Tampereen ammattikorkeakoulu

MODERN LIVE RECORDING

Multitrack recording a live band with digital live audio equipment

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Bachelor’s thesis
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Music Production

ABSTRACT

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Recording equipment have evolved drastically from acoustical and electrical recording to computer based digital audio. This thesis focuses on how digital live audio equipment has enabled one person to be able to mix a show and record a live band simultaneously. The purpose of this thesis is to give an overall view of factors surrounding live recording and then provide a practical example of how that knowledge can be put into action.

The theoretical study for this thesis involved written material such as books, e-books, web articles, patents, product brochures, manuals and websites surrounding the topic. This study consists of theoretical and practical part. The theoretical part covers briefly the history and fundamentals of acoustic and electrical recording before covering the concept of digital audio. The practical part of this thesis depicts one way of pre-producing, recording and post-producing a live band in digital domain.

The recorded band was Lemmenlautta and the recordings for this thesis took place during autumn of 2018. As the technical side of their show is a one-man operation, the simultaneous live mixing and multitrack recording would not have been possible if the signal distribution, audio signal processing and recording gear were not working in digital domain.

Key words: live performance, live recording, sound technician, digital audio

CONTENTS

ABBREVIATIONS AND TERMS

AES/EBU	Audio Engineers Society/European Broadcasting Union
Bus	Common connection of many different signals
Decibel (dB)	Audio measurement unit expressing the logarithmical ratio between two quantities
dBFS	Measurement for amplitude levels in digital systems
Fader	Volume controlling potentiometer used to adjust signal level
Frequency response	Range of frequencies that a microphone can detect or the range of frequencies an audio device can reproduce
Front of house	Location of a sound mixer in a venue
Impedance	The opposition to the alternating current in an electrical circuit
Mixing console	A device that controls the relative level of combined signals with additional tone, panning and signal routing controls
PA	Public address system. Used to amplify sound for audiences
Plug-in	Software add-ons that can be controlled with the host software
SPL	Sound pressure level
Stage plot	Upside view of how a band is set up on a stage
Transducer	A device turning one form of energy into another

1 INTRODUCTION

The preliminary idea for this bachelor's thesis came up when I started working full time as a sound engineer for a band called Lemmenlautta in the beginning of 2018. The band in question tours and performs regularly approximately 100 times a year all around Finland focusing on ballrooms and private events. I was interested in how front of house multitrack recording with digital live audio equipment is done and how would I be able to successfully record a live band only by myself. I figured that my continuous work with the band would provide perfect conditions for practicing the subject. I brought up the idea of having multitrack recordings of their gigs to Lemmenlautta during spring 2018. We agreed that I would record them during autumn and provide them with ready mixed songs during 2019.

First this study takes a brief glimpse on the history and technologies of recording from the late 19th century up until the introduction of digital media. The theoretical part of this thesis is formed on an idea of what an electrical audio signal goes through in a digital recording process and how the process is performed in digital live audio equipment. After laying down the initial knowledge of technologies in digital live equipment and recording systems, the practical part depicts one way of putting that knowledge into practice.

The practical part of this bachelor's thesis portrays the entire process of recording a live band in different venues with available gear at the time. This study will provide information that can be helpful for someone who is seeking information about how recording with digital live audio equipment can be done. This thesis can also work as a tool for someone who is considering in carrying out a similar kind of one man operated live recording project.

2 ACOUSTICAL AND ELECTRICAL RECORDING

In the latter half of the 19th century innovators all around the world were coming up with different ways of capturing sound. After the discovery of acoustical recording advancements in technology during both late 19th and early 20th century lead to the discovery of electro magnetism and transducers which made electrical recording and amplification of electrically recorded sound possible. (Library of Congress n.d.)

2.1 Acoustical recording

Thomas Edison was the first to invent a working recording machine which was able to record and reproduce sound. Edison's phonograph was a hand cranked device that grove a physical representation of sound into a rotating tin-foil or wax covered cylinder. It made use of a speaking tube or a mouthpiece and a diaphragm to move a corresponding indenting-point in vertical motion centred with the diaphragm. Cranking the device made the cylinder rotate clockwise and the indenting-point to groove the tin-foil or wax in an up-and-down motion. When rotating the cylinder, the engraved sound could be played back through a similar tube with a spring, thread and another diaphragm. The moving spring caused the thread to vibrate the diaphragm which made the recorded sound audible again. (Edison, T. 1878. Phonograph or Speaking Machine; Chanan 1995, 24.)

After Edison's invention of the phonograph Emile Berliner invented the Gramophone. Unlike Edison recorded sound onto cylinders, Berliner's gramophone utilized laterally spinning discs. A stylus-head was attached to a diaphragm by a lever which then in coordination to the soundwaves resonating the diaphragm chiselled a wax coated zinc disc laterally to create a physical presentation of the sound. Berliner's gramophone was different to Edison's phonograph by the fact that the gramophone itself could only reproduce previously recorded sound. The recorded master discs were made with another device and the recordings played back on the gramophone. To be able to multiply the previously recorded material, Berliner came up with a process of coating the recorded disc with metal and then use the reversed version, a master, of the record to press copies of the original recording. (Berliner, E. 1887. Gramophone; Library of Congress n.d.)

2.2 Electrical recording

Advancements in radio broadcasting and inventions considering electromagnetic induction let electrical recording become the norm during the 20th century. Microphones gave recording engineers more options in controlling what was going to be recorded and from where as the microphones could be remotely connected to a recording device. The basic idea of a microphone is that it turns acoustical energy into electrical energy. Electric current can be induced in a metal wire when the wire is moved in a perpendicular motion in a magnetic field. The alternating electric current is a reproduction of the sound with accurate frequency and amplitude information. (Rumsey & McCormick 2006, 42.) According to Rumsey & McCormick (2006, 155) inserting basic variable resistors and valve amplifiers to the signal chain let the engineers to control and amplify the microphone signals. Lightbulb resembling vacuum or thermionic tubes were the first inventions allowing amplification of microphone signals (Kadis 2012, 68).

In a vacuum tube a direct current voltage is run through a filament which heats up a cathode letting it to emit free flowing electrons to a plate as its negative charge attracts the electrons. Inserting third electrode, a control grid with a slight negative charge relative to the cathode enabled a small voltage to control the current to the plate. (Kadis 2012, 68.) Larger voltage compared to the control grid can be introduced in the plate which made it possible to amplify the original signal (Kadis 2012, 68).

After the First World War knowledge around electromagnetism led to inventing magnetic recording. Magnetic recorders used two separated magnets with south and north pole for creating a magnetic field and metal oxidized tape for recording media. Electrical current is fed to the first magnet creating a magnetic flux similar to the fed electric current and the tape is run between the magnets. The magnetic flux magnetised the tape and polarized the magnetic particles in the tape to represent the changes in the alternating current fed to the magnet. Later developed tape recording devices capable of recording multiple individual tracks at the same time enabled recording engineers to record sound sources separately which in turn gave possibilities in stereo mixing and better isolation of individual instruments. (Kadis 2012, 118-119; Rumsey & McCormick 2006, 156-160.)

3 MICROPHONES AND DIRECT INPUT BOX

Microphones or mics are usually the first part of the recording signal chain. Microphones are used to transfer acoustic vibrations in the air into electrical energy. There are three types of microphones: dynamic or moving coil microphones, ribbon microphones and condenser microphones. The fundamental of electromagnetic induction is the basis of all the three microphone types. How electrical energy is produced by each microphone varies between the built-in characteristics of each device. (Huber & Runstein 2014, 105; Kadis 2012, 82; Rumsey & McCormick 2006, 41-44.)

3.1 Dynamic microphone

The functioning parts of a dynamic microphone consists of a diaphragm, coil of metal wire and a magnet. The diaphragm is suspended to float in front of the magnet, the metal coil surrounds it and is attached to the diaphragm.

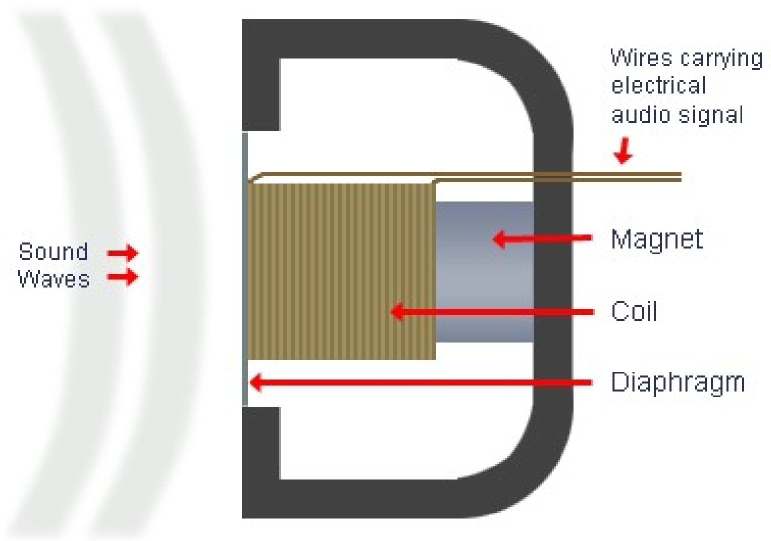


FIGURE 1. Structure of a dynamic microphone (Courtesy of MediaCollege.com)

Soundwaves reaching the diaphragm vibrates it and makes the metal coil move. When the coil of metal wire is moved in a perpendicular motion towards a magnet the magnetic field produced by the magnet creates an alternating electrical current into the wire. The alternating current has the same frequency and proportional amplitude as the soundwave that caused the diaphragm to vibrate. (Rumsey & McCormick 2006, 42-43.)

3.2 Ribbon microphone

A ribbon microphone has two magnets facing each other and a long and thin conductive metal strip or foil in between them. When the pleated metal strip is vibrated with sound in between the two magnets the magnetic field induces electrical signal in the vibrating metal strip. That electrical signal is then carried forward with connector leads attached to the metal strip. (Rumsey & McCormick 2006, 43.)

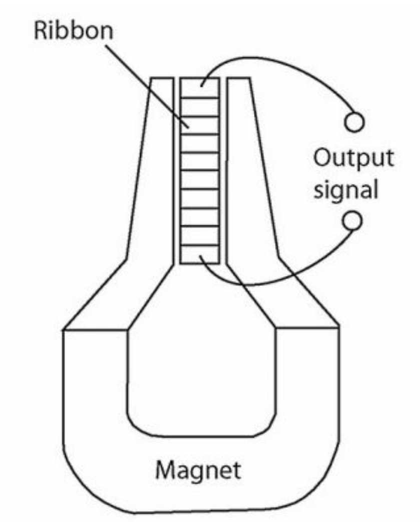


FIGURE 2. Ribbon microphone (Courtesy of ProSoundWeb)

The ribbon mic is more delicate build than a moving coil microphone as high sound pressure levels can harm the ribbon microphone (Rumsey & McCormick 2006, 45).

3.3 Condenser microphone

Condenser microphone is built a little different than a dynamic or ribbon microphone. According to Rumsey and McCormick (2006, 44) a condenser microphone has a diaphragm situated in front of a backplate which are separated by an insulator. The microphone is fed with phantom, 48 volts direct current, power and it creates an electrical capacitance in the backplate. Soundwaves reaching the diaphragm moves the diaphragm back and forth. The varying distance between the backplate and the diaphragm causes changes in capacity between the voltages created by the moving diaphragm and the phantom power fed to the microphone and results as an electrical reproduction of the soundwave. (Huber & Runstein 2014, 109; Rumsey & McCormick 2006, 44.)

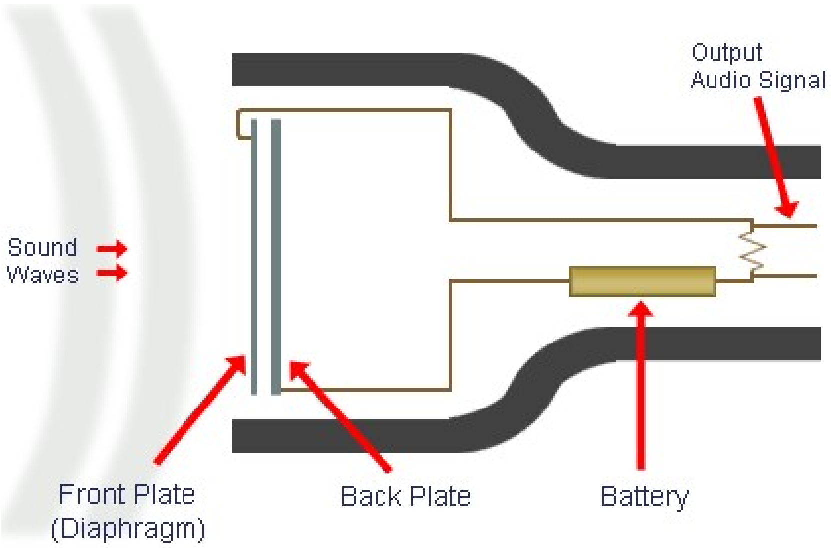


FIGURE 3. A condenser microphone with the 48v phantom power provided by a battery (Courtesy of MediaCollege.com)

The resulting signal is of high impedance and to have a correct frequency response out of the microphone a preamplifier is situated inside the condenser microphone. The preamplifier reduces signal level loss, hum and noise pickup by the microphone. As the phantom power is needed for creating the polarized voltage for the backplate and diaphragm, the preamplifier also utilizes that supply voltage. (Huber & Runstein 2014, 111.)

3.4 Line level signal and Direct input box

Instruments such as keyboards can produce signals with high voltage output which can overload the input device. If the line level source is too powerful, a DI box can be placed between the sound source and the input device. Direct input boxes transform line level signal into lower voltage microphone signal relieving the input device from overload. If the line level signal is provided with phono jack output and the input to the recording signal chain is an XLR input a DI box can be used to turn the phono jack into an XLR. Professional, well designed and manufactured DI boxes can have signal attenuation tools to give audio engineers more tools and possibilities in attenuating the electric signal. (Rumsey & McCormick 2006, 305-351.)

4 ANALOGUE AUDIO SIGNAL DISTRIBUTION

The electrical energy produced by a microphone or line level instrument must be carried from the capturing device to an amplifier, mixing desk or an independent recording device. This is carried out with a selection of analogue audio cables such as XLR or phone plug cables. Both cables and connector types are universally used with microphones and instruments with built in microphones. (Bartlett & Bartlett 2007, 6-7.)

4.1 Balanced XLR connector

The three-pin XLR balanced connection provides robust enclosing for the electrical signal and it provides balanced connections with reduced interference. Interference in analogue signals can be heard as a buzz or hum and it is worthwhile avoiding. (Rumsey & McCormick 2006, 337-343.) This connector type provides a balanced connection created with opposite phases between the second and the third pin. The XLR also has a dedicated ground pin for common earth connection between devices. The three pins are used in a way that the first pin is for ground, the second pin is for live signal and the third pin is for return signal. (Rumsey & McCarthy 2006, 341-343.)

PICTURE 1. Male 3-pin XLR connector (© Neutrik® AG)

The male XLR end of an XLR cable feeds the signal to its respective female counterpart that is either the input of a device or another XLR cable.

PICTURE 2. Female 3-pin XLR connector (© Neutrik® AG)

The female XLR connector type receives the signal and transmits the signal forward in the signal chain.

4.2 Unbalanced phono jack cable

The second previously mentioned analogue cable connector type that is commonly used in instruments and audio related gear is the phono jack cable (Rumsey & McCormick 2006, 337.) The phono plug, jack plug or TS connector is an unbalanced connector type which carries the live signal from the tip and ground from the sleeve (Bartlett & Bartlett 2007, 39).

PICTURE 3. Professional phone plug (© Neutrik® AG)

In a phono plug the tip and sleeve are separated with a black ring. As the phono jack does not have another, neutral, phase reversed signal in it, it is bound to suffer more from interference than a balanced cable when used to carry signals over longer distances (Rumsey & McCormick 2006, 336). A phono jack cable has a male plug in both ends of the cable and it is inserted into a female counterpart.

4.3 Analogue stage box and snake

Live bands often perform with multiple instruments. This results in multiple microphones and instrument signals that needs to be carried to a mixing console. Having multiple individual microphone cables carrying information for long distances is inconvenient and that is why a stage box and a snake are used to carry analogue audio signals from stage to the mixer. (Bartlett & Bartlett 2007, 18-19.)

PICTURE 4. Analogue stage box and snake (©Thomann GmbH)

The microphone cable on stage is plugged into an analogue stage box which has multiple female XLR inputs built in its chassis and each connector has a unique number. The multiconductor snake is then used to carry the multiple inserted signals from the stage box to the mixer where the snake divides into corresponding numbered male XLR connectors. The male XLR connectors can then be connected to the mixing console’s inputs. (Bartlett & Bartlett 2007, 18-19.)

5 FUNDAMENTALS OF DIGITAL AUDIO

The age of digital media is the biggest change in history of communication by far. How digital data is brought to life, combined and distributed has given people more ways to handle information that was not as easily possible with analogue equipment. The capabilities of personal computers have given the ease of storing, editing and mutilating digital audio to a point that it was not possible before the age of digital media. (Huber & Runstein 2014, 199.) The following section introduces terms and processes that are crucial factors in digital audio signal processing.

5.1 Analogue to digital conversion

The electrical current produced by a microphone must be converted from analogue electrical signal to a digital form of representation for the computer to be able to handle the analogue information in sound that the microphone captured. This is done by having an analogue to digital converter or A/D converter between a microphone and a computer. (Rumsey & McCormick 2006, 193-200.) According to Rumsey and McCormick (2006, 200) the A/D converter measures the amplitude of analogue information at given time and quantises it to a digital binary number at each measurement.

The analysed data has detailed information of the fed electrical signal regarding frequencies and amplitude in a digital form of representation enabling a computer to read and process the formed digital audio signal. To get the desired sound quality out of the recording the A/D converter needs to be technically competent enough and operating with specifications that enable it to accurately analyse the signal. The specifications and technical qualities of the A/D converters play a significant role in determining the sound quality of the recording. (Rumsey & McCormic 2006, 200-201.) As Rumsey and McCormick (2006, 201) put it once the audio is converted it can never be made better afterwards.

5.1.1 Sample rate or sampling frequency

One key factor in A/D conversion and audio sampling is how many samples are taken of the original transducer produced electrical waveform. According to Rumsey and McCormick (2006, 201-202), the sampling process can be considered as taking still images of the induced electrical waveform with regular frequency and when put in sequence they form a digital waveform. Analysing the fine details in the electrical signal the rate of samples needs to high enough to gather enough data of the waveform. How fast the sample rate needs to be is determined by the highest frequency that is wanted to be recorded. According to Nyquist's theorem the sample rate or sample frequency needs to be twice as high as the highest sound frequency that is wanted to be recorded. (Rumsey & McCormick 2006, 201-202.)

Sampling frequency determines the bandwidth of audio which will be analysed by the A/D converter (Rumsey & McCormick 2006, 206). Generally audio frequency bandwidth is considered to top around 20 kHz. This would imply that the highest sampling frequency needed would be little bit higher of 40 kHz. (Rumsey & McCormick 2006, 206.) According to Rumsey & McCormick (2006, 206-207) there are two standardized sampling rates considering high quality audio work, the compact disc sample rate of 44.1 kHz and 48 kHz considered as so-called 'professional' sampling frequency. Higher sampling rates than the previous two can give better results if the recorded audio is to be stretched and modified within a DAW (Mayzes 2019). Considering hard disk storage space, higher sample rates generate more data as more audio information is analysed. For example, the 44.1 kHz sample rate produces 10 percent less data than the 48 kHz sample rate. (Rumsey & McCormick 2006, 207-208.)

5.1.2 Bit rate and dynamic range

When the A/D converter is fed with the analogue audio signal it, as previously discussed, analyses the signal in certain frequency and assigns binary value for the measured points in the signal (Rumsey & McCormic 2006, 200-201). The number of bits forms a resolution for the analysed signal and in respect to the sample frequency the A/D converter quantizes the samples to the closest quantization interval. According to Rumsey and McCormick (2006, 209-211) the bit rate needs to be high enough to accurately analyse the audio data and to avoid unnecessary digital distortion. This in turn improves the signal-to-noise ratio of the recorded signal and maintains fidelity in the recorded sound (Rumsey & McCormick 2006, 208-209).

Bit rate is an important factor in both accuracy in analysing audio and determining the dynamic range of a digital recording. Different bit rates provide different dynamic ranges on a decibel scale. (Huber & Runstein 2014, 208.) According to Kefauver & Patschke (2007, 29) one binary bit represents in signal-to-noise ratio approximately six decibels of dynamic range. Knowing the dynamic range one bit holds, the digital audio system can be set up to operate with the right bit rate to suit the recording. The higher the bit rate is the more steps there are for the analogue to digital converter to quantize the audio samples to and prevent overloading the converter. Overloading the converter with high signal levels leads it to run out of possible sampling bits and this results in hard-clipping, severely distorting, the waveform. (Rumsey & McCormick 2006, 214.)

Bit rate of 16 has been considered the industry standard for many years as it provides a good signal-to-noise ratio exceeding 90 dB (Rumsey & McCormick 2006, 214). There are applications of rates exceeding 16 bits offering wider dynamic range and those can be utilized as well. Higher bit rates can be necessary especially in live recording situations where unexpected high peaks of audio may occur. (Rumsey & McCormick 2006, 214-215.)

5.2 File sizes and formats

Multitrack audio can consume a lot of hard disk storage space. Changes in track count, sample rates and bit depth will result in varying file sizes of recorded digital audio. For example, one hour recording of two individual tracks with 44,1 kilohertz sample rate and a bit depth of 16 bits needs 606 megabytes of storage space. If the sample rate and bit depth of the same recording was 96 kHz and 24 bits, the resulting need for storage space would have been 1,9 gigabytes. Before recording digital audio, the recording engineer needs to consider the file sizes and storage space to have enough storage capacity for the recording. (Bartlett & Bartlett 2007, 51.)

Professional digital audio workstations and recording gear are built to work with raw, uncompressed audio data (Kefauver & Patschke 2007, 63). Uncompressed audio formats have all the information left in the files without having lost crucial data in any file compression process. Apple’s Audio Interchange File Format, AIFF (.aif) and The Microsoft Windows format WAVE (.wav) are two examples of file types that provides lossless raw audio. Both sound file formats are supported by most digital audio workstations and both provides the user to have extra information embedded to the sound file. (Huber & Runstein 2014, 247; Kefauver & Patschke 2007, 63.)

6 DIGITAL AUDIO SIGNAL DISTRIBUTION

Digital live audio equipment offers possibilities in dividing the signal processing chain amongst different devices. Those devices can be situated physically in separate locations when connected with cables supporting digital audio data and they remain virtually in the same space and time. For the signal chain to work correctly, all the digital audio devices need to be synchronized together to work as one unit. (Yamaha n.d. Mixers; Kefauver & Patschke 2007, 19.)

6.1 Synchronization

Parts of the digital signal distribution chain can be set up separately and be connected with cables to each other. To have all the parts of the system work as one unit they need to work in same time in relation to each other. This is achieved by introducing word clock to the signal chain. Word clock is a constant integrated signal telling the devices in the signal chain what is the operational sampling rate of the digital audio system (Kefauver & Patschke 2007, 22). If the separate devices in the signal chain fall out of synchronization, the desynchronization causes the devices to work in different sampling times. This error in sample timing is called jitter (Kadis 2012, 145).

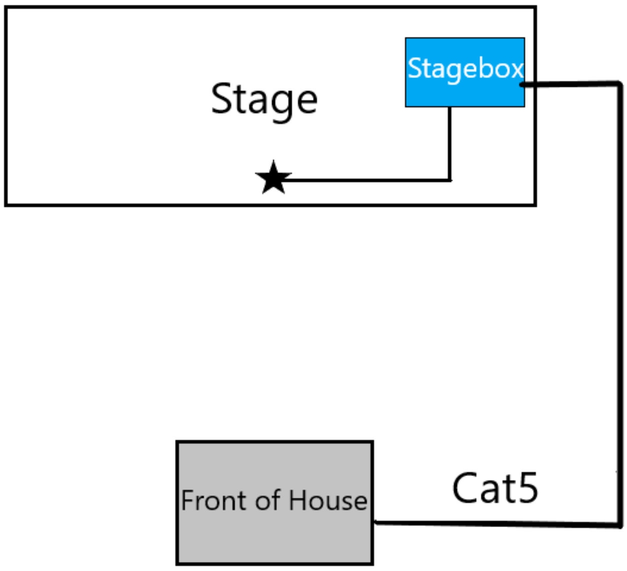


FIGURE 4. Bird's-eye view of front of house live recording setting implemented in the Lemmenlautta live recordings (Piironen 2019)

The figure above shows a simplified view of the recording setup which was implemented in the practical part of this thesis. The star symbols a microphone and its connection to the stage box. In this example, the analogue to digital conversion of the microphone signal takes place in the stage box and is then carried to the front of house along a data cable. The connection between the front of house and the stage box was established with Cat5 (Category 5) cable. The stage box provided AES50 digital audio protocol which carries the word clock synchronization signal in the data stream. The integrated word clock signal eliminates a need for a separate device or a separate cable for synchronizing the devices which in turn is very convenient in live show situations. (Behringer n.d.)

6.2 Digital live audio protocols

Different manufacturers prefer certain digital audio protocols over others. Some professional audio equipment manufacturers provide the possibility in choosing which protocol the end user wishes to use with modular audio cards. Different protocols travel along different cables and use variations of connectors depending on the fundamental of the protocol. This section covers a selection of protocols used in digital audio distribution systems, but it does not go into detail how individual protocols are digitally structured. (Behringer n.d. X32 Brochure; Digico n.d. I/O Modules; Yamaha n.d. Interfaces.)

6.2.1 ADAT

ADAT or ADAT Lightpipe was developed by Alesis in 1991 for their digital audio tape machine (Robjohns 2007). ADAT can transmit up to 8 channels of audio with the sample rate of 48kHz between audio equipment with a standardized Toslink optical cable. ADAT is unidirectional signal from one output to one input of a device. (Robjohns 2007; Rumsey & McCormick 2006, 314.)

PICTURE 5. Toslink optical cable (Courtesy of Amazon.com)

To have information moving between both devices, two Toslink optical cables are needed for connecting output and input ports of both devices (Robjohns 2007). The length of recommended connection can introduce restrictions for applications as ADAT is not recommended to be used with a connection exceeding 5m (Rumsey & McCormick 2006, 314).

6.2.2 AES3 & AES50

Versions of AES/EBU (Audio Engineering Society and the European Broadcast Union) digital audio standards are used in signal distribution devices. The AES3 protocol uses the same type of three-pin XLR connector as the analogue audio signal utilizing the second and the third pin for data and the first pin for grounding (Robjohns 2007). Although the connector is the same than in analogue audio signal cable, it should not be confused with the analogue XLR cable as the cable impedance needs to be different for AES3. The AES3 protocol calls for the cable to provide 110 ohms of resistance for best results. (Robjohns 2007.)

The AES50 is an open source audio protocol widely used in modern digital signal distribution devices. The protocol travels along a category 5 or category 5e (Cat5 or Cat5e) cable fitted with a RJ45 connector. The AES50 is different from AES3 in multiple areas.

As the AES3 only provides two channels of digital data along single cable interconnection, the AES50 protocol has multitrack capabilities as it can provide a high channel count and minimal latency between devices. The AES50 can also provide security of connection with the possibility of two simultaneous connections between devices. (Walker 2011.)

PICTURE 6. RJ45 connector inside a metal casing (© Neutrik® AG)

6.2.3 MADI

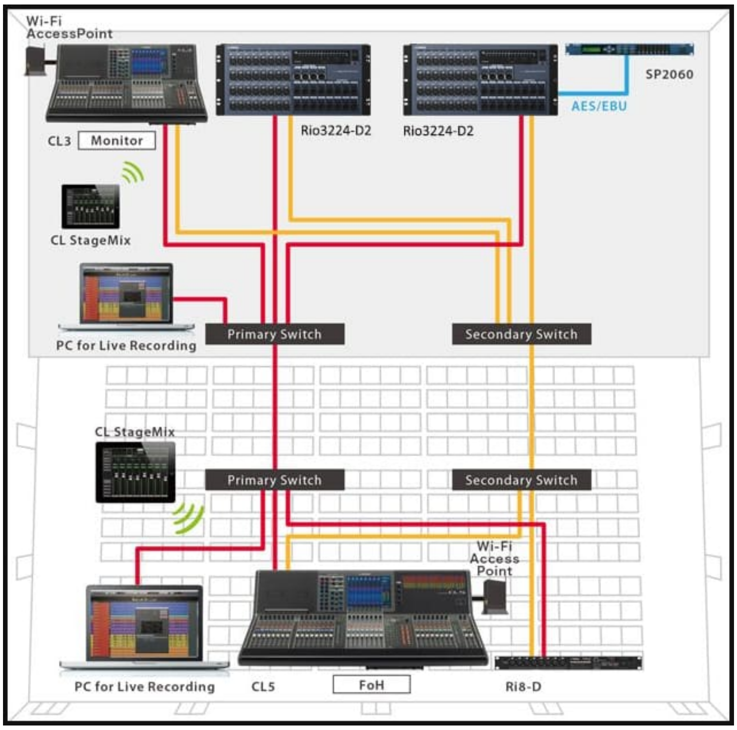
MADI (Multichannel Audio Digital Interface) protocol is clutter free AES standard protocol for transmitting multi-channel audio data (Huber & Runstein 2014, 211). First introduced in 1991 as AES10 it was capable of providing 56 channels of audio along one coaxial cable terminated with BNC connectors.

PICTURE 7. Coaxial cable fitted with a BNC connector (© Neutrik® AG)

Later the protocol has been developed to handle up to 64 channels of audio with the sample rate of 48Khz. (Robjohns 2007.) Like ADAT, MADI information travels to one direction. That is why it is common to have two coaxial cables in MADI system. One from the input device's MADI out and one from the MADI interface's output to the input device's MADI input. (Digico n.d. SD-Rack.) Different from other AES standards MADI does not carry word clock information in the signal. Here the two coaxial cables handle the word clock issue as the MADI interface will provide the word clock along the secondary coaxial cable to the input device. (Robjohns 2007.)

6.2.4 Dante

Dante audio network is a multichannel digital media protocol designed by Audinate. Devices running Dante can be connected with the same Cat5 or Cat5e cable as the AES50 and it can also be connected with category 6 or fibre optic cables (Audinate). Dante makes use of standard networking switches and connection over IP-network which enables easy scaling from simple setups with couple of channels to a network of thousands of channels (Audinate).



PICTURE 8. An example of a Dante based digital audio distribution system (©Yamaha Corporation)

Differing from the AES50 protocol which is a point-to-point connection between devices, Dante network enables different devices supporting the audio network being connected to it via simple Cat5 connection and not needing to have a physical connection to the input device. (Audinate; Yamaha n.d. Mixers.)

7 DIGITAL FRONT OF HOUSE LIVE RECORDING

Digital live audio equipment gives possibilities in configuring systems to fit the purpose of what the system is used for. Different professional live audio equipment manufacturers, digital audio protocols and devices are designed and suited for different kinds of system designs and needs of operation. (Behringer n.d. X32 Brochure; Yamaha n.d. Mixers.) Here discussed topic is focused on front of house live recording.

7.1 Digital stage box

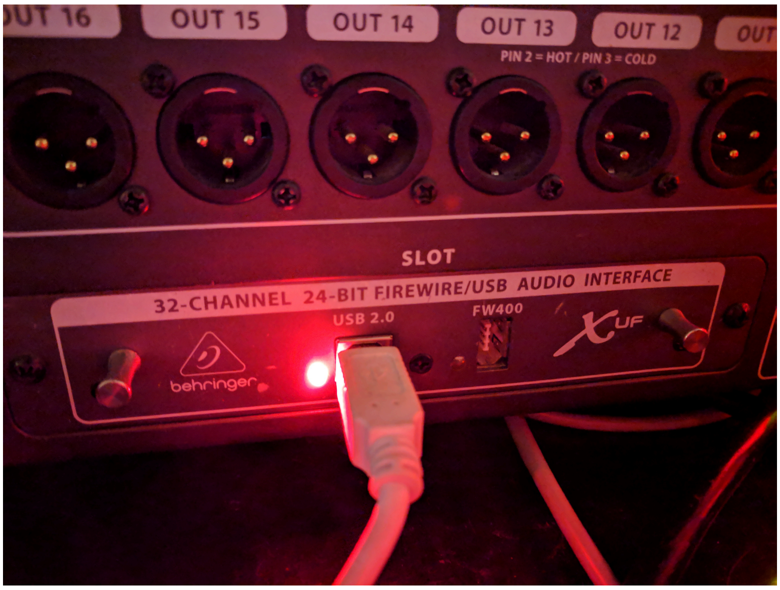
In live audio equipment, the microphone is first connected to a stage box which withholds a pre-amplifier and an A/D converter in it. The pre-amplifier enables the microphone signal to be amplified to a suitable level for the A/D converter to process the analogue audio signal into its digital form of representation. (Yamaha n.d. Interfaces.)

PICTURE 9. Behringer S32 digital stage box (Courtesy of Thomann GmbH)

The stage box packs the digital audio data into its designed digital audio protocol. Then the digitalized audio signal is transmitted forward with a digital data cable to a digital mixer. (Yamaha n.d. Mixers.)

7.2 Mixing console as an audio interface

An audio interface can be an in-built feature of a live mixing console and work as a connectivity bridge between the mixer and a computer (Huber & Runstein 2014, 241). The interface enables the user to connect a computer to the mixing console with a single USB or FireWire connection. A live mixing console can also provide possibility to record two channel “board mix” directly into a multimedia storing devices, such as USB-flash drives without an external recording device. (Behringer n.d. X32 Brochure.) Depending on the console manufacturer the number of recordable tracks vary (Behringer n.d. X32 Brochure; Yamaha n.d. Mixers). A record of a board mix can end up being less accurate than a multitrack recording. For example, individual balances between instruments and vocals can be off as the vocals need to be louder in a PA mix than instruments. (Bartlett & Bartlett 2007, 30-31.)



PICTURE 10. Audio interface of Behringer X32 live mixing console (Piironen 2018)

In the practical part of this thesis the live mixing console, Behringer X32, was connected to the computer running the recording software with a USB 2.0 cable. The cable was connected to the Behringer X32 live mixing console’s 32-Channel 24-bit Firewire/USB Audio Interface which enabled multitrack recording in the recording software.

7.3 Digital Audio Workstation

Digital audio workstation or DAW for short gives access to record, edit and manipulate recorded digital audio. Digital audio workstations enable the user to have centralized control of every single detail of the digital audio recording process. As everything is working in the digital domain, instant saving and recalling makes the recording process a whole lot easier. What DAW one can use depends on the computer and the operating system one is working with. Most digital audio workstations are supported by all computer manufacturers and operating systems except for Apple's Logic which can be operated only on Apple based Macintosh computers. (Edstrom 2011, 39-40; Huber & Runstein 2017, 228-229.)

General and project settings within a digital audio workstation gives its user options to configure the sample rate, bit depth and audio processing speeds also known as buffer size within the DAW (Mayzes 2019). As previously discussed, options regarding sample rates and bit depths determine the bandwidth and resolution of the analysed audio. Third option in a DAW is the time the computer is allocated to process the audio. This is called the buffer size. (Mayez 2019.)

Unlike sample rate and bit depth, buffer size does not affect the quality of the recorded digital audio. It affects the monitoring delay of a sound source in the DAW. (Mayez 2019.) According to Mayez (2019) the smaller the buffer size is the less time the computer is given to process the fed data, and this results in as little monitoring delay as possible. This convenient in situations when the signal can only be monitored through the digital audio workstation like direct input signals. Higher buffer sizes are needed when the DAW is processing more tracks at the same time. This is the issue when the audio is mixed. The computer requires more time to process all the digital audio information at once, that is why a bigger buffer size is required. (Mayez 2019.)

8 PRE-PRODUCTION

Planning and preparing the live recording ahead helps in getting the job done smoothly. Talking with the band members about the recording, channel lists, stage layout plots and block diagrams help in planning a successful recording session. The recordings are easier to conduct, and all the details are easier to handle when every part of the recording system us planned out beforehand. (Bartlett & Bartlett 2007, 45-55.)

8.1 Venue selection

If it is possible to select where the live recordings are conducted will help in planning a recording with good results. A large space with a lot of reverb can turn out to sound muddy with a live band. Preferably the recordings would be best to have in a venue which has good acoustics and a large stage. Larger stages help in reducing leakage from other instruments to dedicated instrument microphones. If it is possible to go to a possible recording location or locations and see if they are suitable for recording, it will help planning the recording sessions. (Bartlett & Bartlett 2007, 45.)

8.2 Gear, channel list and block diagram

It is good to have a clear picture of what is needed for the recording and a have a plan of how instruments are situated on the stage to plan out cable runs for instruments and audio equipment. Listing gear helps to check if a crucial part of the signal chain is missing and needs to be acquired before the recordings. (Bartlett & Bartlett 2007, 45-55.)

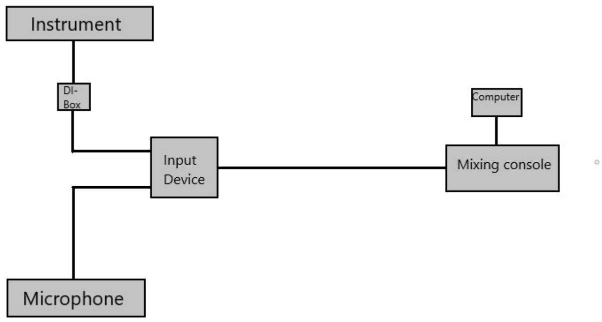


FIGURE 5. Example of a block diagram (Piironen 2019)

Based on the block diagram it is possible to list all the gear needed for the recording session. The list should include all the microphones, microphone cables, microphone stands and booms, instrument cables, direct boxes and the gear related to digital audio distribution system. Writing down a block diagram can also help in determining the amount of power outlets needed for all the instruments and audio gear. (Bartlett & Bartlett 2007, 45-55.)

Pre-planned channel and microphone list holds useful information to have an organized view of all the channel inputs. The list can be laid out as that it holds information of dedicated input channel number, name of an instrument, what microphone or direct input box will that sound source have, does that input source need phantom power and preferably also a note of what kind of stand or mount the microphone will have. (Bartlett & Bartlett 2007, 45-55.)

CH	Instrument	Microphone	+48v	Microphone stand
1	Vocals	Shure SM 58		Tall boom

FIGURE 6. Example of a channel list (Piironen 2019)

8.3 Preparing the band

Before the recording, it is a good idea to have a talk with the band about the recording. When the players know that they are being recorded they will focus on their playing. It is also good to have a talk about the songs that are going to be recorded to avoid extra prolonging in the songs. The band members playing string instruments or drums might want to change strings or drumheads and possibly check the settings and tuning of their instruments before the recording to have the best possible performance and sound out of their instruments. (Bartlett & Bartlett 2007, 45-46.)

9 POST-PRODUCTION OF A MULTITRACK LIVE RECORDING

The gig is done, the multitrack recording did not face any problems and the live performance has been captured on a hard-drive in digital format. The next step after the recording is post-producing the live show in a digital audio workstation.

9.1 Editing

When the recording is done directly to a digital audio workstation there is no need to import the audio files to another project file within the digital audio workstation. The recorded songs can be separated with splitting the individual tracks at song change points with editing tools in digital audio workstation. When each song is as separated from one another it is possible to add fade-ins and fade-outs to the audio tracks and cue audience reaction tracks with better timing to the beginnings and endings of the songs. Track slicing tools also give the option to delete unwanted material in the recordings. For example, the recorded vocal tracks might have excess talking in them or some microphones might have experienced rough handling causing audible thumps which are not wanted on the final recording. (Bartlett & Bartlett 2007, 71-76.)

9.2 Mixing

How to perfectly mix a multitrack recording and viewing all the aspects and factors related to that process is out of the scope of this thesis. Plenty of books, articles and videos have been made about how to mix multitrack recordings so further reading to the subject is advised. Live recordings how ever have certain characteristics and challenges which can be covered here.

A good way to start the mixdown process of a live recording is to seek for the loudest part of the recording and set the fader levels between individual instruments so that the overall stereo mix does not peak over around -3 dBFS when the master fader is set at 0 dB. As the track count increases and there is more audio information, the lower the individual faders are set to prevent overloading the master stereo bus. (Bartlett & Bartlett 2007, 77.)

Writing automations for each fader can be used to change fader values automatically as the song goes on along the timeline. This is useful when balances between instruments is wanted to be changed for example during solos or when the audience reaction tracks are wanted to be more audible. (Bartlett & Bartlett 2007, 77.)

During the live recording the band members and instruments were situated in different locations on the stage. Those positions can be matched in the mixdown with panning. Each track can be panned from left to right into the same direction in the stereo image so that it matches the layout of the band. There are many schools of thought about how to pan instruments in a stereo mix but finding a good balance with instruments in left, centre and right positions can be used as a guideline here. Conventionally the kick, snare drum, bass and vocals are aligned to the centre in the stereo field. (Bartlett & Bartlett 2007, 77.)

The recorded live tracks usually contain unwanted background noises, leakage from other instruments that were situated on the stage and possible electrical buzzes or hums that were present in signals during the recording. Equalisation, later also EQ, and noise gating tools can be used to reduce unwanted characteristics or leakage in the signal. Equalisation can be used to filter out background noise and leakage. Using filters such as high and low-pass in the equalisation tool, unwanted sounds can be reduced around the frequency spectrum of each instrument. For example, the recorded vocal track can have bass leakage so to reduce that leakage the EQ can be set to cut frequencies ranging below 100 Hz. Equalisation can also be used to alter and compensate tonal balance in recorded signals. (Bartlett & Bartlett 2007, 77.)

Noise gate can be used to reduce leakage and buzzes or hum in audio tracks. Noise gate work as the name makes it sound like. It allows only dedicated signals with certain amplitude to pass through a pre-set threshold. When the signal is lower than the amplitude required to break the threshold, the channel stays muted. Noise gate is useful in eliminating buzz in electric guitar channels and reducing leakage in drum tracks such as toms and kick drum. (Bartlett & Bartlett 2007, 77; Huber & Runstein 2014, 490-491.)

Dynamic processing and attenuation tools like compressors can be used in the mixing process to control dynamic changes within the recorded audio. Especially vocal tracks can have extreme changes in signal levels in different parts of the song. These changes can be controlled by having a dynamic processor in the vocal track. (Bartlett & Bartlett 2007, 77.)

10 RECORDING LEMMENLAUTTA

I got the idea of recording Lemmenlautta's live performances when I joined the band as their regular sound technician in the beginning of 2018. I figured that it would be very good practical training of live recording shows as I would be regularly working with the band. During spring that year I brought up the idea of having multitrack recordings of their live shows to the band. We agreed that I would record their live shows during autumn 2018. The recordings were done all around Finland in different venues with the gear that was touring with the band as their live sound reinforcement system. I recorded almost every Lemmenlautta's live show during autumn starting from four gigs performed in Viking XPRS in September to the last show of the year in Imatra in December. The total of successfully recorded shows was 11 and the total amount of audio data recorded was approximately 308 gigabytes.

10.1 Lemmenlautta

Lemmenlautta is a Finnish dance music band formed in 2015. They tour all around Finland focusing on ballrooms and private events. The band consists of six members: a drummer, bass player, guitarist, keyboardist, accordionist and vocalist. During the last couple of years, they have gained active fans all around Finland and they have succeeded in branding themselves as a fresh breeze in the Finnish dance music scene.

PICTURE 11. Lemmenlautta (Viola Virtamo 2018)

Lemmenlautta has released two original songs to music streaming platforms and they have also released original music videos and videos of cover songs on Youtube video streaming service. Their original songs have strong influence from pop rock music with a hint of traditional Finnish pop tunes. (Magnum live, Facebook, Youtube, Spotify.)

10.2 Setting up the show

The technical side of a Lemmenlautta live show is a one-man operation. My work with the band involves loading in the gear from a van to the venue. Setting up the sound reinforcement system loudspeakers, subwoofers and stage monitoring and connecting the public address loudspeakers to their respective amplifier channels with long reaching loudspeaker cables. After the PA system is in place I go on to setting up the front of house live mixing console, lighting controller and playback device.

When the front of house is set up I continue to pull the category 5 digital snake data cable from the on-stage digital stage box to the front of house live mixing console. Also, power cord for the devices and DMX cable for lighting controller is needed. Normally at this point the band members have set up their instruments and I can place microphones to the instruments and connect them to the digital stage box according to a premade channel list that is situated close to the stage box.

PICTURE 12. One type of a stage layout with Lemmenlautta (Piironen 2018)

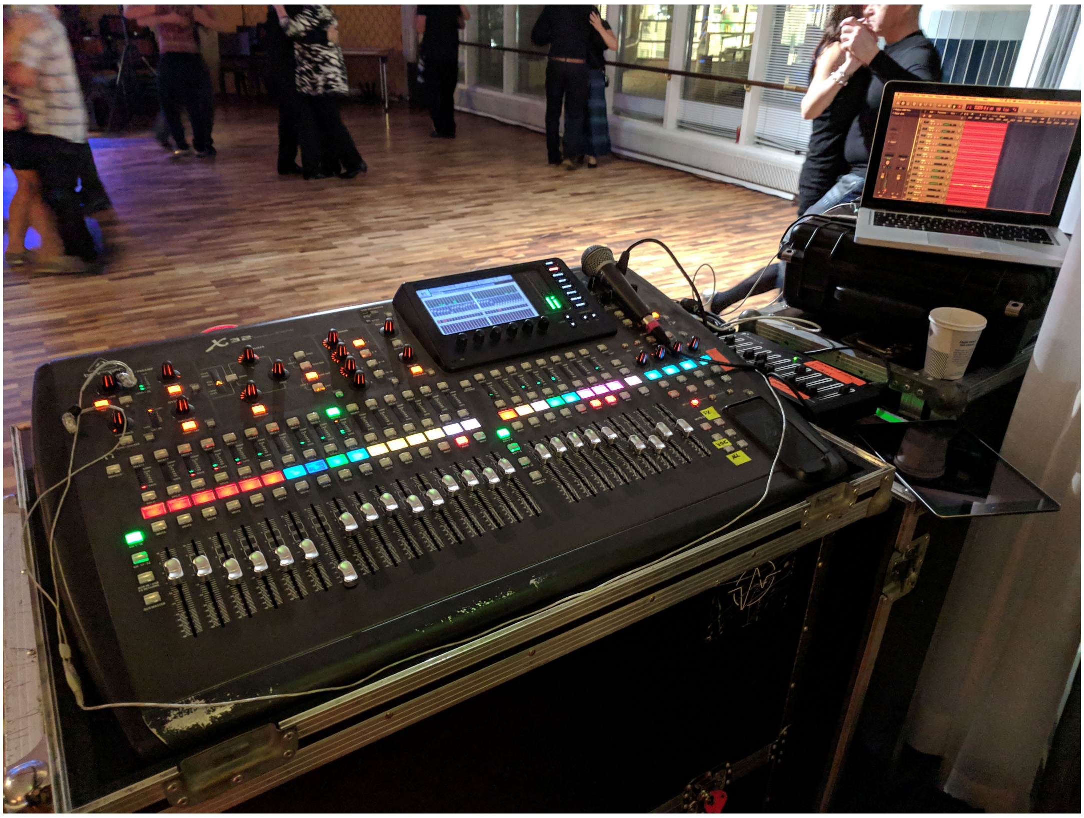
When the equipment is connected and powered up we have a sound check to check that all the signals are functioning properly, all the microphones and signals sound good and to hear what the PA system sounds like in the venue. After the sound check is done usually there is a small brake between the sound check and beginning of the show. Normal show with Lemmenlautta consists of four 45-minute sets of music.

10.3 The front of house live recording system

The live recordings were implemented with gear that was already in use as parts of the public address and amplification system for the band. The mixing console in question was Behringer’s X32 digital live mixing console which was connected to a S32 digital stage box with a category 5 data cable. The mixing console had a 32-channel 24-bit Firewire/USB audio interface expansion card installed which enabled an USB connection between the mixing console and the computer.

The digital audio workstation that I used to record the audio was Apple's Logic Pro X. I had no particular preference over other digital audio workstations with the Logic Pro. I simply had previously acquired it on my laptop and I knew how to operate all the functions in it. The Logic Pro recording sessions were saved to an external hard drive for two reasons: storage space and reliability.

As the file size of one individual show would exceeded 30 gigabytes the in-built hard drives of the computer would fill up very fast with multiple recording sessions. It is always good to be on the safe side with storage space and have big enough hard drive for the recordings. External hard drive also secures the recordings if the computer decides to die. In case that happened, I would have had the recordings stored outside the computer. I organized all the recordings in different folders and naming them according to the place and date of the recording. Each recording session was saved in different folder to prevent mix-ups between the recorded shows. Later I would go through all the recorded material and vet out the best performances.



PICTURE 12. Lemmenlautta front of house live mixing and recording setup (Piironen 2018)

The picture above shows the Lemmenlautta live show recording setup in the front of house. It consisted of the live mixer, headphones, a computer running the digital audio workstation, a 12-channel DMX lighting controller, a playback device for playing background music in between sets and the mandatory cup of coffee to fuel the show.

10.4 Microphone techniques used in the recordings

It is a common practice that audio engineers choose their selection of microphones in accordance to how and where the transducers will be used. One might select a certain microphone to be used in a studio recording and another microphone for recording a live performance and amplifying an instrument or other sound sources. The sound engineer needs to take into consideration how the selected microphones will work with public address systems and which microphones will give out the best possible results considering both recording and reinforcement. (DPA Microphones)

10.5 Close-miking

Close-miking technique is a way of enabling the highest sound pressure level (SPL) of given instrument or sound source to be captured in the greatest amount. At the same time it is made sure that the SPL of other surrounding sound sources are captured the lowest. When employing microphones with amplification systems microphone feedback can be a nuisance. Close-miking prevents feedback as the individual sound source is close to the microphone and the microphone does not need a lot of gain for the sound source to be picked up correctly. Close-miking enables to separate sound sources when they are played live in the same space. (DPA Microphones; Bartlett & Bartlett 2007, 62.)

I used close-miking technique practically for every single instrument on-stage in the live recordings. The drumkit which consisted of a kick drum, snare, two tom-toms, hi-hat, ride and two crash cymbals had individual microphones close to each drum, hi-hat and ride cymbals and also accordion and vocals were close-miked. Each microphone used in close-miking were equipped with cardioid polar patterns. The cardioid polar patterns ensured a directional frequency response and sound pick-up to avoid feedback with the PA system.

10.6 Ambience-miking

Ambience microphones are used to capture the sound of the venue and the natural reverberation of sounds in that space. Later the recorded natural reverberation of the space can be mixed together in a DAW with microphones that were placed closer to the sound source. (Huber & Runstein 2017, 135; Bartlett & Bartlett 2007, 78.)

I did not use specific ambience microphones in the recording because I figured that I don’t want to capture the amplified sound produced by the public address system. Also, placing microphones amongst the audience would have been hazardous. Instead of deploying ambience microphones in the venues, I had one audience microphone to capture some of the venue sound but mainly focus on recording the audience reactions.

10.7 Audience microphones

Recording a live band, audience reaction mics are essential to capture applauses and general feeling of the live concert (Bartlett & Bartlett 2007, 66). Placing audience microphones with moving and dancing audience can be difficult, but this can be averted with setting up the audience microphones on the stage facing the audience, at the front of house or hanging the microphones from the ceiling if possible. Later the recorded reactions can be edited and mixed at endings of songs to have a live performance feeling in the recorded songs. (Bartlett & Bartlett 2007, 66-67.)

When recording Lemmenlautta I had one small diaphragm condenser microphone in the front of house position to capture the audience reactions. The small diaphragm condenser microphone was equipped with an omnidirectional polar pattern to pick up audience reactions around the front of house position. I could place the small condenser microphone at the front of house without using additional microphone stand. I laid the microphone on top of the mixing console’s case on its microphone pouch to avert resonation rattling noises. Having the microphone on top of the case without having it on a microphone stand eliminated the possibility of an audience member knocking the microphone down while they were dancing.

10.8 Channel list, Signal routing and Stage plot

Planning the recordings ahead and preparing the gear for recording is crucial to have a successful recording session. Having a list of channels and a stage plot helps to plan a live recording session. Sometimes when arriving to a venue there can be a tight schedule for the band to set up their show. When there is a tight schedule and little time to set up, good planning and preparation beforehand helps tremendously in having successful recordings despite the rush.

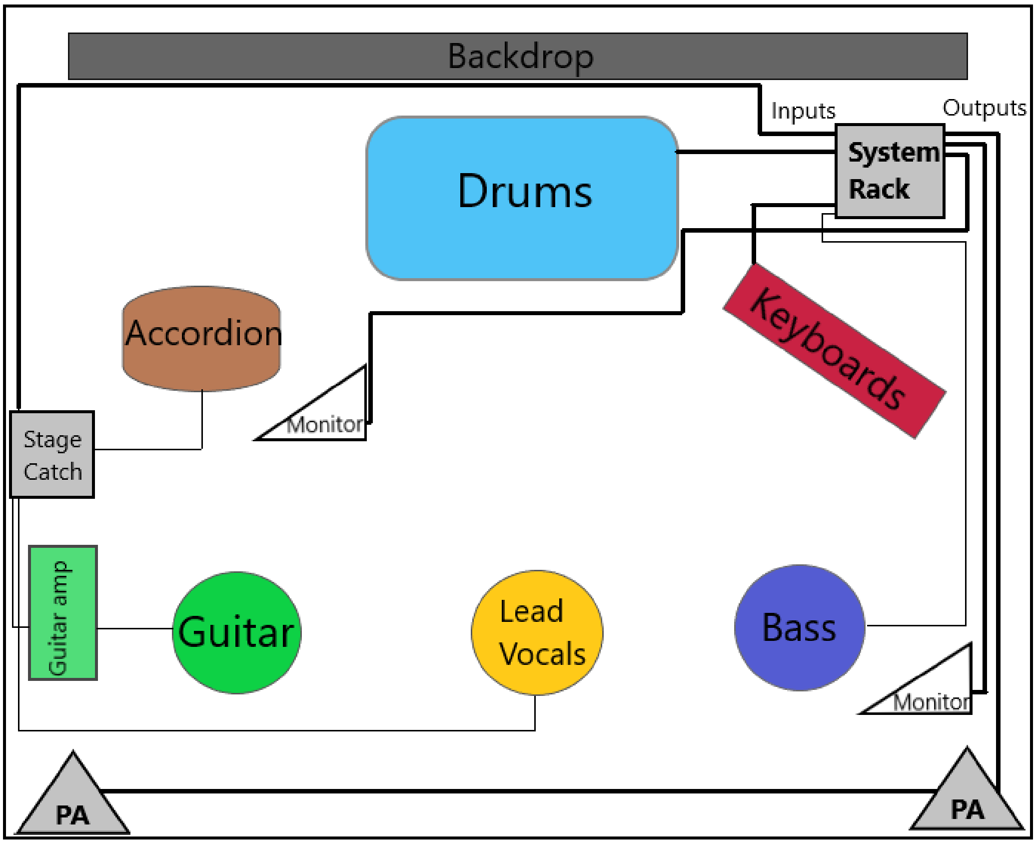


FIGURE 6. Stage plot and cabling plan of Lemmenlautta (Piironen 2019)

Stage plot and cabling plan helps to visualize where all the instruments and gear are situated and how the cables should be run to have a clear stage. Sloppily laid down cables can cause a safety hazard when performers move around on the stage. Loose cables can tangle around legs and make band members to fall possibly harming themselves, equipment or others around them. Gaffers tape is the best tool to secure cables from getting in the way during the show. Gaffers tape is also preferred as it does not leave traces of glue to cables.

The figure above shows an upside view cable management on-stage. As it can be seen from the figure, all the cables on-stage are laid down and connected to the stage catch and system rack in such fashion that there is room in the middle of the stage. Especially the lead singer in Lemmenlautta tends to move a lot during the shows.

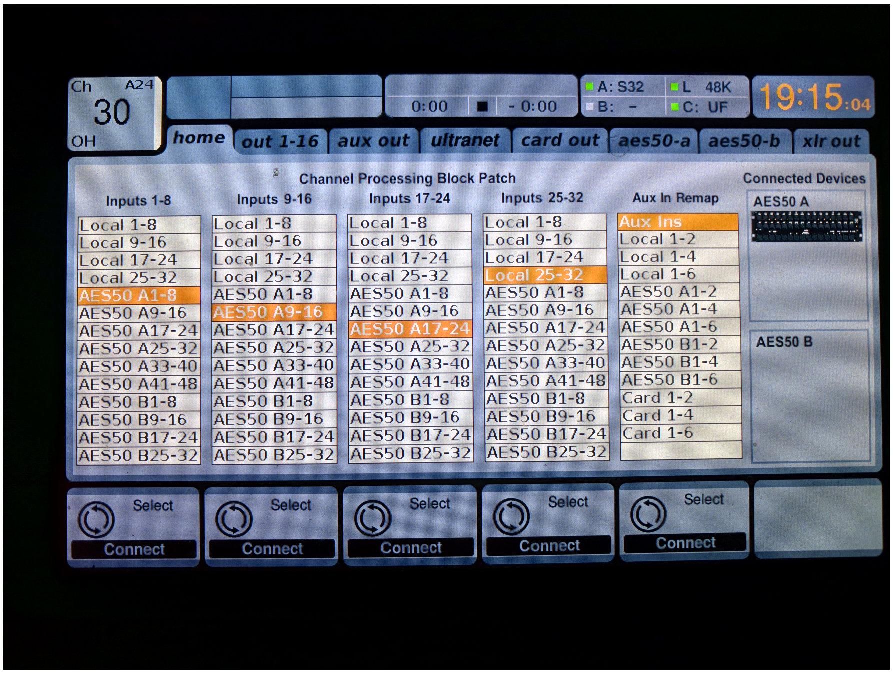
Leaving the centre of the stage clear lowers the risk of cables getting tangled with the singers or other performers feet. Good cable management also makes the striking down after the show faster. When the cables are laid down in an orderly fashion they are faster to coil and put into cases. The faster the show is struck down the faster everyone gets to go home.

CH	Instrument	Microphone	+48v	Microphone stand
1	Kick drum	Sennheiser e902		Inside kick drum
2	Snare drum	Audix i5		Clip
3	Rack tom	Sennheiser e904		Clip
4	Floor tom	Sennheiser e904		Clip
5	Hihat	Rode M5	X	Clamp mount
6	Ride	Rode M5	X	Clamp mount
7	Overhead	SE Electronics X1		Tall boom
8	Bass	DI-signal		
9	Electric Guitar	Linelevel-signal		
10	Mandolin	DI-signal		
11	Accordion	t.bone Ovid CC100	X	Clip
12	Keyboard 1 L	DI-signal		
13	Keyboard 1 R	DI-signal		
14	Keyboard 2 L	DI-signal		
15	Keyboard 2 R	DI-signal		
16	Acoustic Guitar	DI-signal		
17	Backing vocals 1	Shure sm 58b		Tall boom
18	Backing vocals 2	Shure sm 58		Tall boom
19	Backing vocals 3	Shure sm 58		Tall boom
20	Lead Vocals	Sennheiser e935		Straight boom
21	Ambience FOH	Beyerdynamic MM-1	X	

FIGURE 7. Channel list of Lemmenlautta live recordings (Piironen 2019)

The chart above shows how many input channels was needed in the recordings, what microphones were used for each sound source and which of the instruments were direct input boxes or line level signals. The channel list also includes information of the channel names, which inputs needed phantom power from the front of house console and what kind of microphone stands the selected microphones are mounted to.

The input channel list can be used when the input channels are labeled and programmed inside the live mixing console and in DAW. Each individual input is signed with its own dedicated digital signal processing channel inside the live mixing console.
Like in a professional analogue mixing console each digital signal processing channel (DSP) is equipped with gain, gate, compression and equalization settings. The live mixing console settings also include labeling and routing options for each digital signal processing channel.



PICTURE 13. Digital signal routing settings in Behringer X32 (Piironen 2019)

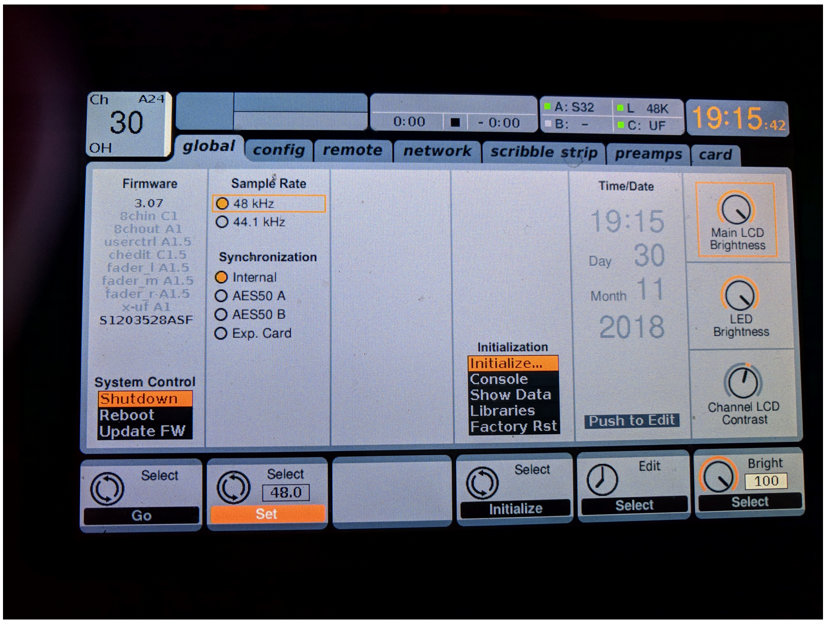
When the category 5 digital snake between the Behringer X32 live mixing console and its dedicated Behringer S32 digital stage box was connected the devices recognized each other and the mixing console indicated with a green light that the connected digital stage box device is online and operable. Within the console routing settings, the device gives the possibility for a sound engineer to choose what inputs are used in blocks of 8 channels.

In the picture above, local inputs in the routing settings mean the inputs within the console situated in the backside of the Behringer X32 mixing console. The AES50 inputs are the inputs situated in the digital stage box. The AES50 is the digital audio protocol used to transfer information from the stage box to the mixer. For example, all the instrument and microphones on-stage were connected to the stage box, but the front of house audience reaction microphone was connected to the local input 32.

The local input needed to be activated in the signal routing settings by selecting the block of local inputs starting from input channel 25 and ending to input channel 32. Selecting the local input channels 25-32 deactivated the matching inputs from the digital stage box. Those channels could not be used to receive digital audio signals from the stage until reactivated from the routing settings. The local inputs could be activated because the amount of input channels connected to the stage box would not exceed 24 input channels. When recording audience reactions through the live mixing console it is important to remember to unassign the audience reaction microphone signal from PA system to avoid feedback issues.

10.9 Settings in Logic Pro X and the live mixing console

When the digital live audio system is up and running, the computer needs to be connected to the front of house live mixing console. The recording project and the DAW needs to be working in the same sampling rate and bit depth as the live console. What sampling rate and bit depth the live audio system is operating at can be determined in the live console’s settings. Depending of the manufacturer and the console the variety of selectable options in the audio processing settings vary.



PICTURE 14. Global settings view in Behringer X32 (Piironen 2018)

The master device for the synchronization is also selected in the mixing console settings. In X32 the options are internal, AES50 or expansion card. Selecting the synchronization setting as internal the mixing console to provides the wordclock synchronization signal for the stage box and the DAW. This ensures that all the parts of the digital signal chain work in same virtual time.
The live mixing console in question had the possibility to select the sample rates between 48 and 44.1 kHz. Bit rate of the system was a factory fixed 24 bits and it could not be changed. As previously discussed, having a larger bit depth makes the sampling process less prone to hard clip the sampled digital audio.

When the computer is connected to the mixing console the digital audio workstation recognizes it. The input source settings in the DAW must be set to configure the mixing console as the input device. After the console has been configured as the input device the physical inputs can be routed within the DAW to match their corresponding tracks. When the input routing in the DAW is done, the multitrack recording is set for action.

Before the recording during the sound check I checked that all the signals reach the DAW. I could also monitor the signals through the headphone port in the laptop to hear what the signals sounded like. If there were some problems in the signals or the sound picked up by the microphone did not sound good I could go and change the XLR cable or move the microphone to a better location.

10.10 Recording the show

At the beginning of the show I armed all the tracks in the DAW and started recording couple of minutes before the show started. Starting the recording couple of minutes before the show I could just leave the recording on and focus on live mixing. During the show, I regularly checked the DAW that the recording was alright and I stopped the recording between sets to be able to playback the recorded material.

Listening to the recorded set I could hear if there were any problems during the recording that I could improve before the beginning of next set. Then after the break between the sets I started the recording again couple of minutes before the first chord. I kept repeating the same procedure on every Lemmenlautta show that I recorded and got good results with it.

10.11 Post production

When I had all the shows recorded on my computer I could start going through all the performances and takes from different venues. I had had a conversation earlier with the band members about what songs they would have wanted to be recorded and post produced. They gave me a list of twelve songs that they figured that would be good to record. I knew that after the recordings that the vast amount of songs would take time to listen through. I needed to come up with a way to collect data of all the songs to be able to compare them. In addition to that I wanted to mix the songs in Pro Tools, so I would need to export all the tracks from Logic Pro as Wave files to be able to import them into Pro Tools.

10.11.1 Choosing and exporting the best performances

The total amount of recorded shows was eleven three to four hour shows and the compiled amount of data was 308 gigabytes. To make the song vetting process easier for myself, I formed a spreadsheet which I filled in while listening through the recordings. The spreadsheet had information of the venue, date of recording, song, starting time, ending time, tempo consistency, overall performance, notes and a numerical grade from zero to five. This way of collecting data of all the recorded shows helped me to choose the best takes from the vast amount of takes. After logging down all the details of different songs I could go through the best takes of songs and choose which ones to use as final versions.

When I had my mind set on the recorded takes that I would mix later I exported each song from Logic Pro to individual folders on my external hard drive. The exported files needed to be on an external hard drive because I would mix the songs in Pro Tools. Every individual song was exported to a folder named after the venue and the name of song. Each audio file inside the folder was named with a number before the name of the instrument representing the input channel number. This helped to keep everything organized when importing the songs into Pro Tools later.

10.11.2 Editing and mixing

When I had all the songs exported and ready for mixing I set up a Pro Tools session for editing and mixing the tracks. I created a session in the digital audio workstation running on 44,1 kHz sample rate and the bit depth of 16 bits. When importing the audio files into Pro Tools the program automatically organized the tracks in chronological order following the numbering in the audio file names. I imported all the songs into Pro Tools and aligned them one after another in one session. This way I enabled the same processing to be used for all the channels and with automations I could control everything in the same session.

After importing all the tracks into Pro Tools, I started with deleting tracks which were recorded but did not contain any audio information. This was the issue with for example some backing vocal tracks. Then I moved on to editing the close-miked drums such as tom-toms, snare and kick. I removed all the parts from the recorded drum tracks which did not contain any audio information. This helped in overall leakage in the drum signals. I also added noise gate plug-ins to the close-miked drums which enabled a better control over for example how long the toms would ring.

Mixing wise I wanted to maintain the overall sound of the band as it would sound on a live gig. Rather than harshly modifying and changing the overall tone of every instrument and pitch tuning the vocals to absolute perfection I wanted to have a reproduction of natural sounding live performance with good frequency response and stereo imaging. The band plays well together, and they have their own way of arranging the songs they perform. The natural balance between all the instruments in the band played the biggest role in the mixing.

I started the mixdown with setting the fader levels between the instruments so that the overall balance was good, and everything was audible. Then I panned the channels to represent the places where the instruments were situated at on the stage. After setting the balance between instruments and their position in the stereo field I did equalization either before or after other signal processing.

I used Steven Slate’s analogue pre-amp audio circuitry emulators when mixing the songs. Every channel in the project was equipped with Slate Virtual Mix Rack plug-in running the VCC pre-amp emulator set on either a Neve pre-amp or RC-tube modeler. I found out that the individual channels equipped with the saturation plug-ins made the overall frequency response of the stereo mix smoother when played together.

I used little to none of compression plug-ins in the instrument channels. Some instrument channels like kick and snare had in the Virtual Mix Rack an analogue compressor plug-in with slow attack time and fast release to bring up more of the tail of both signals. I did not want to overly compress the instrument channels to prevent killing the sense of dynamics in the recordings. The hardest compression I used in the vocal channels. As the singer moved while singing and he had a hand-held microphone, there were changes volume wise in the vocal signal which resulted in not hearing the vocals all the time. I reduced the dynamic changes in the vocal channel to have easier control over the vocal sound.

When I had all the signal processing in place and I was happy with how the band sounded I was ready to write automations for every song. Automations in the project mainly focused on changing the level balances between instruments for example where solos occurred. There was quite a lot of automation needed for changing volume level in the vocal track. A mixing control surface connected to Pro Tools made the automating faster as I was able to write, sort of live mix, the automations with faders as I listened to the songs.

After I was happy with the overall sound and the balances between instruments I bounced each song as individual stereo mixes as Wave files. When I had bounced the songs, I could master the songs in another Pro Tools session. The mastering included checking for the frequency response with an analyzer plug-in, raising the overall level loudness with an enhancing plug-in and saturating the stereo mixes with a tape machine plug-in. Final part of the mastering process was to bounce the tracks again to print all the processing done in the Pro Tools session to the final Wave files.

11 DISCUSSION

There are multiple factors that sound engineer needs to be aware of when working with live audio and simultaneously live recording a band. One needs to have practical knowledge of how to build and set up a live show within a timeframe that sometimes can be demanding. Especially when something goes wrong the ability to quickly solve the problem and reach a practical solution is imperative for the show to go on. To be able to accurately troubleshoot problems with live audio equipment the sound engineer needs to have knowledge of all the analogue and digital devices working in the system.

Having the signal processing and recording in digital domain has not exactly made life easier for sound engineers. Digitalization has brought new problems for sound engineers to troubleshoot. Knowledge of all the processes and sophisticated devices is nowadays required. However, multitrack recording a live band as a one-man operation would not be possible without modern digital live audio systems.

During the process of this thesis I learned that having all the settings, parameters and recording media centralized to the digital live mixing console and the digital audio workstation made it possible for me to simultaneously live mix and multitrack record a live band. If I had done a similar live recording project with only analogue gear, I would have needed to have a second mixing console and analogue multitrack recording devices to run the recording and another mixer for live mixing the band. That would have been physically and practically impossible to do for one person.

I can see that this thesis can work as a tool for an aspiring sound engineer trying to wrap their head around all the subjects involved with digital audio gear and live recording. Although all the fine details of analogue and digital audio are not included in this thesis I believe that my work can give insight for a future live recording project or for someone who is interested in the subject.

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