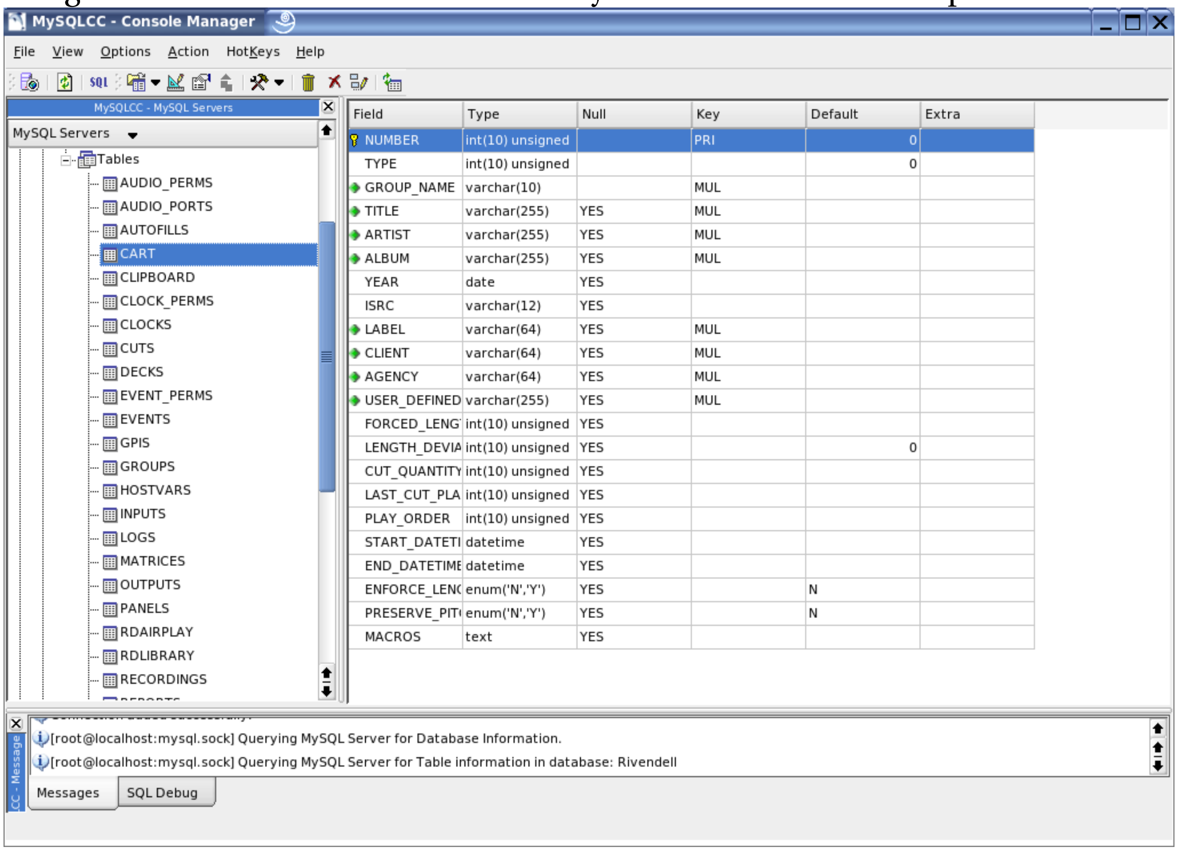
Rivendell is the horse power and brain of the radio station. It is an application carefully designed to have in mind systemic approach to the operation of the radio station. A radio station needs to have a collection of recordings in order to arrange various playlists for their audiences. In many countries radio stations are obliged to log their programs for a certain period of time in case anyone requests a legal investigation in to what has been broadcast ed. Therefore, radio stations need to have a relational database management system in order to collect, sort out, manage and store recordings. *MySQL* is a free relational database software capable of handling this and even much more complicated tasks where speed and stability is needed in heavy duty environments. Radio technicians and sound engineers often use GPIO cards to interface computers with other equipment that should be computer controlled. However, radio software should have a part of it dedicated to a General Purpose Input Output (GPIO) drive in order to perform efficiently the remote control of various devices such as CD players, audio processors, antennae etc. All these parts should be accessible from the computers that are dedicated to traffic, production or control room. The multiuser and networking architecture of GNU/Linux allows such a complex design to exist and is materialized in the software package *Rivendell.* Rivendell may be installed on GNU/Linux and Windows platform.[4]



Rivendell design meets criteria of complex radio station system

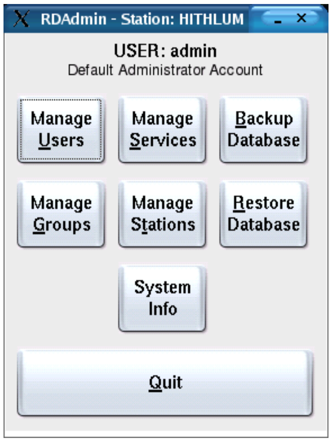
4 This diagram is used from Salem Radio Labs www site.

MySQLCC is a frontend that enables users to manage a database that is initially created by the installation of *Rivendell* itself. Actually, *Rivendell* initializes a set of tables that are designed in a way to manage sets of data needed to manage a radio station. In those tables users can define and manage music genres, playlists, times of broadcast, properties of audio files etc. A database that is initialized by installing *Rivendell* is sufficient enough for management of very complex radio station so the user does not need to have expertise in relational database design. However, basic skills and knowledge on relational database design would be useful in order to modify database to the user's specific needs.



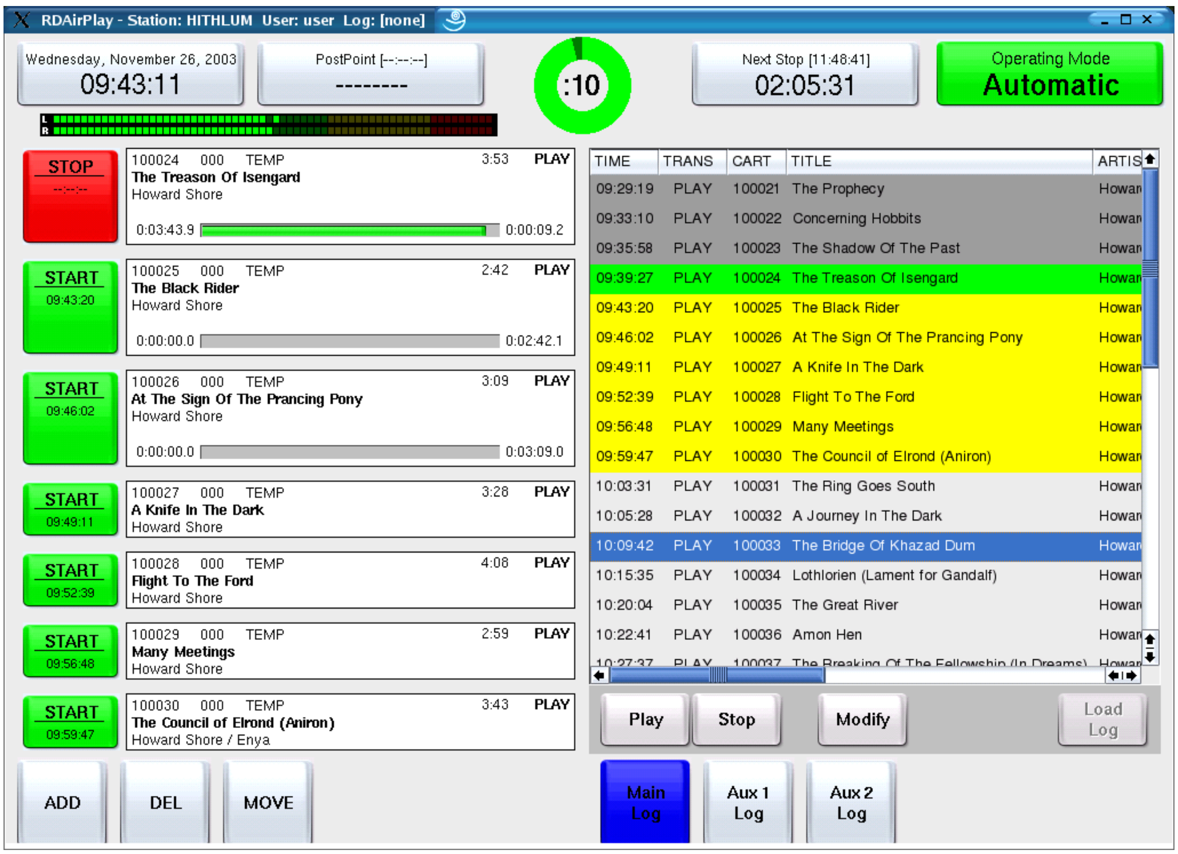
Rivendell initialize by default complete and comprehensive radio station database

A person responsible for administration of a radio station relational database may administer it easily by using *MySQLCC* or *RDAdmin*. *RDAdmin* is a small but important component of *Rivendell* that enables administrators to manage users, groups, services, backup and restore databases. Thus, a radio station that does not have enough funds to employ a relational database design expert, may be able to do this on their own. The application is designed in a way that ensures simple and straightforward operations for one radio station or network of radio stations. Actually, well organization of work with data requires well organized database and vice versa. Those two elements of work are inextricable for successful for. Instead of traditional numerous rooms for storage of audio recordings it is possible to do the same job on computers and database software.



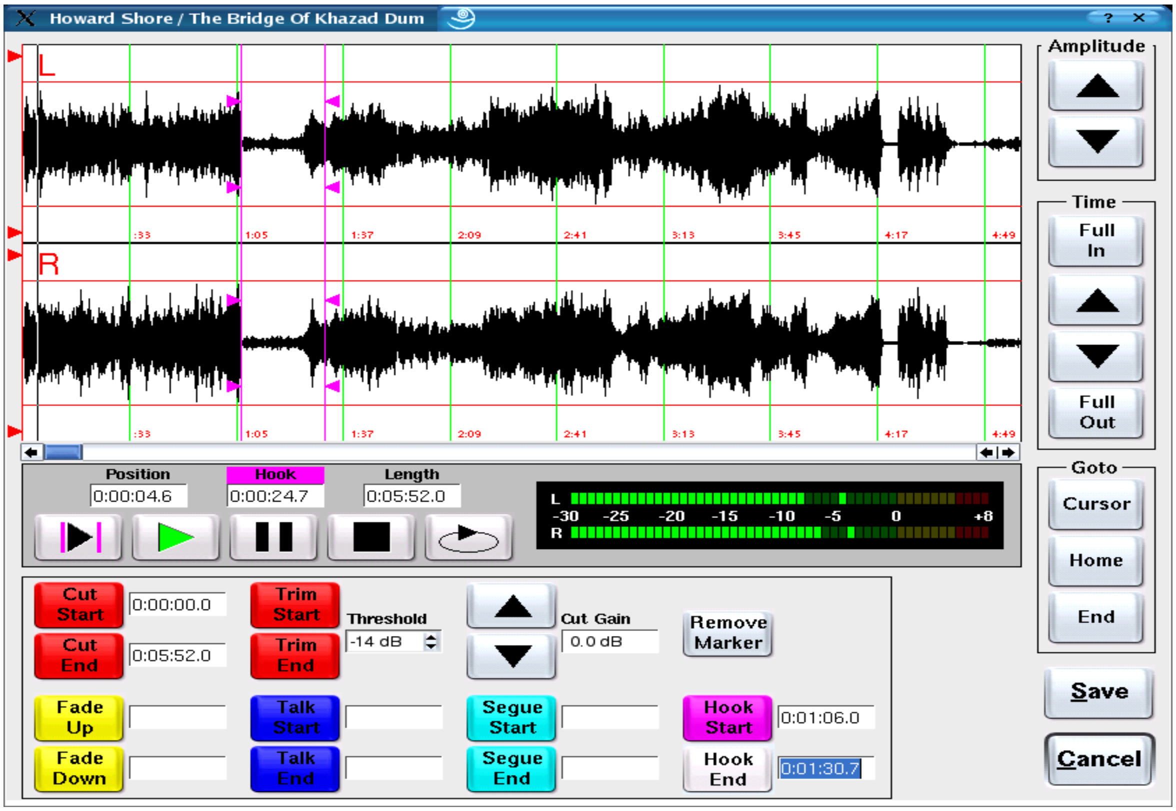
Simplicity of management is necessary precondition for successful operation

RDAirPlay is a component of *Rivendell* that reminds experienced radio technicians and sound engineers of devices that were very costly just a couple of years ago. Today, it is possible to perform the same and even more complex tasks by using personal computers and free software. Precise and comprehensive definitions of the play lists, timing and interaction are very important, especially as speakers have to interrupt playback in order to broadcast hot news, announcements or continue the talk program after commercials or music.



RDAirPlay is a tool for on-air program management

The self-explainable user interface makes management of the program seamless and straightforward. The successful management of playlists and the way that songs play after each other, or how various audio files fade in or fade out, is a very important part of the program management that makes the program pleasant to listen to for the listeners. Since a radio program often tends to communicate with listeners in a rather intimate environment and atmosphere it is very important that audio material is prepared taking that fact in to consideration. Segue is a term that explains how two songs or two audio files cross-fade across each other in a smooth and seamless way. A special part of *Rivendell* is dedicated to defining the duration of segues[5]. The sound engineer may see the files that are supposed to segue, showing set markers where the segue should start and finish. By doing so, the sound engineer can ensure that the production of the radio program reaches the kind of professional level only otherwise possible by using expensive equipment. Since in many cases independent media and advocacy groups are faced with budget constraints, such a software package can greatly minimize the costs of operation of radio stations.



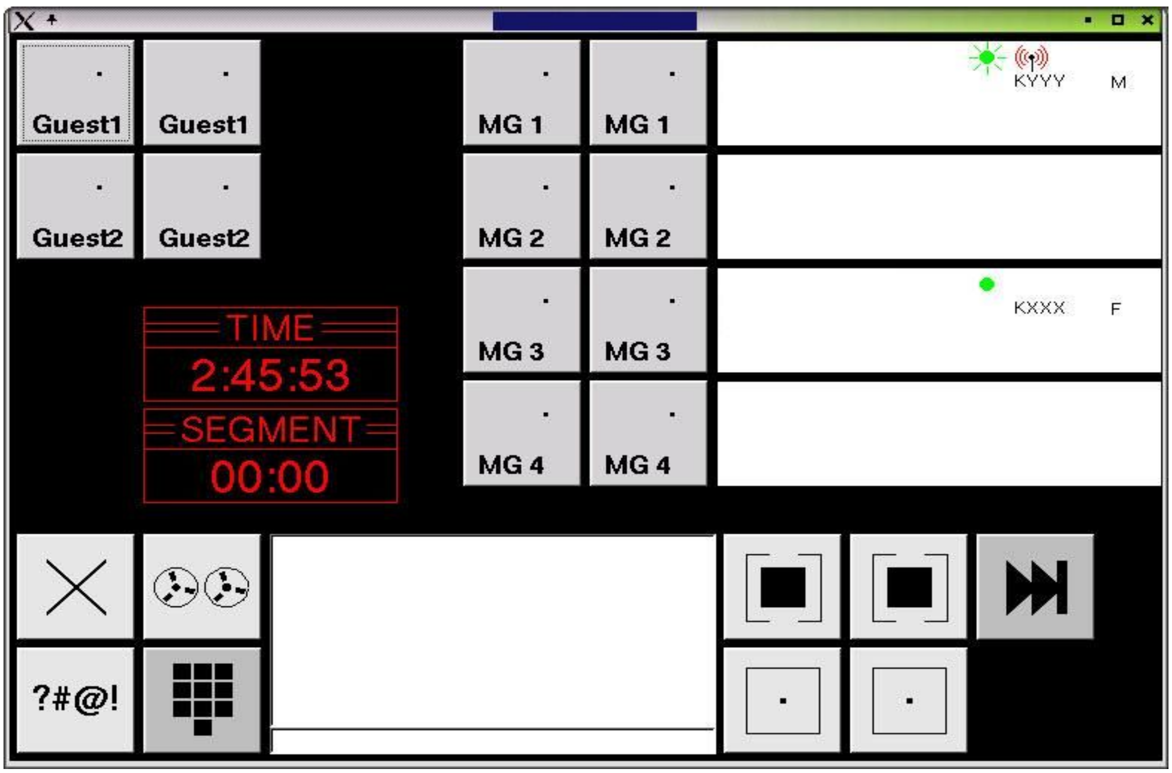
Segues are small but significant spices in radio program production

Call Commander

One very important characteristic of radio is the 'call-in', that is also at the core of the mission of independent media and advocacy groups. In fact, giving an opportunity to citizens to say publicly what they think about certain topics, to communicate with other people expressing their opinions, attitudes, wishes, complaints, solidarity or any other important message is important. This genuine human need is at the core of radio station and civic activities that some may call "from below" paradigm. Radio stations do this by connecting telephone

5 *Segue is smooth proceeding from one tune into another tune or voice so the sounds are smoothly intervowen making transition pleasant to listeners.*

lines, for this they use a specially designed device for that purpose called telephone hybride. Since telephone lines have a different signal than normal audio, the purpose of telephone hybrides is to enable the telephone line signal to be heard in a radio audio environment. With the aim of integrating that part of the radio on-air program in to the productions activity, software developers have developed a software called *Call Commander. Call Commander* is a software package that interfaces personal computers equipped with audio cards with telephone hybrides. It may be used in GNU/Linux and Windows platform. It provides the user with all the necessary controls needed for the successful management of telephone calls and including them in on-air radio program. the management of telephone calls needs concentration, and easy handling as well as a timely and efficient reaction to the dynamics of on-air telephone conversations .



Call Commander does have simple controls

Indeed, *Call Commander* may be used in any environment when interviewing citizens by using telephone lines is concerned. Thus various polls of public opinion, social research, artistic interactive work, psychological support to victimized people that need assistance of professional may be just some of many fields of application where *Call Commander* may be applied.

Summary:

Computer audio system itself is a kind of system that does have a lot of technical requirements to function properly. Actually, it requires a computer capable to perform audio tasks, stable and efficient operating system, available drivers, good quality audio card, stable and precise software with enough features and available good quality peripheral devices such as amplifier, loudspeakers, microphone etc. Audio software is not like text editor, it is much more complex. We can say that really capable and easy to use audio software is not older than 6-7 years which is not much. As far as audio software for GNU/Linux is concerned we can say that powerful and easy to use software exist 2 - 3 years. But, audio applications written for GNU/Linux develop fast and there is increasing number of new software packages. While many commercial software vendors have been acquired by larger companies in the recent years, we can say that rapid development of GNU/Linux audio applications is getting more attention even in professional audio magazines. This overview is focused on small number of GNU/Linux packages that may be used in music, media organizations, civic advocacy groups etc. Since idea of freedom of software is gaining its popularity it is worth to try it and join the community of developers, users, manual writers, software localizers and practice civic social responsibility by contributing to providing access to information and knowledge to all.

Software packages aimed for audio recording and editing, mixing, radio station management, automation, radio program broadcast are briefly presented in order to help those who may be interested to understand what they can to do and how they may be used. However, deeper insight and learning requires hands – on experience. If you are newcomer to the world of free software it may be helpful if someone who is familiar with GNU/Linux may assist you to prepare your system for your first hands-on experience. However, some packages may be used on Windows and Mac platforms and they are sufficient for many tasks that user may need. This overview should follow fast developments in the world and in the time of finishing of this overview I received information that some radio packages will be presented in Washington DC, USA on the meeting of broadcast professionals. Often invention and testing of new telecommunication protocols require from all of us to be curious and hard-working in the process of re-definition of use of technology. Thus, we are driving fast on adventurous and bumpy road of creation of freedom.

As more people are contributing to the development of software, writing and localizing manuals, requesting new features there will be more fields of application of use of existing or newly created software packages. This scan of free audio software will meet its goal if it will attract more people to use free software.

An Overview of GNU/Linux Distributions for Audio

AGNULA – AGNULA is GNU/Linux Audio collection of software that is based on two distributions i.e Red Hat and Debian. Red Hat based collection is called ReHMuDi and Debian is called DeMuDi. It is well integrated although collection of software is more oriented towards artists than media organisations. It does have discussion list so the users may ask questions that are usually asnwered by friendly maintainers. AGNULA is free to download. http://www.agnula.org

Planet CCRMA – planet CCRMA is a collection of software packages that you can add to a computer running Red Hat 8, RedHat 9 or Fedora Core 1, 2 or 3 to transform it into an audio/video oriented workstation. It is well designed, update and installation may be done by using apt-get procedure. It does have discussion list so the users may ask questions. Members are often very skilled and accurate with advices and suggestions. CCRMA is acronym for the Stanford University Center for Computer Research in Music and Acoustics, USA. Collection of software is targeted for audio and video artists and a huge collection of software may be used in various fields of application. Many technically complicated operations such as kernel patching etc. are done automatically by using apt-get procedure on CCRMA. Obviously, CCRMA people have done a great job for the users. Planet CCRMA packages or whole CD image may be downloaded freely from the Planet CCRMA site. http://ccrma.stanford.edu/planetccrma/software/

Studio To Go - is a complete music production environment on a single CD. It is prepared by the company Fervent Software that is known for the package RoseGarden that is primarily music notation package. It is done and integrated in a way that the user does not need to do additional tweaking and configuring sofware and hardware and may be brought away since it is put on bootable CD. Studio to Go is sold with very low price and detailed information may be found at the site http://www.ferventsoftware.com

AudioSlack – Although Audio Slack is not distribution itself it deserves to be mentioned since Slackware is pioneering distribution of GNU/Linux known as stable, well integrated and conceptualized. You can find more details on http://www.audioslack.com Those who like Slackware should not miss this link and join mailing list and ask for help so free audio software may be installed and used on Slackware GNU/Linux distribution. Vector Linux is Slackware based GNU/Linux distribution that is very fast and quite usable on computers with weaker technical specifications.

Glossary of Terms:

Note: This glossary of terms is intended to help readers of this document to understand some technical terms in this document. However, audio field is rich with technical jargon and author's intention is to present reader with explanations of some basic term in order to understand easier some important points in using software and hardware in audio field

Acoustics - Acoustics is a general term describing the action of sound in a given space. It is also science that study behaviour of sound in various ambiences and architectural designs

Acquisition - Data acquisition is the process by which events in the real world are sampled and translated into machine-readable data
.

ADC – device that converts analog audio signal to digital signal. It may be a chip that is mounted on audio card, built in audio recording device or it may be a separate device

AES – Audio Engineering Society, an international organisation dedicated to recommend technical rules, standards and practises in designing audio devices, performing audio recording, playback etc. http://www.aes.org

ALSA – Acronym for Advanced Linux Sound Architecture, a system that provide audio and MIDI functionality to GNU/Linux operating system http://www.alsa-project.org

Amplifier – device that can receive low-level signals and amplify them so they can be audible by the listeners

Amplitude - The height of the wave from its highest point or peak to its zero level in any given sample or at any point along the line of the wave.

Analog signal - Analog refers to a non-digital continuous waveform signal, which may contain an infinite number of points along a frequency range.

Audio – something that is heard or having to do with transfer of audible signal to or from any device capable for that kind of activity

Audio card - device – hardware that is designed with purpose to be capable of recording and playback of audio signal in computer or other digital audio recorder device

Backend – a part of software that deals with processes set by th euser

bit – the smallest chunk of digital data. One byte does have 8 bits. Bit is used in describing resolution of recording of digital audio.
Blogging – using and administering weblogs, browser based text, audio, video publications that may be updated by the public

Cross – Fade -operation which provide smooth transition from one sound to another in a way that ones is fading out while the next one is fading in. Cross – fade is place where those two sounds are played back simultaneously and their fading out and fading in are audible as pleasant mixture of sounds

DAC – device that converts digital audio signal to analog signal. It may be a chip that is mounted on audio card, built in audio recording device or it may be a separate device.

Decibel (dB) – named after Graham Bell. Decibels are most often used to describe sound pressure level (commonly referred to as volume). However, the term is also used in various other measurements such as signal-to-noise ratio, gain and dynamic headroom. In these instances, decibels refer the measurement of signal increase or signal strength instead of sound pressure level, but the scale and basic idea behind it remains the same.

Digital signal – a stream of discrete chunks of data that are defined – quantized. Digital signal is not continuous as is analog signal. Digital signal may be easily edited and trasnmitted to other devices or networks.

Digitisation - Method of encoding analog **audio** signals into digital bits of information typically resulting in high-quality signals that suffer from very little distortion and noise

Distortion – an amount of unwanted change of signal. It is assumed that distortion should be minimized so the signal may be intact.

DSP – acronym for Digital Signal Processing that describe various techniques of processing of digital signal

Dynamic range is the ratio of the loudest (undistorted) signal to that of the quietest (discernible) signal in a piece of equipment or a complete system, expressed in decibels (dB)

EBU – acronym for European Broadcasting Union that is professional association of broadcasters. http://www.ebu.ch

Ethernet - (name comes from the physical concept of ether) is a frame based computer technology for local area networks (LANs). It enables distribution of data in large inter-networks

Fade In - a smooth increase of loudness of sound from the certain point until th epoint where it reaches its normal loudness

Fade out - a smooth decrease of loudness from the point where loduness started to decrease from its normal value to the minimal one.

Fire-Wire – a standard known as IEEE1394 which define fast transfer of data between two digital entities that may be hard disk, camera, audio interface etc. It is very fast and reliable and it becomes quite popular in everyday use of computers

Free software - software that is written with idea of freedom in mind. Free software is free to study, modify, distribute, share with others. It is licensed with idea that none is allowed to stop freedom of users to study, modify, distribute and share it

Frequency range – a range within which may be audible low, middle or high frequencies. It describes capability of audio device or software package to reproduce or record frequencies within a given range

Frequency band – a small frequency range that is used to be defined by its borders and edited by software or hardware capable of changing its characteristics

Frontend - a part of software that deals with user himself/herself allowing him/her to change parameters, visualize data, manage software features

Gain - Gain is simply the increase of a signal. Amplifiers measure gain in decibels (dB). An amplifier's gain is derived by dividing the output created by the amplifier by the input given to the amplifier (output over input). The gain of an amplifier describes the amount by which it increases the output of an incoming signal.

GPIO – acronym for General Purpose Input Output interface that is user configurable. It is used for various devices communication and control of transfer of data. Due to its flexibility almost any peripheral device may be interfaced and controlled with GPIO.

Headroom - ratio of the largest undistorted signal possible through a unit or system, to that of the average signal level. For example, if the average level is +4 dBu and the largest level is +26 dBu, then there is 22 dB of headroom.

LADSPA – acronym of Linux Audio Developer's Simple Plugin API which enable numerous plug-ins to behave as integral part of certain audio software application

Loudspeaker – electromechanical device that translate electric signal into sound by accepting electrical signal in its voice coil which cause its cone to move/vibrate thus create sound

Low – latency - allows human-unnoticeable delays between an input being processed and the corresponding output providing real time characteristics.

Microphone – electromechanical device which at its heart does have a wire coil which moves within a magnetic field by being pushed by air pressure caused by source of sound. Move of that coil develop voltage and current that is transferred through the microphone cable.

MIDI – acronym for Musical Instrument Digital Interface that is communication protocol between two musical instruments or digital devices that may be controlled by using MIDI control messages to each other

Mixdown – Mixdown is resulting file of rendered existing parts of any recording that should be and seamlessly made as a whole

Noise – unwanted disturbance of signal that sometimes do have unpleasant audible appearance

Normalization – process in which user define maximum value of amplitude which is very useful when all recordings should have the same amplitude level which makes them easy to listen

Nyquist-Shannon theorem - when sampling a signal (e.g., converting from an analog signal to digital), the sampling frequency must be *greater than twice* the bandwidth of the input signal in order to be able to reconstruct the original perfectly from the sampled version.

Patchbay – a device that is consisted of panel and connectors attached to it so various devices may be hooked up and their signal may be transmitted between devices connected to that patchbay

Plug -in - A *plug-in* is a software module that adds new functions or extensions to a host application, without the need to recompile the program. The plug-in functions are stored within a shared object file (*.so in Linux, *.dll in Windows) that may contain one or several plug-ins.

Quantizing is the process of determining which of the possible values (determined by the number of bits) is the closest value to the current sample of incoming signal – hence assigning a quantity to that sample. Quantizing,by definition then, involves deciding between two values and thus always introduces error.

Real -time audio In a real-time Digital Signal Processing (DSP) system, the analyzed (input) and/or generated (output) samples (whether they are grouped together in large segments or processed individually) can be processed (or generated) continuously in the time it takes to input and/or output the same set of samples independent of the processing delay. Consider an audio DSP example: if a process requires 2.01 seconds to analyze or process 2.00 seconds of sound, it is not real-time. If it takes 1.99 seconds, it is (or can be made into) a real-time DSP process.

Sampling rate – a frequency of taking and defining samples within one second during the process of digitisation. It is measured in Hz (hertz)

SCSI - acronym for Small Computer System Interface which is parallel interface for attaching computer peripheral devices such as hard disk, archiving tape, printer. It provides fast data transmission and it is popular in professional use

Segue - Segue is smooth proceeding from one tune into another tune or voice so the sounds are smoothly interwoven making transition from one sound to another pleasant to listeners.

Signal – a flow information that can be represented as mathematical function

Software synthesizer – a software package capable of generating -synthesizing sound according to parameters set by the user

Sound Pressure Level (SPL) - a measure of the strength or intensity of a sound, expressed in decibels (abbreviated dB). The sound pressure level generated by a steady source of sound will usually vary with both distance and direction from the source.

Stereo - Two channels of audio information recorded and played back in such a way as to recreate a sound stage giving depth and breadth to audio reproduction. Stereo systems use a left speaker and a right speaker with their own, independent amplification and signals. The two speakers operate together to give reproduced sound a sense of spaciousness.

Streaming – process of delivery- distribution of various content (audio, video, multimedia etc.). The content is distributed by using various network protocols. In highly computer networking environments streaming is becoming more present and important for the future development of distribution of various content.

Threshold – point above which or below which certain function may be applied

Telephone hybrid – device that enables telephone signal to be sent to audio devices so the telephone signal may be heard in studio environment

VST – acronym for Virtual Studio Technology, a plug-in architecture defined by the Steinberg corporation (acquired by Yamaha) that enables various plug- ins to be used in audio software for Mac and Windows platforms. The latest developments enabled users of GNU/Linux to use VST plug-ins as well.

Useful resources:

AES - Audio Engineering Society http://www.aes.org
AGNULA – http://www.agnula.org
ALSA - http://www.alsa-project.org/
Ardour – http://ardour.org
Audacity - http://audacity.sourceforge.net/
Audio Science Inc. - http://www.audioscience.com
Audio Slack – http://www.audioslack.com
Cirrus Logic – http://www.cirrus.com
Connexions – a collection of free scholarly materials http://cnx.rice.edu/
Ecasound – http://eca.cx
EBU - European Broadcasting Union http://www.ebu.ch
Egoboobits – http://www.egoboobits.net
Free Software Foundation – http://www.gnu.org
Icecast – http://www.icecast.org
Jack - http://jackit.sourceforge.net/
Linux Sound – http://linux-sound.org
Magnatune – http://magnatune.com
Measurement Computing – manufacturer of GPIO cards http://www.measurementcomputing.com
MySQL – http://www.mysql.com
NAB – National Association of Broadcasters – http://www.nab.org
Opsound – http://www.opsound.org
Rane Corporation - http://www.rane.com
Rezound - http://rezound.sourceforge.net/
Salem Radio Labs – http://www.salemradiolabs.com
Studio to Go – http://www.ferventsoftware.com
Tactical Technology Collective – http://www.tacticaltech.org
Streaming Suitcase – http://www.streamingsuitcase.com
Wikipedia – http://wikipedia.org

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