Adam Rosiński

Microphone Techniques

in Stereo and Surround Recording

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Contents

Introduction

Sound engineering is one of the fastest-growing branches of music production. Regular scientific conferences, multiple specialist journals, and publications in many languages all point to the importance and the topicality of the issue. In the mind of a sound engineer, the concept of any audio production concerns:

- the careful selection and arrangement of microphones between musicians,
- control over room acoustics in the recording space,
- the human factor which defies all projections, measurements, or calculations,
- the unique sound of every voice and instrument,
- the unique character of the musical performance,
- the representation of the exceptional value residing in the production by noting its "spiritual" element, present in the performance and the recording, which will determine its status as a work of art and not a mere registration of sounds generated by people or musical instruments.

All of these elements combine to establish the original character of the work completed by the sound engineer, who approaches every recording individually, case-by-case. The creation of a recording is thus the act of making a new, distinct **work of art**, recorded in the digital or the analog form. In this view, the perception of sound engineering as a phonogram-making process – involving a range of fixed procedures (miking techniques and suitable devices) which guarantee success in any situation – falters, as it would lead to the unification of completed audio recordings. The need for a broad-based discussion on the issues constituting the **art** of sound engineering persists and loses none of its relevance, revealing that sound engineering should not be investigated only in the mathematical and physical context (musical acoustics) or the engineering aspect (signal processing and modification).

Publications on sound engineering represent two main directions. The first group comprises texts on issues such as the physical and mathematical characterization of acoustic fields in rooms, noise control, acoustic measurements, the speed of sound propagating in various media, or acoustic parameters of various devices. In this framework, it is regarded as an area at the juncture of mathematics and physics, deprived of the creative musical element, and focused on mathematical and physical problems relating to the analysis of signals in a given medium. The issues of the recording process and music itself are of secondary importance and often remain unexplored and unexamined. These publications are aimed at readers with a mathematical or a technical background rather than members of the art world; their readership includes university or technical university graduates with a degree in electronics or physics. Literature of this type is of little use for readers with a musical background (musical conservatories or academies, art universities), especially if they do not aspire to analyse mathematical formulas and complete specialist physical calculations, but rather seek knowledge and inspiration for solutions regarding the relationship between stereo and surround sound. Solutions that may be correct in terms of acoustics but forgo complex calculations form the central axis of musical acoustics.

The other group of publications concerns sound engineering in the strict sense, which also has a very loose relationship to music as an art discipline. In this case, the focus falls on the technical matters related to signal recording, to the detriment of music understood in this context as the processing of a single sound or a larger musical structure described in a non-musical manner, for instance with a computer program. However, it should be noted that music is not a single sound but emotions, a spiritual experience felt through the sonority of the entire musical phrases, overlapping sounds originating a harmony that accounts for the outstanding auditory value of the musical track, etc. The sound in this framework, if not discussed in the mathematical and physical context, is comprehended as a sequence of characters in the form of a zero-one code used for saving, sending, transmitting, modifying, or reading data in the language of electronic devices, much different from the human language. Sound engineers design innovative electronic devices, increasingly advanced computer programs and a variety of algorithms and they are involved with issues such as sound quality, timbre and space. Their field of expertise encompasses a far broader scope of working with sound than that outlined here, which constitutes the foundation of sound director's effort. However, a musician recording his or her material does not need to know the exact construction of an equalizer in the mixing console, the impedance of the attenuator resistors, or the software used by the console (if digital). What he or she does need to focus on are issues such as timbre, functionality, user-friendliness and intelligibility of the software. Thus, technical sound engineers use a fundamentally different body of literature, which concerns the same topic viewed from the perspective of a professional engineer rather than a musician.

Publications targeted primarily at musicians are few and far between, which is why the mutual understanding for different priorities which effectively concern the same issues faced by the engineer, the acoustician and the musician, seems to be a complex problem and the main concept explored in this publication. The author analyses the production of musical recordings from the musician's perspective, without deprecating the relevance of the physical, mathematical, and engineering considerations which are a prerequisite for the sound director's work. However, the physical and mathematical characterization, as well as the presented engineering aspects, are limited to the minimum to provide a comprehensible content for an artist musician. If the readers wish to expand their knowledge on topics related to mathematics, physics, or engineering, they will consult other publications which abound on the publishing market.

Thus, this book is intended for musicians or sound directors, but also acousticians and sound engineers wishing to learn how the musicians think. The monograph is also addressed to musicians who intend to record their material in the studio in the near future, but do not possess knowledge on studio construction, studio workflow or the art of recording. It seems important to familiarize the musicians with the reality that awaits them on the other side of the glass, thus fostering their responsibility for the work jointly produced by them – entering the studio – and the sound director. The musicians must understand the recommendations of the sound director, which may slightly deviate from the goals of the soloist or the band. These differences arise as the main idea when working in the studio is to capture music in a manner that is natural and artistic for both the performer and the listener.

This book may also be of use for students of faculties involved with sound engineering, recording, and processing in computer games, virtual reality and television.

The monograph consists of six chapters.

The first chapter discusses the elementary characteristics relating to the engineer's work with the microphones. The contents include the construction, parameters, types and directionality characteristics of the microphones used in studios, music and theatre stages and movie productions.

The second chapter covers the performance of various loudspeakers, describing a range of speaker types and parameters, as well as the layout, calibration, and parameters of playback monitors in the context of elementary knowledge on acoustic adaptation and recording studio equipment.

The third chapter presents speaker technologies for surround sound and multichannel microphones. The information covers both the solutions used for many years and the state-of-the-art technology for surround sound playback and recording with separate microphone devices which, consolidated with a few smaller microphone capsules, exhibit a complex internal construction.

The fourth chapter is intended for beginners to sound engineering and provides basic information on connectors and connections used in various sound systems. This elementary knowledge allows the sound engineer to navigate the labyrinth of cables, sockets and plugs and to comprehend the diversity of data transfer formats and protocols in different devices. The chapter provides answers for many questions regarding electroacoustic connections between analog and digital devices.

The fifth chapter focuses on stereo techniques, while providing explanations on monophonic recordings as well. In addition, it contains a description of recording the human voice and a selection of acoustic and electrical instruments such as the piano, the violin, the alto, various guitars (acoustic, electrical), wind instruments, drums, etc.

The sixth chapter presents an array of techniques for surround sound productions, from the simplest ones to the most complex, which have yet to be fully verified and tested in terms of panning consistency (crosstalk, delays) and other aspects. Apart from outlining many new multichannel miking techniques, often lacking a sufficient description in source literature, the chapter elaborates on setups using a baffle or a binaural dummy head. Finally, it describes two- and three-layer techniques for surround sound, currently developed for 4K and 8K cinema and VR software.

The text is illustrated with visual material for reference and educational purposes – to provide a schematic representation and explanation of some acoustic and musical phenomena that prove hard to describe. The figures have been prepared by Joanna Kaczmarczyk, Marcin Piotrowicz, Andrzej Wojnach and Mateusz Ustyjańczuk. I would also like to thank the employees of the music stores “Elka” and “Riff” from Olsztyn for providing music equipment for the photos that have been included in this book.

It is worth emphasizing that the topic discussed herein is so vast, multilateral, and yet unexplored in full that it defies an exhaustive rendition. The current state of the art remains incomplete and limited due to the continuous changes and the emergence of new needs in the area of surround sound engineering.

Completing an education in sound engineering is challenging and requires many skills aside from narrow, specialized knowledge. We need to realize that the training of a modern sound director must go far beyond the framework of a classical musical background. The curriculum should venture into the areas of specialist information technology for sound engineering (the use of computer programs), digital and analog electronics (expertise in signal processing and the use of devices in the recording) and musical acoustics (the ability to record a range of instruments and voices in varied, sometimes unsatisfactory acoustic conditions). As sound directors require an array of special abilities, they need to continually broaden their knowledge and expand their skillsets, which will considerably improve their versatility and achievements manifested as high-quality audio productions.

Chapter 1. Microphone parameters, types, and characteristics

This subchapter presents the most popular microphones used in professional and amateur recordings. Note that the list is not exhaustive as it should also include carbon (button), piezoelectric, electret, and laser devices[1]. They are left out from this discussion because of their withdrawal from the market, replacement with new technologies, or lack of application in musical recordings due to their structure, characteristics, or specialized design for use in the military or elsewhere.

The selection of the right microphone is mainly dependent on its parameters, which have a significant impact on the final result and should fit the type of recording.

1.1. Microphone parameters

1.1.1. *Frequency response characteristics*

A perfect microphone should exhibit the so-called flat frequency response characteristics, which implies uniformly converting different acoustic waves. Unfortunately, due to technical limitations, such a device proves virtually impossible to construct. The parameter is

[1] Żyszkowski Z. (1984). *Podstawy elektroakustyki*, 3rd edition, Wydawnictwa Naukowo-Techniczne, Warszawa, p. 415.

of critical importance because any frequency response variations during sound processing cause major discrepancies between the input and the output. To ensure high fidelity of the recorded material, it is recommended to use suitable microphones, i.e. those that exhibit flat frequency response primarily over the frequency range (bandwidth) of the recorded instrument[2] (fig. 1 a) – d)).

1.1.2. *Sensitivity*

Sensitivity determines the output voltage level produced by the sound pressure of 1 Pa (pascal) at the frequency of 1 kHz (kilohertz). This parameter also indicates the amplification required to produce a signal of sufficient volume. This information is extremely important in recording or reinforcing faint sound sources because excessive amplification of the input signal will add artifats to the recording[3]. Try to remember that:

a) dynamic microphones have a sensitivity of 1–3 mV / Pa,
b) condenser microphones have a sensitivity of 5–50 mV / Pa.

1.1.3. *Maximum sound pressure level (SPL)*

The parameter determines the maximum volume of sound that the device can capture without causing perceptible wave distortions. Max. SPL is usually provided in decibels (dB) or pascals (Pa)[4]. Some microphones manufactured nowadays can handle volumes of up to 150 dB[5].

[2] Sztekmiler K. (2003). *Podstawy nagłośnienia i realizacji nagrań. Podręcznik dla akustyków*, 2nd edition (enlarged), Narodowe Centrum Kultury, Warszawa, pp. 32–34.
[3] Żyszkowski Z. (1984). op. cit., pp. 415–416.
[4] Pohlmann K. C. (2010). *Principles of digital audio*, 6th edition, McGraw-Hill/TAB Electronics, New York, pp. 3–4.
[5] Sztekmiler K. (2003). op. cit., p. 37.

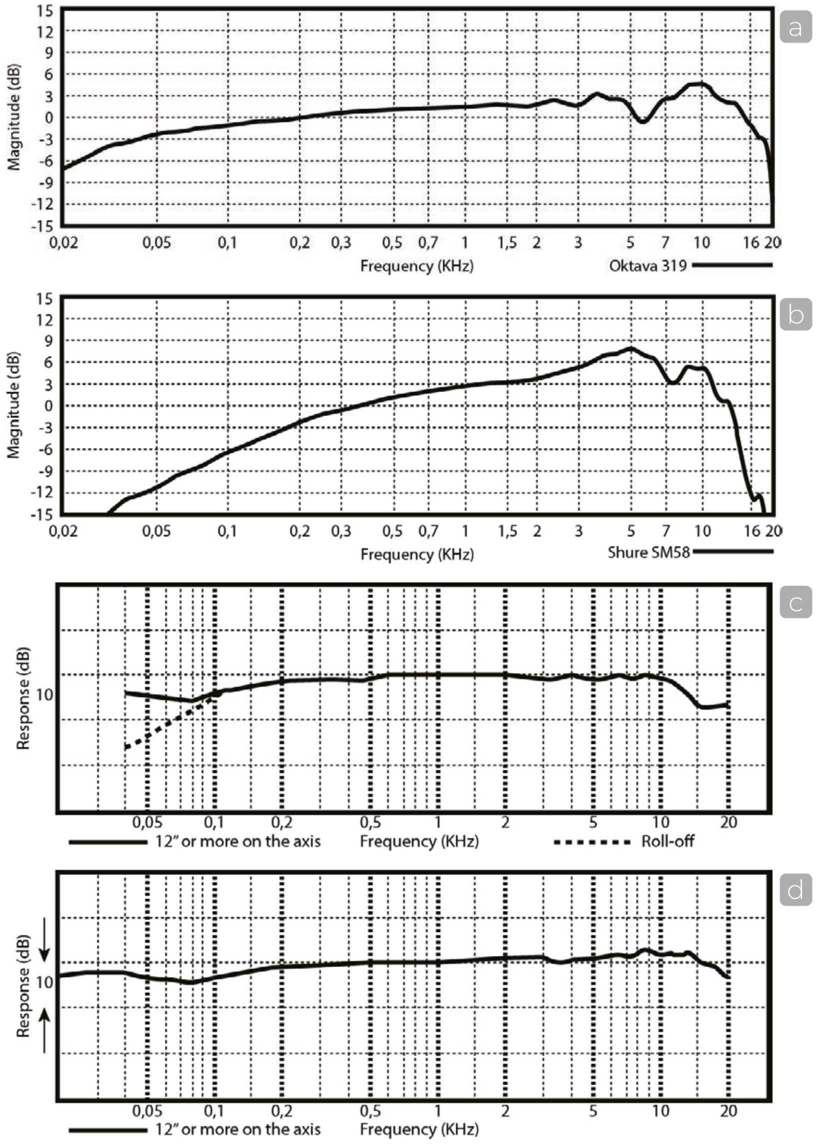

Fig. 1. Device-dependent differences in the processing of various frequencies by: a) a large-diaphragm condenser microphone, b) a dynamic microphone, c) a small-shield condenser microphone mounted on a clip, d) a large-diaphragm condenser microphone. The devices process the same bandwidths in a significantly different manner. Some receivers are suitable for capturing the drums, others – for individual voices, such as the treble, the alto, the tenor, the bass, yet another group – for wooden and tin wind instruments, and so on.

Table 1. Example values of acoustic pressure PA and the SPL, as compared to the basic thresholds of human hearing

Sound pressure [Pa]	SPL [dB_{SPL}]	Reference point
2×10^{-5}	0	Threshold of hearing
1.1×10^{-4}	15	Background in a recording studio
6.3×10^{-3}	50	Conversation
6.3×10^{-2}	70	Vacuum cleaner
6.3×10^{-1}	90	Heavy transport
2	100	Pneumatic hammer, disco
6.3	110	Rock concert
20	120	Threshold of pain
200	140	Jumbo Jet take-off, 50m away

1.1.4. *Handling noise*

Every powered electronic device generates disturbances referred to as handling noise. The quieter the noise, the better the device. The disturbances are caused by various factors, such as the subtle movement of the voice coil (which results from temperature fluctuations) or the low-level hum of the air in the internal wiring[6].

1.1.5. *Dynamic range*

The parameter indicates the highest and lowest intensity of a sound that may be captured by the microphone without distortion[7].

1.2. Microphone types

The microphones discussed in this chapter may be divided by one more criterion – the size of their diaphragm. **Small-diaphragm**

6 Ibidem.
7 Ibidem.

microphones (fig. 9 a) – d)) and **large-diaphragm microphones** (fig. 8 a) – d)) differ only in the size of their membrane, which has a bearing on the sensitivity and conversion quality of different frequencies (depending on the model).

1.2.1. *The dynamic (magnetoelectric) microphone*

This is one of the most popular microphones on all sorts of stages and in amateur studios (fig. 3 a) – c), 4 a) – d)). A dynamic microphone contains a magnet, a movable coil, and a diaphragm[8] (fig. 2). Sound waves cause vibrations of the diaphragm which – in combination with the coil – behaves like an electromagnet and induces electrical current which can be recorded by a designated device. Note that since a dynamic microphone operates like a reverse loudspeaker, it is the corresponding, reverse connection in the audio signal path that elicits the sound[9].

Dynamic microphones ensure fair sound and quality combined with a certain firmness, as they rarely break down even when dropped from a height. In comparison with condenser microphones, they do exhibit inferior sensitivity. However, this limitation can be a blessing in disguise since it makes the devices perfect for use on a stage, where the air is pervaded with both musical and non-musical sounds[10].

Dynamic microphones exhibit optimal performance in the capture and reproduction of nearby sound sources. In terms of parameters, they are generally inferior to condenser microphones but may be used in amateur recording studios, where the room is not soundproofed and the goal is to ensure the best reproduction of the sound produced at the closest proximity to the microphone[11].

8 Rossing T. D., Fletcher N. H. (2004). *Principles of Vibration and Sound*, 2nd edition, Springer-Verlag, New York, p. 240; Zettl H. (2018). *Video Basics*, 8th edition, Wadsworth Cengage Learning, Boston, p. 137.

9 Eargle J., Foreman C. (2002). *Audio engineering for sound reinforcement*, Hal Leonard, Milwaukee, pp. 55–56; Nulph R. G. (2013). “Sound Track: Microphone Types”, in: Burkhart J., *The Videomaker Guide to Video Production*, Completely Revised and Updated 4th edition, Focal Press, Burlington, p. 53.

10 Whitaker J. C., Benson B. K. (2002). *Standard handbook of audio and radio engineering*, McGraw-Hill, New York, pp. 4–16, 4–17.

11 Butler T. (1994). *Połączenia. Podstawy profesjonalnej elektroakustyki i nagłaśniania*, Firmowy podręcznik Fendera, Fender Musical Instruments Corporation, Los Angeles, pp. 74–76; Connelly D. W. (2017). *Digital Radio Production*, 3rd edition, Waveland Press, Long Grove, pp. 52–54.

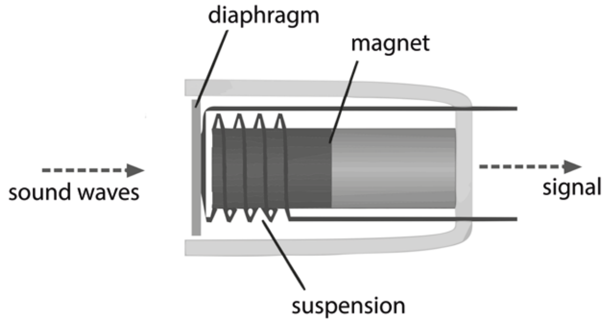

Fig. 2. The structure of a dynamic microphone. Vocalists use a range of dynamic microphone types, each equipped with a differently shaped head. The head covers the diaphragm, which converts the acoustic signal to the electric signal.

Fig. 3. a) – c) A selection of dynamic microphones used onstage by the vocalists. Each device comes equipped with a uniquely shaped head which covers the diaphragm converting the acoustic signal to the electric signal.

Fig. 4. a) – d) As the discussed microphones ensure proper parameters of the registered sound, they are typical stage and studio equipment. The microphone on the right is also recommended for sound reinforcement and the recording of various instruments and electric guitar amplifiers.

Fig. 5. a) and b) The figure presents dynamic microphones often used by the vocalists. Their structure differs from traditional dynamic microphones as the diaphragm remains in the vertical position or at an angle.

Fig. 6. a) and b) Large-diaphragm dynamic microphones are used for recording or reinforcing the bass drum or an amplifier of the bass guitar or the double bass. The characteristic use of the large diaphragm allows capturing and processing sounds of high acoustic pressure and low frequencies without damage to the microphone.

1.2.2. *Condenser (capacitor) microphone*

This device contains similar electroacoustic elements of a dynamic microphone with the notable addition of an electronic condenser which accumulates the electrical charge (like a battery) and releases it by inducing an electrical current. Another noteworthy point is the diaphragm is made of a conductive material. The names of the device refer to the condenser and its capacitance for charging and discharging[12] (fig. 7).

Condenser microphones exhibit far greater sensitivity and accuracy in the reproduction of the entire aural bandwidth than their dynamic counterparts. They are used for audio recording in semi-professional and professional studios where their precise frequency response makes them a desirable choice for capturing instruments and the human voice (fig. 8 a) – d), 9 a) – d)). Condenser microphone can be used in soundproofed spaces adapted for sound recording, as well as in rooms with qualified acoustics and

[12] Eagle D. S. (2005). *Instant Digital Audio*, CMP Books, San Francisco, p. 36.

open spaces. In extreme cases, it can be sensitive enough to convert the engine of a car passing 100 m away from the studio. Finally, bear in mind that the device is highly delicate. Any drop, even from a small height, typically causes permanent (irreparable) damage[13].

Another important matter is the power supply of condenser microphones. Phantom Power[14] (usually indicated on electronic devices as +48V, 48V, or P48)[15] characteristically uses DC voltage on the power supply sockets. Even though the power and the sound travel over the same cables, the main audio signal is not distorted or degraded by the electrical energy[16]. Common power supplies include mixing consoles, recording devices, microphone preamplifiers, etc. Most Phantom Power receivers that ensure proper performance are condenser microphones and active DI boxes. Phantom Power may also be encountered in the following variants: +24, 24V or P24 or +12, 12V, or P12. However, those occur sporadically, being the remnants of the history of electroacoustic devices, as the first condenser microphones were powered with 9– 12V[17].

Note that some microphones are powered with an internal power supply within the casing (a battery). This type of power supply should be provided in a small mixing console or a recording device with a microphone plug-on (over a regular XLR cable). Phantom power is required for activation and proper operation of a condenser microphone[18].

13 Jones D. (2012). *Music Technology A-Level. A practical guide for students using Cubase*, Darren Jones Press, Bournemouth, p. 243, 246; Chappell J. (1999). *The Recording Guitarist. A Guide for Home and Studio*, Hal Leonard, Milwaukee, p. 53.

14 Owens J., Millerson G. (2012). *Video Production Handbook*, 5th edition, Elsevier/Focal Press, Waltham, p. 237; Rudolph T. E., Leonard V. A. Jr. (2001). *Recording in the Digital World: Complete Guide to Studio Gear and Software*, Berklee Press, Boston, p. 113.

15 Jones D. (2012). op. cit., p. 252.

16 Reese D. E., Gross L. S., Gross B. (2006). *Radio Production Worktext. Studio and Equipment*, 5th edition, Focal Press, Burlington, pp. 64–65.

17 Bartlett B., Bartlett J. (2013). *Practical Recording Techniques: The Step-By-Step Approach to Professional Audio Recording*, 6th edition, Focal Press, Waltham, pp. 445–446; Davis G., Jones R. (1989). *Sound Reinforcement Hanbook*, 2nd edition, Hal Leonard Corporation, Yamaha Corporation of America, Buena Park, p. 130; Huber D. M., Runstein R. E. (2010). *Modern recording techniques*, 7th edition, Focal Press/Elsevier, Burlington, pp. 117–118.

18 Thompson D. M. (2005). *Understanding Audio: Getting the Most Out of Your Project or Professional Recording Studio*, Berklee Press, Boston, pp. 22–23.

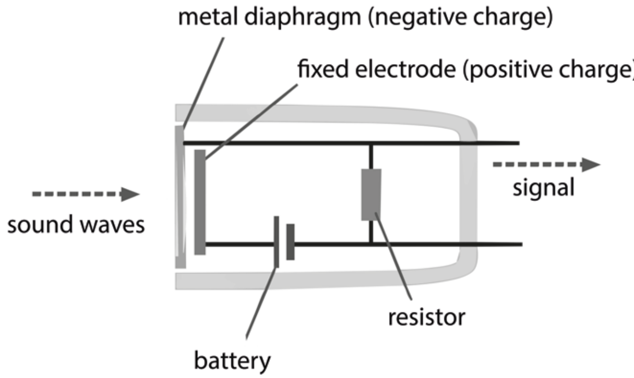

Fig. 7. The structure of a condenser microphone.

Fig. 8. a) – d) Large-diaphragm condenser microphones exhibiting multi-directionality (their sound pickup patterns can be switched by the user) are deployed in recording studios. They operate properly on phantom power, supplied from such sources as the recording device or a mixing console. The multi-pattern directionality allows for optimally positioning the microphone relative to the source of sound propagation so that the device captures the audio from the selected space fragment only while suppressing other (unwanted) sounds. Additionally, such microphones often come equipped with a signal attenuator labelled as "pad" (several to twenty dB) and can cut off selected low frequencies to eliminate thumping during the recording process.

Fig. 9. a) – d) Recording studios use a variety of small-membrane condenser microphones with a single pattern (that cannot be changed by the user), all operating properly on phantom power. Additionally, microphones of this type often come equipped with a signal attenuator labelled as "pad" (several to twenty dB) and can cut off selected low frequencies to eliminate thumping during the recording process.

1.2.3. *Ribbon microphone*

A dynamic microphone stands out with its structure. Up till now, we have discussed coil microphones (with a coil moving in the magnetic field). In contrast, ribbon microphones have only one type of wire in the form of a thin, aluminium ribbon suspended between electromagnetic poles, which is the key element of the entire construction[19] (fig. 10). When the ribbon in the magnetic field is excited by the sound wave, it induces an electrical current carried over the cable to the recorder[20]. Ribbon microphones used to be bulky and extremely delicate (even a cough of the performer could damage the device). Modern models are small (fig. 11), handy, and high-quality products (little noise, warm and clear sound), not

19 Stark S. H. (2003). *Live Sound Reinforcement: A Comprehensive Guide to P.A. and Music Reinforcement Systes and Technology*, 8th edition, MixBooks, Vallejo, pp. 22–23.

20 Gilette M. J. (2005). *Theatrical Design and Production: An Introduction to Scene Design and Construction, Lighting, Sound, Costume and Makeup*, 5th edition, McGrawHill Press, New York, pp. 21–22.

as rugged as dynamic microphones but sturdy enough to endure most common incidents in the recording room. A ribbon microphone ensures high-quality sound over a wide hearing range, with a relatively flat frequency response. It is used for recording musical instruments, notably tin and wooden wind instruments[21].

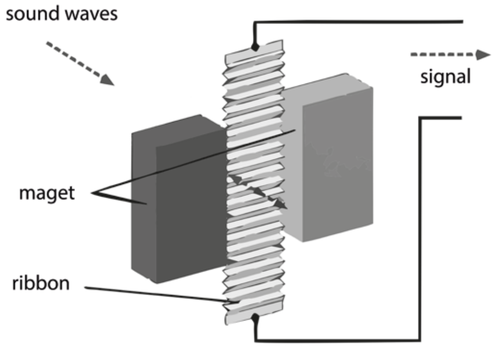

Fig. 10. The structure of a ribbon microphone.

Fig. 11. A ribbon microphone.

[21] Butler T. (1994). op. cit., pp. 73–75; Alten S. R. (2011). op. cit., p. 65; Rising R. W. (1996). "Aural Pickup Devices", in: Whitaker J. C. (ed.), *The Electronic Handbook*, CRC Press/IEEE Press – Technical Press Inc., Beaverton, p. 224; Corbett I. (2015). *Mic It! Microphones, Microphone Techniques, and Their Impact on the Final Mix*, Focal Press, New York – London, pp. 73–75; Thompson D. M. (2005). op. cit., pp. 21–22.

1.2.4. *Contact microphone*

Devices of this kind receive sound through direct contact with a sound source, as they are attached to the resonator, e.g. a violin, guitar, etc. Such a microphone receives sound waves through contact with the vibrating surface of the instrument, unlike with other instruments, when the sound is received through air vibrations[22].

Piezoelectric (including dynamic) microphones, used in acoustic (electroacoustic) guitars, are among the best-known types of such transducers. There are also varieties in the form of an elastic strap, which is a kind of condenser. A contact microphone acts by transforming vibrations of the resonator into an electric charge, which can be sent to a microphone pre-amplifier, by which the instrument sound is amplified. Such microphones are used when there is a high risk of audio feedback, and other microphones cannot be used in given acoustic conditions.

The major disadvantage of contact microphones is the narrow frequency range which they can transform with fidelity. These devices change the timbre of the sound being amplified. Due to the complex characteristics of resonators, it is very important where such a microphone is placed, as microphones attached at different places can produce sounds of different quality. The sound timbre also depends on the way the microphone is fixed to the resonator[23].

A contact microphone should not be used in a studio as it does not transform a wide frequency range with fidelity. It is recommended when a different microphone cannot be used for experimenting with the timbre and sound by sound designers, electroacoustic and electronic music composers and for creating unusual sound effects.

1.2.5. *Tube microphone*

A microphone of this type has a built-in tube which reinforces the structure but plays no part in the process of acoustic-to-electrical signal conversion (fig. 12 f)).

[22] Nisbett A. (2013). *Sound Studio: Audio Techniques for Radio, Television, Film and Recording*, 7th edition, Focal Press, Burlington, pp. 79–80.

[23] Davis G., Jones R. (1989). *Sound Reinforcement Hanbook*, 2nd edition, Hal Leonard Corporation, Yamaha Corporation of America, Buena Park, p. 118.

For proper performance, tube microphones require an external power supply[24] provided by a special transformer locked in an airtight and additionally shielded tin box (included with the device by the manufacturer) (fig. 12 c) – d), g)).

Note that the supply voltage may vary depending on the model. Though typically much higher than the phantom power voltage, it could also be lower. The decisive factors are the type of tube used and the right amperage and voltage, which ensure the smooth operation of the device[25].

Tube microphones are appreciated in recording studios for their warm and open sound. The tube allows for adding the so-called even-harmonics into the recording, which enhances the perceived quality of the sound and thus the experience of the listener[26] (fig. 12 a) – b), e)).

Specialist tube microphones are typically costly, with prices exceeding EUR 2,500[27]. There is a recent market trend to manufacture more affordable models (starting from EUR 250), however, such devices are wanting in the quality of the critical element – the tube itself. They produce unsatisfactory sound and do not compare to specialist tube microphones in terms of timbre[28].

[24] Senior M. (2015). *Recording Secrets for the Small Studio*, Focal Press, Burlington, p. 119.
[25] Ibidem, p. 121.
[26] Gibson B. (1999). *The AudioPro Home Recording Course*, Mix Books, Vallejo, p. 245; Corbett I. (2015). op. cit., pp. 75–76.
[27] Rudolph T. E, Leonard V. A. Jr. (2001). op. cit., p. 119.
[28] Hansen C. H., Snyder S. D. (1997). *Active control of noise and vibration*, E & FN Spon, an imprint of Chapman & Hall, London, pp. 1184–1185.

Fig. 12. a) – g) Tube microphones feature a tube which imbues the processed sound with the characteristic warm tonality as even harmonics are added to the audio signal. Tube microphones always come with a specialized cable for the power supply unit, which transmits both audio and the electrical current. The power supply generates electricity to power the tube correctly and allows for power modulation, which produces various directionality characteristics. The sockets are usually constructed to prevent a direct connection to the mixing console with an XLR connector since phantom power could damage the device. A tube microphone connected to the mixing console through the supply unit can operate without phantom power, as the device sends a correct sound signal which does not need an additional power supply.

1.2.6. *Shotgun microphone*

A microphone of this type is a device with a characteristically oblong structure (fig. 13), a length of up to several dozen centimetres and a diameter of approximately 1.5–2 cm. The device exhibits a strictly hypercardioid sound pickup pattern. It is used for movie and television productions. A shotgun may sit far (even a few meters) from the lips of the speaker and still capture the voice with excellent quality because of its directionality characteristics. The shotgun displays high sensitivity, which is a blessing as much as a curse, since it will process not only the actor's voice, but also the rustles of the trees, birdsong, or the hum of cars driving past (even 1 km away!). An ambulance passing in strong wind conditions may be picked up from the distance of 3 km, measured in a straight line from the device. Microphones of this type typically use a 48V power supply. Additionally, some models have the option of battery use if the recording device is not equipped with a phantom power supply[29].

Fig. 13. A shotgun microphone.

The shotgun is mounted on a special stand referred to in the moviemaker's lingo as the boompole. The boompole, carried in hand by a soundman, is fitted with a special anti-vibration pad (not unlike the basket housing a condenser microphone), and the microphone features inside. With this setup, even if the boompole accidentally bumps into something, the unwanted sound will not reach the

[29] Stark S. H. (2003). op. cit., pp. 84–85; Harrington R., Weiser M. (2011). *Professional Web Video: Plan, Produce, Distribute, Promote and Monetize Quality Video*, Focal Press, Burlington, pp. 44–45; Zettl H. (2012). *Television Production Handbook*, 11th edition, Wadsworth Cengage Learning, Boston, pp. 173–176; Grove E. (2004). *Raindance Producers' Lab Lo-To-No Budget Filmmaking*, Focal Press, Burlington, p. 84; Wolsky T. (2005). *Video Production Workshop*, CMP Books, San Francisco, pp. 139–140; Rose J. (2008). *Producing Great Sound for Film and Video*, Focal Press, Burlington, pp. 112–113, pp. 245–255; Alysen B. (2002). *The Electronic Reporter. Broadcast Journalism in Australia*, Deakin University Press, Sydney, pp. 66; Eagle D. S. (2005). op. cit., pp. 56–58.

microphone. The boompole is a telescopic device. When extended, it is several meters long, so the soundman can stay outside the camera frame without disrupting the video recording by, for instance, casting a shadow that could be captured by the camera. As an add-on, you may fit the device with a separately purchased windscreen made of foam or artificial hair resembling animal fur. The windscreen will dampen the sounds such as wind noise, which could reach the microphone during the recording.

1.3. Other kinds of microphones

1.3.1. *Stereo microphone*

Continued technological progress brings a never-ending chain of new solutions. The microphones discussed in this book are monophonic devices which can serve for producing stereophonic recordings with proper techniques. Current trends include the increased production of stereo devices which consist of two identical microphones (fig. 14). For proper operation, stereo condenser microphones require double phantom power (for each microphone separately)[30].

Fig. 14. A condenser microphone with two capsules – stereo.

30 Rumsey F., McCormick T. (2009). *Sound and Recording*, 6th edition, Focal Press, Amsterdam – London, p. 61; Herrington J. D. (2005). *Podcasting Hacks. Tips & Tools for Blogging Out Load*, O'reilly, Sebastopol, p. 85; Bartlett B., Bartlett J. (2014). *Recording Music on Location: Capturing the Live Performance*, 2nd edition, Focal Press, Burlington, p. 183.

1.3.2. *Surround microphone (ambisonics)*

With the developments in sound technologies, new recording devices equipped with multiple microphones appeared on the market thus far dominated by traditional microphones used for mono and stereo recordings. The operating principle of the new devices is based on a capsule (or a protecting net) which houses several microphones of various characteristics and directionality. All of them need to be connected to a designated matrix which distributes the captured signal among individual channels. Matrix distribution (fig. 15) allows arranging the signals in space according to the 5.1 or 7.1 standards (fig. 16).

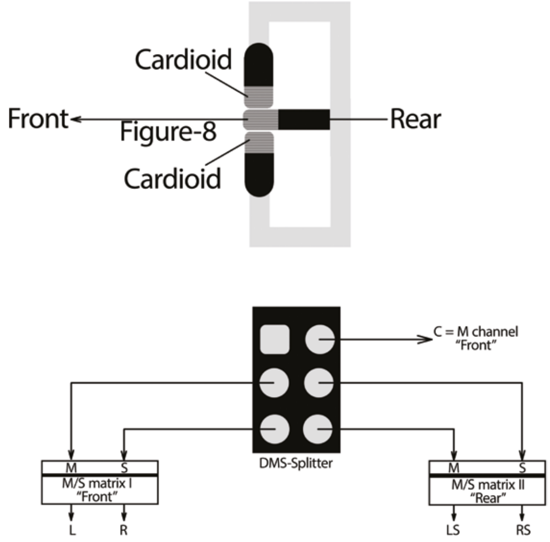

Fig. 15. A set of microphones for Double MS recordings with a matrix for signal distribution among individual channels (More on Double MS recordings in chapter 6.1.7).

In reality, similar effects to those produced with specialized multichannel (surround) microphones can be achieved by the proficient use of several monophonic microphones – their proper placement and adjustment to recording conditions – combined with a high command of multichannel recordings. In other words, surround sound in the recording may be generated without multichannel microphones. Those devices were created chiefly to streamline the work of a sound engineer.

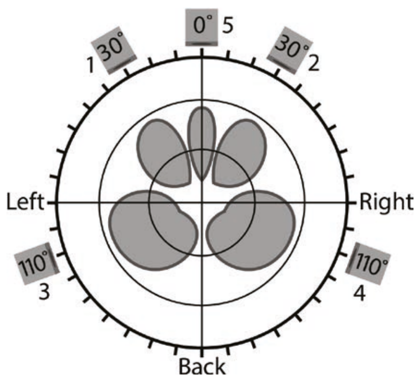

Fig. 16. Computer software for switching sound pickup patterns of the microphones installed inside the main device to generate surround sound.

Stereophony can also be transformed into a 5.1 or 7.1 recording with the use of special techniques. However, the quality and precision of spatialisation in such solutions falls short in comparison with multi-microphone set-ups. Stereophonic tracks can undergo surround treatment. With DSP and special effects, such adaptation may bring them close to a multichannel production (fig. 17–21)

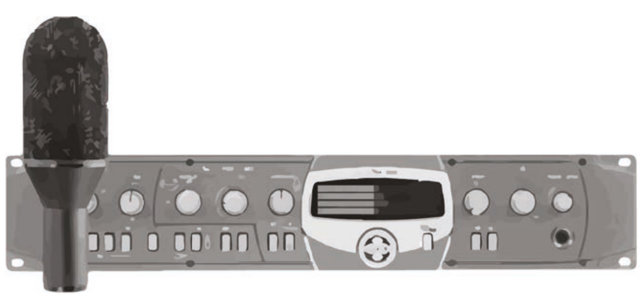

Fig. 17. A multichannel (ambisonics) microphone with the recommended device for advanced modification of various microphone parameters and the separation of surround signals.

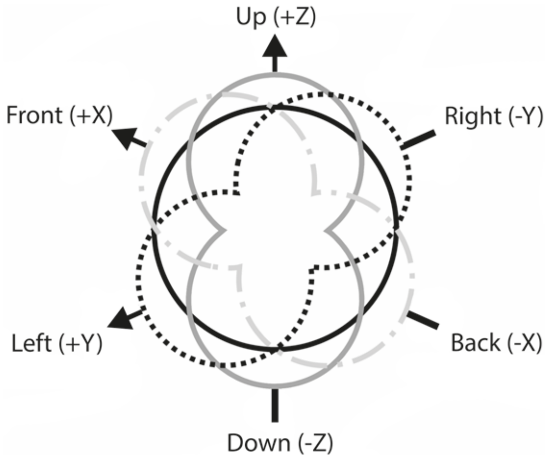

Fig. 18. Directionality characteristics for multichannel (ambisonics) microphones.

Fig. 19. A multichannel microphone (ambisonics) with the recommended device for advanced modification of various microphone parameters.

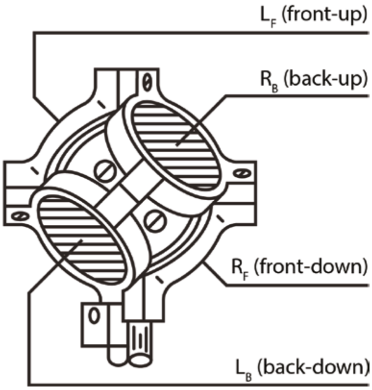

Fig. 20. The structure of a multichannel (ambisonics) microphone.

Fig. 21. A multichannel (ambisonics) microphone.

1.3.3. *Dummy head microphone*

Dummy head microphones are specialized devices comprised of two or more microphones with different directionality characteristics mounted inside a special "sphere" (fig. 22–25). The "sphere" can be made of many materials, e.g. special acoustic foam, artificial skin imitating human skin, silicone, etc. Its shape can mimic the human head, including detailed shapes of the earlobes, with microphone membranes taking the place of the eardrums. Dummy heads can take also other forms, resembling actual spheres, ovals or other geometric shapes.

Fig. 22. A multichannel microphone (dummy head) made of special material. Since the microphones are placed to mirror the location of human ears, the device can simulate binaural hearing.

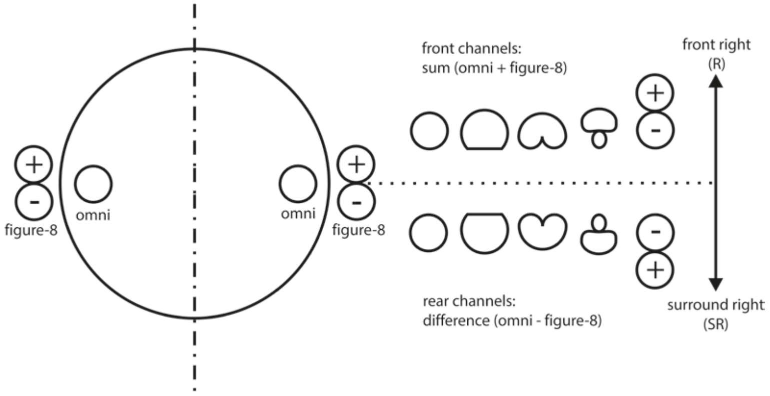

Fig. 23. Directionality characteristics of the microphones used for a multichannel construction in the form of a dummy head.

Dummy head in the fig. 24 can be used not only for surround sound but also for holophonic recordings. Such productions are often referred to as acoustic holograms, 3D or binaural recordings (since both ears are simulated). In contrast to other solutions, holophonic audio may be played on stereo headphones without setting up an entire surround sound system. In this case, headphones offer superior precision and a sense of spaciousness than a set of columns. With headphones, the listener of a holophonic recording gains double aural sensation in the left and right ear. The phenomenon is based on psychoacoustic principles of human hearing, i.e. the relations between the size and shape of the human head, the shape of the earlobes, and time-arrival differences between signals reaching both ears, including differences in the phase of the acoustic signal and volume amplitudes.

Fig. 24. Another version of a multichannel microphone (dummy head).

1.3.4. *Rosiński dummy head surround microphone*

An artificial head for surround recordings, developed by the author together with basic miking techniques allowing for registering audio in a 5.1, 7.1, or 9.1 space. The head structure is based on an oval and resembles a rugby ball. It may house 6–10 condenser microphones exhibiting omnidirectional/cardioid/supercardioid pick-up patterns. The position of individual microphones is presented in the figure 25. Additionally, the sound engineer may switch the directionality of every device separately, guided by his or her taste. Pattern choice depends on the ensemble, the recording room, and the desired quality of the spatial effect. The figure 25 schematically presents how to place the dummy head relative to the performers to properly register the accurate sound image of a band or an orchestra.

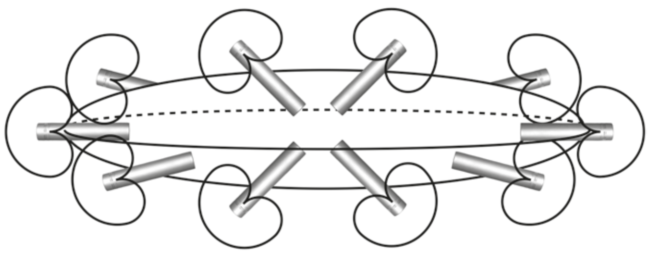

Fig. 25. A rugby-ball-shaped device with microphones mounted inside it – a dummy head.

1.3.5. *22.2 Surround microphone*

Surround microphones for 22.2 recording are highly specialized devices with many variants in their construction, as shown in figures 26 a) – d). Importantly, the device is comprised of 24 traditional microphones mounted to pick up sound from different directions. It can take many forms with the following options:

- 24 microphones distributed in a circle (fig. 26 a)),
- microphones placed in special acoustic cases in order to separate sound reaching individual devices (fig. 26 b)),
- microphones placed on stands at different heights, and pointing in different directions (fig. 26 c)),
- microphones placed as if “radiating” from the same point, covering the surrounding area (fig. 26 d)).

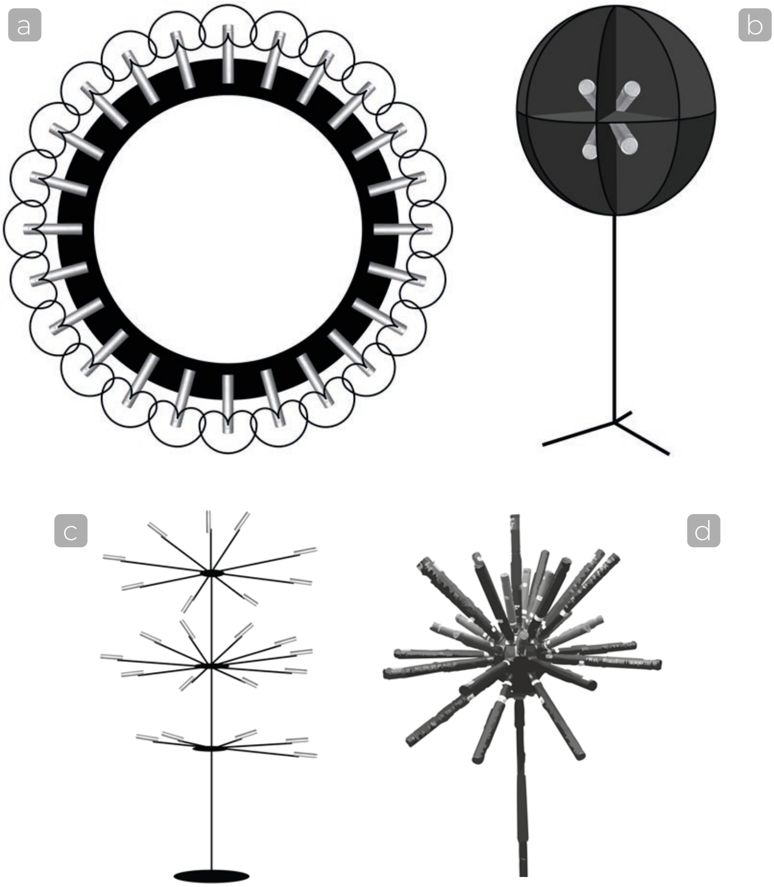

Fig. 26. a) – d) Various surround microphones for 22.2 technique.

1.3.6. *32 / 64 channel ambisonic microphone*

Two distinct miking devices which record the surrounding sound in all directions, ensuring spatial representation with the coverage angle of 360°. They are not techniques but devices which house 32 or 64 inbuilt microphone capsules mounted on a larger, circular head (fig. 27 a) – b)). Their arrangement is grounded in mathematical and physical data established in the course of broad-based experiments and research. Devices of this type are used in high-quality recordings for the 4K and 8K cinema and augmented reality.

The 64-channel microphone is still in the development phase and thus unavailable on the market as at the moment when this publication is drafted. However, completed engineering samples in the form of prototypes are now tested by a range of companies involved with recordings, research into room acoustics, and other sound-related experiments. Devices of this type are presented herein to broaden the horizons of sound engineers with regard to modern, currently developed technological solutions which will soon be ready for use.

Fig. 27. a) 32-channel surround microphone and b) 64-channel surround microphone.

1.4. Sound pickup pattern (directionality)

The microphones discussed above have yet another parameter to consider – directionality. Otherwise referred to as the pickup pattern, it determines the sensitivity of the microphone to its surrounding space, informing where to point the device for optimum sound recreation. Inaccurate placement leads to incoherent productions (where every instrument sounds as if recorded in a room with different acoustics) whose timbre leaves a lot to be desired.

The pickup pattern follows from the construction of the internal casing. A microphone contains a network of channels and openings, invisible from the outside, which suppresses sound waves coming to the head from unwanted directions and converts the right ones to electrical energy[31].

1.4.1. *Omnidirectional pattern*

Omnidirectional microphones are equally sensitive to signals arriving from every direction (fig. 28). In other words, a sound wave reaching the microphone from any point will be picked up uniformly. Omnis typically have an even frequency response, so they are used in soundproofed recording studios with no ambient noise. Contrarily, they cannot be deployed onstage because they convert sounds from all sides – including noise, beat, thumping, etc.[32]

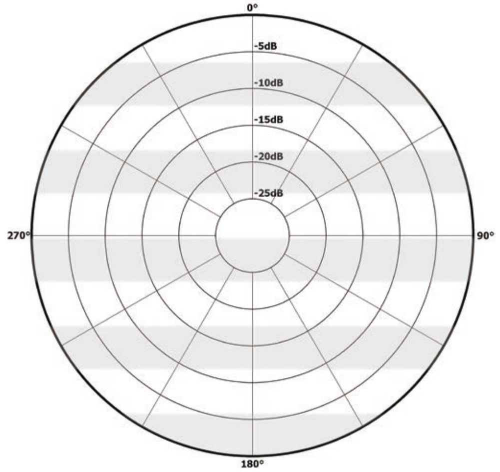

Fig. 28. Omnidirectional pickup pattern.

[31] Butler T. (1994). op. cit., pp. 75–77.

[32] Whitaker J. C. (2003). *Master Handbook of Audio Production*, McGraw-Hill, New York, pp. 120–136; Rumsey F., McCormick T. (2009). op. cit., pp. 53–55; Sauls S. J., Stark C. A. (2016). *Audio Production Worktext: Concepts, Techniques, and Equipment*, 8th edition, Routledge, New York – London, pp. 67–68.

1.4.2. *The subcardioid pickup pattern*

Subcardioid pattern is the middle ground between omnidirectional and cardioid patterns[33] (fig. 29). It has recently gained increased popularity, which is the consequence of the advancement in manufacturing technologies and the new need for innovative miking techniques and forms developed (among others) for surround sound. With modern methods of recording the sonic signal, pickup patterns that were previously unheard of or underused may find an application.

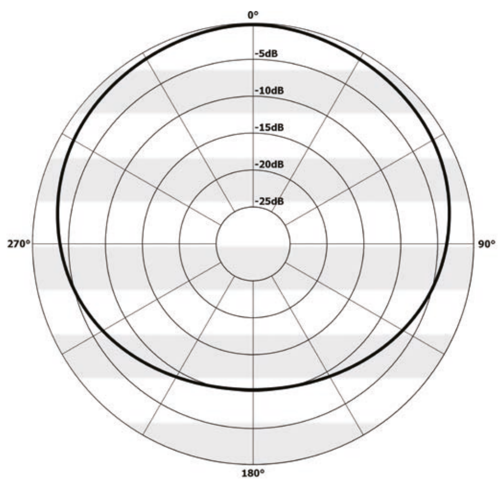

Fig. 29. The subcardioid pickup pattern.

1.4.3. *Cardioid pattern*

This heart-shaped pickup pattern, a combination of figure-8 and omnidirectional variants, is the most commonly encountered type of directionality (fig. 31). Since they can be positioned to pick up waves from a particular sound source[34] (fig. 30).

[33] Rumsey F., McCormick T. (2006). *Sound and Recording: An Introduction*, 5th edition, Focal Press, Burlington, p. 515; Gibson B. (2020). *The Ultimate Live Sound Operator' Handbook*, 3rd edition, Rowman & Littlefield Publishers, Lanham – Boulder – New York – London, p. 184.

[34] Talbot-Smith M. (2002). *Sound Engineering Explained*, Focal Press, Oxford, pp. 44–47.

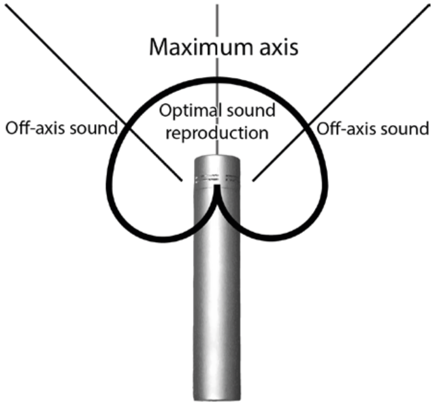

Fig. 30. Significantly inferior reproduction of off-axis sounds, which may not be picked up at all due to insufficient volume.

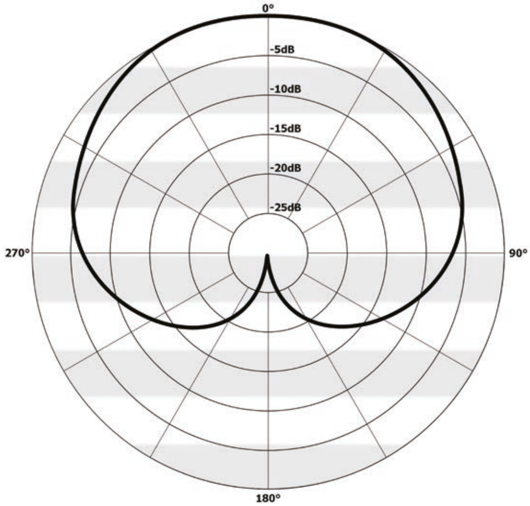

Fig. 31. Cardioid pickup pattern.

1.4.4. *Supercardioid and hypercardioid patterns*

Their singular sensitivity angles ensure high attenuation of sounds from unwanted directions. Super- and hypercardioids (fig. 32 and 33) may be used for audio reinforcement and the recording of the sources located up to several meters away. Keep in mind that the sensitivity of sound picked up from the rear will increase[35].

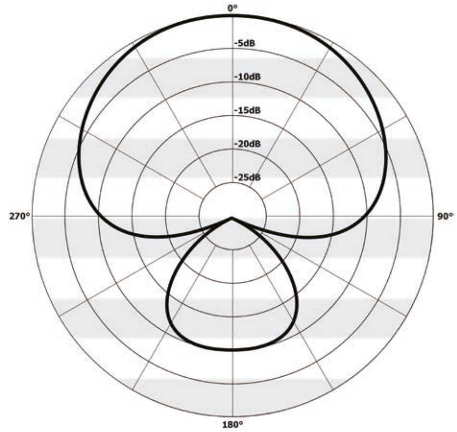

Fig. 32. Supercardioid pickup pattern.

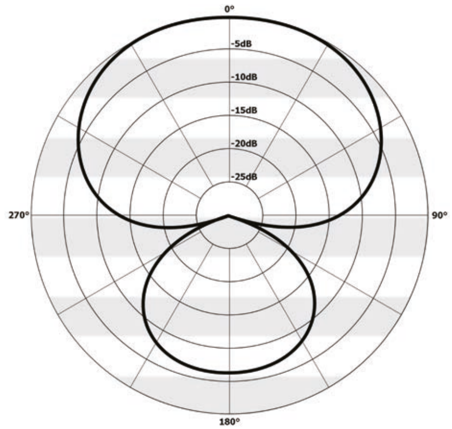

Fig. 33. Hypercardioid pickup pattern.

35 Sztekmiler K. (2003). op. cit., pp. 34–36; Lubin T. (2009). *Getting great sounds: The microphone book*, Thompson Course Technology, Boston, pp. 15–16.

1.4.5. *Bidirectional pattern*

Bidirectionality is also commonly referred to as the figure-8 pattern because of the characteristic shape of its sensitivity graph (fig. 34). A figure-8 microphone is the most sensitive to the on-axis waves, while the sounds coming directly from the top, the bottom, or the sides are less inaudible, depending on the position of the microphone relative to the sound source. Bidirectional microphones are often used during interviews and in recording studios[36] where they allow recording two vocalists facing each other. Furthermore, they are a convenient choice for recording drums since a single figure-8 device placed between the tom-toms may suffice to capture the entire set[37] (it is not a professional solution, only used in the absence of a large number of microphones).

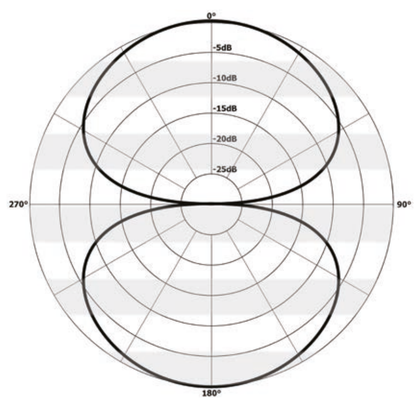

Fig. 34. Bidirectional pickup pattern (figure-8).

When discussing microphone directionality, you need to consider yet another special parameter, namely volatility. This outline (fig. 35 and 36) presents pickup patterns and their generally accepted applications in various types of recordings, microphone techniques, or rooms. However, since directionality depends mainly on sound

[36] Rumsey F., McCormick T. (2009). op. cit., pp. 55–57.

[37] Butler T. (1994). op. cit., p. 78; Owsinski B. (2005). *The Recording Engineer's Handbook*, ArtistitPro Publishing – Thompson Course Technology, Boston, p. 14; Kefauver A. P. (2001). *The audio recording handbook*, A-R Editions, Middleton, p. 66.

conversion frequency, the same device may exhibit different pick-up patterns for different frequencies. Microphone directionality is usually gauged over the range of 250–16 000 Hz, with measurements taken at breakpoint frequencies such as: 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, 8 kHz, 16 kHz.

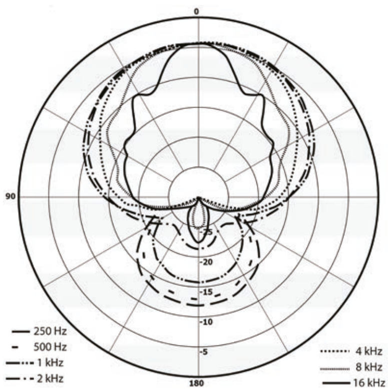

Fig. 35. Sound pickup patterns (directionality characteristics) exhibited by different microphones for sample frequencies reveal that two devices located in the same position may process the sounds of the surrounding space differently (a comparison of two different microphones).

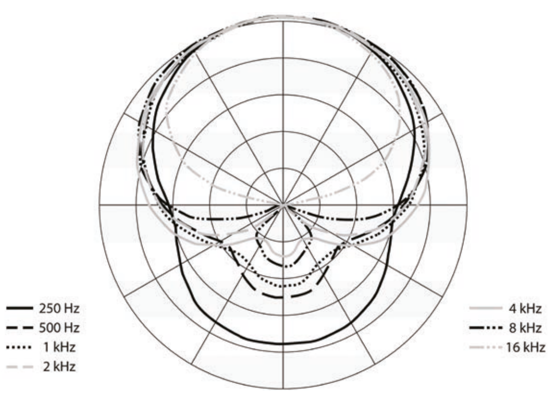

Fig. 36. Sound pickup patterns (directionality characteristics) exhibited by different microphones for sample frequencies reveal that two devices located in the same position may process the sounds of the surrounding space differently (a comparison of two different microphones).

Frequency-dependent directionality variations and the preponderance of basic patterns offer perfect grounds for experimenting. An original configuration of multiple microphones aimed to establish a new recording method may create a complex blend of overlapping pickup patterns. An unconventional arrangement of microphones – and thus their pickup patterns – results in a novel and experimental recording of the sound source. With adequate adjustments, it could improve the timbre, optimize the impact of room acoustics, or record and reflect the width of your stereo base in a different manner. Figure 37 presents a combination of patterns exhibited by four supercardioid microphones which can be used to elicit surround sound (for more information on surround sound, see: chapters 3 and 6) or explore stereophonic recordings.

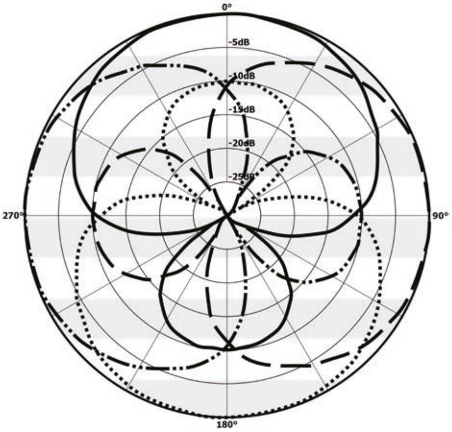

Fig. 37. Auditory space processing by four supercardioids angled at 90 degrees to each other.

For instance, a choir should be recorded by having the whole group circle one omnidirectional microphone, with each member facing the device at an even distance from the centre (fig. 38). The vocalists should stand close to each other, while their distance from the microphone depends chiefly on the size of the group.

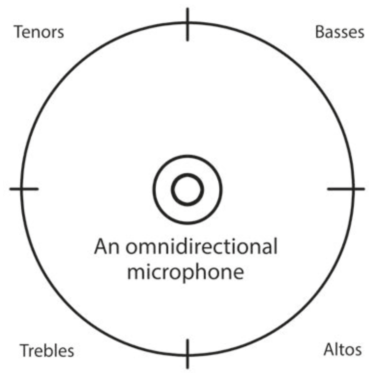

Fig. 38. Choir arrangement in a circle formation around an omnidirectional microphone. The simplest "home" solution for beginners. It is not used in professional sound production – can be used as a recording of a choir rehearsal.

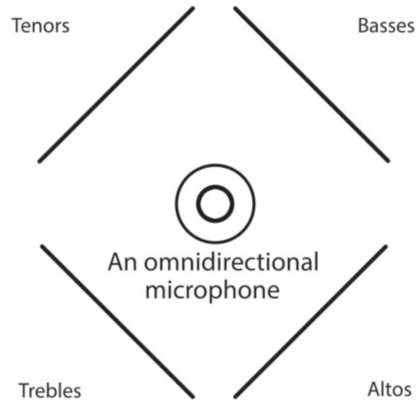

Fig. 39. Choir arrangement in a square formation angled at 45 degrees relative to an omnidirectional microphone. The simplest "home" solution for beginners. It is not used in professional sound production – can be used as a recording of a choir rehearsal.

Alternatively, the same omnidirectional microphone could be employed in a less traditional setting. Each voice could stand in a row – or form multiple rows in a larger ensemble – with all members facing the microphone as shown in figure 39. Such an arrangement allows placing vocalists at various distances from the microphone to achieve a more optimal result and an improved timbre of each voice, as well as the entire ensemble.

Chapter 2. Speaker parameters and types. Arrangement and calibration of studio monitors

Loudspeakers are electroacoustic transducers which convert electrical energy to acoustic energy and emit output signals into closed or open spaces. Many types of loudspeakers, including magnetic, electrostatic, and piezoelectric devices, feature an electromechanical transducer which puts the diaphragm in motion. Some (based on laser, ion, or plasma) operate on a fundamentally different principle. However, as their design prevents stage or studio use, and a sound technician will not encounter them in daily work, they will not be discussed in this book.

The two most common types of speakers are divided by the type of their acoustic energy emitter into:

a) cones – electroacoustic devices which emit signals directly and use the diaphragm for energy propagation. The diaphragms are made of various materials and can take different shapes and sizes.
b) horns – electroacoustic devices which emit signals indirectly and use the horn mouth for energy propagation. The horn may be made of various materials.

Not all cone speakers actually look like a cone. Their diaphragm may also take the shape of an ellipse, resemble a sphere or a cylinder, and so on.

Cone-shaped speakers are usually manufactured with magnetic and piezoelectric transducers, whereas speakers with a cylindrical, flat, or another similar diaphragm typically use electrostatic transducers[38].

Years of research and laboratory experiments have shown that the best speakers on the market – considering electrical, mechanical, musical qualities and the costs – are electroacoustic cone devices because of their use of the magnetic coil. They are commonly manufactured for consumer purposes and account for almost 100% of products regardless of the application (professional, amateur, studio, onstage, home, etc.). Other types of speakers are usually produced for the military or research, development or academic purposes[39].

Speaker units usually feature dynamic (electromagnetic) drivers which exploit electrodynamic power produced when the electrical current flows through the magnetic field. The construction of the speaker follows the reverse design of a dynamic microphone. Thus, an electromagnetic speaker is essentially an engine with an attached diaphragm, which activates with electricity and electromagnetic power.

The construction of the speaker relies on a permanent magnet with a persisting magnetic field around the coil. If direct current flows through the coil, it will form another magnetic field, with the orientation of coil displacement (including the diaphragm) depending on flow direction (field polarity). However, if the current is alternating, the movement of the coil will match the changing direction of the current. As the coil swings back and forth, the attached diaphragm animates the surrounding air particles to create a sound wave corresponding to the wave generated by the amplifier[40].

Construction of a loudspeaker with decent parameters is not easy. The weight of the coil and the membrane make it extremely difficult to eliminate speaker latency. The softness of the membrane (lack of sufficient rigidity) or poor quality of the suspension may produce distortions manifesting as resonance and non-linear response. A traditional dynamic loudspeaker creates non-linear disturbances (around 2–3%) caused by the uneven distribution of the magnetic field, the noise produced by the moving coil, and the spiders[41].

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38 Żyszkowski Z. (1984). op. cit., p. 311.
39 Ibidem.
40 Borwick J. (2001). *Loudspeaker and headphone handbook*, 3rd edition, Focal Press, Oxford, pp. 54–60.
41 Żyszkowski Z. (1984). op. cit., pp. 61–64.

A loudspeaker with a flat frequency response is a technical impossibility (just like a corresponding microphone). Linear generation of extremely high and low frequencies by a single speaker is virtually infeasible since the concurrent reproduction of high- and low-frequency sounds causes intermodulation distortions. High-amplitude low frequency sounds emitted by the speaker move the apparent source of high-frequency sounds closer and farther away, which may bring about unwanted changes in the frequency of the high pitch (the Doppler effect – a similar phenomenon occurs when a passing train produces audible up-and-down shifts in the high pitch)[42].

Loudspeakers are categorized by the frequency range they can reproduce (considering their relatively flat response) into:

a) subwoofers (up to 20–150 Hz),
b) woofers (20–400 Hz),
c) mid-range speakers (4000–5000 Hz),
d) mid-range speakers and tweeters (mid-high range speakers) (800–20 000 Hz),
e) tweeters (5000–20 000 Hz),
f) super tweeters – convert ultrasounds,
g) full-range – convert most of the human hearing range[43].

2.1. Speaker parameters

2.1.1. *Efficiency*

Efficiency is a parameter of critical importance. It shows how much electrical energy fed to the device will be converted into acoustic energy. The efficiency of dynamic speakers ranges from 0.5–14%. The remaining energy is converted to heat. Importantly, only state-of-the-art designs reach a 14% efficiency level, often to the detriment of other parameters such as audio response and non-linear distortion. Efficiency is provided in decibels (dB) and determines the sound intensity generated at 1 m from the speaker when the power of 1 watt is delivered. The magnetic field concentrates on

[42] Whitaker J. C., Benson B. K. (2002). op. cit., pp. 5–34, 5–40.
[43] Sztekmiler K. (2003). op. cit., p. 92.

the coil located in the magnet gap. Higher parameter values indicate greater efficiency[44].

In Hi-Fi equipment, efficiency ranges from 84–92 dB / W on average, typically 89 dB / W. Professional stage speakers have an efficiency in the range of 96–105 dB / W (for woofers and mid-range speakers, since tweeters have an efficiency of 101–115 dB / W)[45].

2.1.2. *RMS power*

The maximum power that can be delivered to the speaker continuously for more than two hours (ha) without causing non-linear distortions or damage to the device (according to catalogue data of multiple companies, the tests are frequently performed for 100 ha, with the results provided in watts)[46].

2.1.3. *Program power handling*

The power that can be delivered to the speaker over the time corresponding to 1 second (s) without causing non-linear distortions or damage to the device. The program power is usually twice the RMS power of the speaker (expressed in watts)[47].

2.1.4. *Peak power handling*

The power (impulse) that can be delivered to the speaker for up to 10 milliseconds (ms) without causing non-linear distortions or damage to the device. Peak power can be up to four times greater than the RMS power (expressed in watts).

2.1.5. *Frequency response characteristics*

The level of speaker efficiency in the scope of frequency response. As discussed above, the construction of an ideal speaker is an unattainable feat. However, some speakers are single-purpose devices only (e.g. compatible with bass guitars only). Bass speakers have a flat audio response in the low-frequency range. When connected

[44] Whitaker J. C., Benson B. K. (2002). op. cit., pp. 5–32, 5–33.
[45] Sztekmiler K. (2003). op. cit., p. 93.
[46] Ibidem, p. 93.
[47] Long M. (2006). *Architectural Acoustics*, Academic Press, San Diego, pp. 638–639.

to an electric guitar (which has a higher frequency range), a bass speaker alters its sound due to the frequency response which suits the designated purpose of the speaker. Such alterations could disturb the frequency proportion between high and low sounds and produce an unnatural timbre[48].

2.1.6. *Bandwidth*

Provided separately for single drivers and entire speaker units. This parameter is usually used when the efficiency of the speaker or speaker unit drops by 10 dB. Bandwidth limits are defined by the highest and the lowest value of the frequency response. Studio monitors and highly specialized stage speakers have a bandwidth of 3 dB or less. A parameter of efficiency reduction of 3 dB or less is provided only for high-quality and specialty-use speaker units because their frequency response is usually far flatter than that of traditional units. A flatter frequency response suggests a more neutral sound, with less coloration of specific frequencies, and a relative fidelity of the sound generated by the speaker[49].

2.1.7. *Impedance*

Impedance refers to the size of the speaker set, whereas resistance is a real value measured at the terminals and expressed in ohms. Speakers typically have a resistance of 4 or 8 Ω, whereas Hi-Fi sets may exhibit a 6 Ω value. Note that impedance is not a constant but changes with sound frequency. That is why it may not always match the values provided by the manufacturer (and constitutes a frequent cause of underperforming amplifiers)[50].

Since failure-free sound reinforcement is significantly dependent on the amplifier, the careful choice of a device eliminates most issues. First and foremost, the RMS power of the amplifier should not surpass the RMS power of the speaker. A smaller and weaker amplifier can be combined with a louder speaker set on the condition that the impedance of the set is not smaller than the minimum

48 Sztekmiler K. (2003). op. cit., p. 94.
49 Ibidem.
50 Self D. (2009). *Audio engineering*, Newnes/Elsevier, Amsterdam, Boston, p. 47.

impedance required by the amplifier. A reverse combination may damage the speaker system[51].

2.2. Cone speaker

The cone is the best-known speaker type, employed in most music listening devices. Its construction relies on a cone-shaped paper diaphragm coupled to the coil and receiving vibrations induced by coil displacement in the magnetic field. Cone speakers feature in multiple environments such as Hi-Fi equipment, cars, small radios, big speaker sets, and intercoms. Their sizes range from 3–18 inches (diameter measured through the centre of the speaker)[52].

As already discussed, the coil and the diaphragm form a vibrating element which carries the sound wave (fig. 40). The diaphragm is attached to a dedicated frame or basket in two places jointly referred to as the suspension. The part placed in proximity to the coil is the spider, whereas the part on the edge of the speaker, generally made up of several layers and secured to the frame, is the surround. The central dome (or cap) covers the coil moving in the gap and protects the device against dirt. Power connection clips are mounted on the terminals in the rear of the basket (next to the permanent magnet)[53].

Just like microphones, loudspeakers have directionality described in the section on microphone parameters. Note that the pickup pattern of a cone speaker results from the relation between speaker size and sound wavelength. The longer the wave, the more omnidirectional the pattern. Conversely, as sound frequency rises and wavelength drops, the pattern becomes narrower (a critical piece of information for sound engineers handling concerts and studio work)[54].

[51] Borwick J. (2001). op. cit., p. 268; Watkinson J. (2013). *The Art of Sound Reproduction*, 2nd edition, Focal Press, Burlington, pp. 189–190.

[52] Whitaker J. C., Benson B. K. (2002). op. cit., pp. 5–28; White H. E., White D. H. (2014). *Physics and Music: The Science of Musical Sound*, Dover Publications Inc., Mineola, pp. 307–308; Brice R. (2001). *Music Engineering*, Newnes, Oxford, p. 416.

[53] Borwick J. (2001). op. cit., pp. 66–68; Ballou G. (2009). *Electroacoustic devices: microphones and loudspeakers*, Focal Press, Amsterdam – Boston, pp. 201–202; Newell P. R., Holland K. (2007). *Loudspeakers for music recording and reproduction*, Focal Press, Oxford, pp. 23–28.

[54] Butler T. (1994). op. cit., pp. 124–125.

If the cone speaker features in a speaker unit as the only sound source in a large room, low sounds radiate across the entire space (omnidirectional wave propagation). However, the mid- and high-frequency sounds are best processed and perceived only to the person standing in front of the speaker. That is why advanced users (in a studio or onstage) employ many constructions and combinations of speakers to elicit optimal sound for the particular purpose and conditions. The selection of speaker type or size often depends on the characteristics of the playback room or the outdoor space in need of sound reinforcement[55].

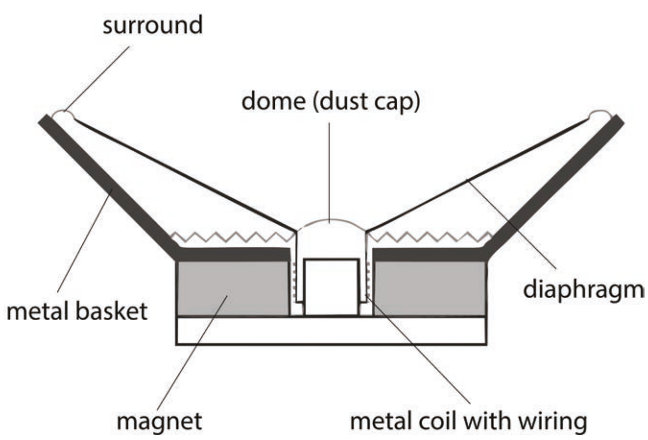

Fig. 40. The structure of a cone speaker.

2.3. Horn speakers

Horn speakers are used for reproducing mid- and high-range frequencies (fig. 41 a) – b)). They are light, relatively compact, and exhibit a smaller diaphragm displacement amplitude (when compared to the cone diaphragms).

Despite their many advantages, cones have one fundamental drawback which prevents their use for energy conversion at

[55] Sztekmiler K. (2003). op. cit., p. 92.

tremendous powers – small efficiency of up to 5%, which puts column designers in quite a predicament. Extending the diameter of the diaphragm invariably increases its mass (and the mass of the coil), impairing average speaker performance. Currently, the size of speakers used in columns is proportional to the gain obtainable with the present-day technology without raising costs or lowering the average performance.

Speaker efficiency can increase with the separation of the diaphragm and the surrounding space with a properly shaped horn which acts as an acoustic transformer. The horn mouth is better suited to energy propagation than the membrane of a cone speaker, so more energy carried to the tube may be radiated outside.

The name of horn speakers comes from their original casing. These devices exhibit extreme acoustic efficiency, usually ranging up to 40%, even though some constructions can radiate even 80% of the acoustic energy. Accordingly, they are used for the generation or high acoustic pressure[56].

Proper operation of a pressure speaker requires an electromagnetic coil system. The coil is located in the gap of the permanent magnet (just like in a cone), where it remains coupled to a dome-shaped (rather than a cone-shaped) diaphragm[57]. The speaker uses a phase plug which modifies the lengths of the waves emitted by various parts of the diaphragm. The phase plug is attached to the horn mouth and sheathed with a dedicated cover which gives it protection against dust and dirt[58].

The phase plug-in speaker design is the result of multiple trade-offs. In an ideal speaker, the dome-shaped diaphragm should take the form of a flat disc, which is practically unfeasible. Above all, the dome is stiffer and simpler to manufacture. In this case, a phase plug simultaneously guides sounds from various places to the same point. Its absence could disturb the flat frequency response of the device and cause wave interference which, in turn, could impair the quality of sound signal processing[59].

Compression speakers are single units consisting of a diaphragm and a right horn. The assessment of their performance,

[56] Żyszkowski Z. (1984). op. cit., pp. 379–380.
[57] Rossing T. D., Fletcher N. H. (2004). op. cit., p. 246; Stark S. H. (2003). op. cit., p. 141.
[58] Borwick J. (2001). op. cit., pp. 30–36.
[59] Sztekmiler K. (2003). op. cit., p. 92.

audio response, and efficiency should be based on the cooperation of the speaker with the mounted horn (thus, only complete devices should be tested). Compression speakers are not intended for use in open spaces alone. In combination with cones, they serve to compensate for the limitations of the devices[60].

Fig. 41. a) and b) Various shapes of horn speakers.

2.4. Passive speaker units

A passive speaker unit does not have a built-in power amplifier, which may be an advantage or a disadvantage (depending on the application)[61].

A single speaker unit consists of several drivers which convert selected segments of the bandwidth. The signal is separated into smaller frequency bands by the so-called crossover. This device guides particular signals to the drivers suitable for converting the identified frequencies, which yields flatter frequency response

[60] Eiche J. F. (1990). *The Yamaha Guide to Sound Systems for Worship*, Yamaha Corporation of America, Buena Park, pp. 252–253; Butler T. (1994). op. cit., pp. 125–130.

[61] Huber D. M., Runstein R. E. (2014). *Modern Recording Techniques*, 8th edition, Focal Press, Burlington, p. 554.

characteristics of the output and compensates for intermodulation distortion[62].

In a passive unit, the audio signal distributed by the mounted crossover (also passive) is already amplified. Depending on the crossover and the number of drivers, we can distinguish two-, three-, and four-way speaker units[63].

2.5. Active speaker units

An active speaker unit has the crossover and the power amplifier mounted inside the enclosure. An active crossover splits the signal into frequency bands before amplification. Later on, separate power amplifiers reinforce the signal in all bands[64].

Significant complications in constructing a crossover which would filter well multiple frequencies, as well as the associated problems with achieving a flat frequency response, convinced many manufacturers to opt for active designs. A digitally controlled crossover splits the audio signal with far greater accuracy and significantly narrows down the transition regions in the division of particular frequencies[65].

In contrast to a passive speaker unit, an active enclosure has many safeguards against burnout or damage because amplifiers are explicitly selected for the mounted drivers. In addition, active units use equalization, which compensates for imperfect room acoustics or phase displacement[66].

With speaker sets intended for concert use, the difficulty lies in the generation of high powers, which is related to heat abstraction issues (due to the amplifier mounted inside the enclosure). The necessity to power every unit is another common headache[67].

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62 Sztekmiler K. (2003). op. cit., p. 94.
63 Eiche J. F. (1990). op. cit., pp. 77–78, 99; Izhaki R. (2012). *Mixing Audio: Concepts, Practices and Tools*, 2nd edition, Focal Press, Waltham, pp. 77–78.
64 Harrington R., Carman R. (2010). *Video Made on a Mac: Production and Postproduction Using Apple Final Cut Studio and Adobe Creative Suite*, Peachpit Press, Berkeley, p. 162; Izhaki R. (2012). op. cit., pp. 77–78; Izhaki R. (2008). *Mixing Audio: Concepts, Practices and Tools*, Focal Press, Burlington, pp. 81–82.
65 Borwick J. (2001). op. cit., pp. 275–279.
66 Eiche J. F. (1990). op. cit., pp. 77–78, 99.
67 Rumsey F., McCormick T. (2009). op. cit., p. 90.

2.6. Studio monitors

Active speaker units are employed to a good effect in recording studios (where amplifiers and drivers must remain in perfect harmony). The safeguards eliminate the risk of column blow-out which could occur during cable-switching, to name just one possibility. Those units are also highly resistant to cracks and other sudden, loud sounds which could be caused by switching cables on the sound signal path[68].

Note that active systems (playback monitors) used in recording studios feature two or three independent power amplifiers (in bi-amp and tri-amp systems, accordingly). They are designed to reinforce particular frequencies after the digital crossover has filtered the signal. All amplifiers are mounted in the interior casing on the column which requires a single power supply[69] (fig. 42–50).

The use of several amplifiers designed for handling particular frequency bands aims to enhance the audio response of playback monitors for various frequencies and to reflect the original (real) instrumental timbre when the musical material is recorded[70].

Another important issue is the power of playback monitors connected to the music card of the computer. It is an individual choice, which depends on the application of the device, the needs and finances of the user, and the cubic area of the room used for mixing or mastering.

The following studio monitors are distinguished.

1) Near-field – compact devices with a small speaker diameter, used in small recording studios, postproduction studios, semi-professional studios, and home studios[71]. Usually positioned on the engineer's desk on isolation stands; deployed for confronting the listening material with the use of various monitors[72].

   Sound postproduction is a processing stage completed to achieve the right sound on various speaker systems. It refers to the tasks performed after filming on the audio which

68 Harris B. (2009). *Home Studio Setup: Everything You Need to Know from Equipment to Acoustics*, Focal Press, Burlington, p. 94.
69 Izhaki R. (2012). op. cit., pp. 77–78.
70 Sztekmiler K. (2003). op. cit., pp. 95–96; ibidem, pp. 77–78.
71 Harris B. (2009). op. cit., pp. 94–95; Calderan P. (2007). *Musica con il PC: Creare musica con uno studio di registrazione digitale*, Apogeo, Milan, pp. 45–47.
72 Alten S. R. (2011). op. cit., pp. 53–54.

consists of several independent overlapping sound layers mixed together. The audio includes:

a) basic information: dialogue, conversation chatter, narration;
b) sound effects: Foley effects, such as the sound of a dog running or a spoon dropping to the table, nonsynchronous sounds[73], which have no distinct tempo or no clear beginning and end, such as many cars simultaneously driving through the street, special effects (SFX), background noise and the so-called ambience (or *atmos*) such as the quiet music of a string quartet, sound design, e.g. machinery noise generated by synthesizers, various synthesizers and oscillators;
c) music: immanent (source) music, relevant within the frame, and background (commentary) music outside the frame, which may not have direct relevance to the plot[74].

2) Mid-field – used in medium-sized rooms; usually sit on special stands behind a mixing console or a computer; deployed in project and specialist studios; located a minimum of 1.5m away from the engineer to reflect the full range of sound frequencies[75].
3) Far-field – high-power, expensive, and large woofers (even up to 15”) used in huge, specialized recording studios. Placed on stands or mounted in the walls of the playback room. Usually located up to a dozen meters from the engineer; often combined with near-field monitors to check the sound of the material in various sound reinforcement systems and acoustic conditions[76].

The monitors discussed above are used in recording studios. Each of them is an active column operating with specialist inbuilt monitors perfectly suited to the entire construction. They are designed by multiple manufacturers and selected to match the crossover and drivers of the particular model.

[73] See: http://kabumstudio.vxm.pl/oferta/postprodukcja-dzwieku [accessed: 24.06.2022].
[74] Clark B., Spohr S. J. (2013). *Guide to Postproduction for TV and Film Managing the Process*, 2nd edition, Focal Press, Burlington, pp. 15–16, 24–25, 27–28.
[75] Harris B. (2009). op. cit., pp. 94–95.
[76] Alten S. R. (2011). op. cit., pp. 52–53; Harris B. (2009). op. cit., pp. 94–95.

Fig. 42. a) and b) Front and back of an active near-field studio monitor with a circular bass reflex vent at the back.

Fig. 43. a) and b) Front and back of an active near-field studio monitor with a bass reflex port in the form of a long and narrow rectangular vent, mounted at the front.

Fig. 44. a) and b) Front and back of an active near-field studio monitor with a bass reflex port in the form of a long and narrow rectangular vent, mounted at the back. One of the sides for low-frequency propagation is wider and semi-circular in shape.

Fig. 45. a) and b) Front and back of an active near-field studio monitor with a circular bass reflex vent at the back.

Fig. 46. a) and b) Front and back of an active near-field studio monitor with two circular bass reflex vents at the front, between drivers.

Fig. 47. a) and b) Front and back of an active near-field studio monitor with two circular bass reflex vents at the front, positioned in a vertical line.

Fig. 48. a) – d) Front and back of an active near- and mid-field studio monitor with two circular bass reflex vents at the front together with a subwoofer for processing low frequencies. In this case, the subwoofer expands the range of the low frequencies processed, complementing primary drivers. Its bass reflex port is a long and narrow rectangular vent with rounded sides, mounted at the back.

a b

Fig. 49. a) Studio active column (front panel) and
b) Studio active column (rear panel).

a b

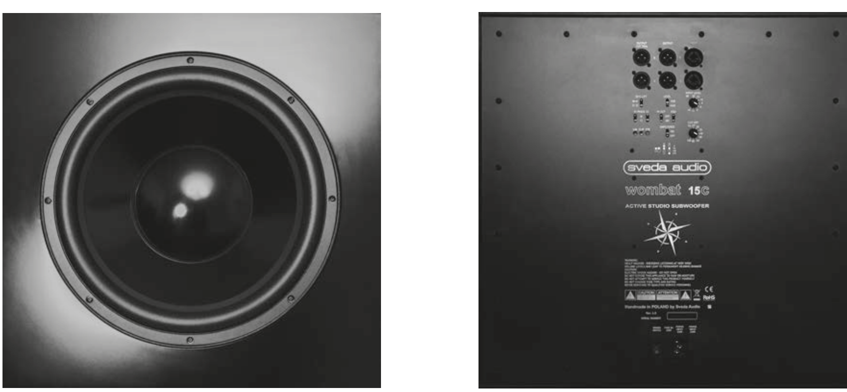

Fig. 50. a) 15” studio active woofer (front panel) and
b) 15” studio active woofer (rear panel).

Monitor controllers are devices which redirect an audio signal to different monitoring systems during recording and playback (fig. 51–54). The primary requirement is that they not distort the signal (which in this case means they do not change the timbre). In other words, they should be acoustically "transparent". This quality is crucial, because the signal reaching the amp is distributed to different sets of audio monitors, in order for the audio engineer to hear how the material will sound on different kinds of speakers or columns. The engineer has the ability to turn each speaker system on and off, or to engage all of the studio's monitors at once. In addition, some monitor controllers have a talkback microphone installed, which lets the engineer speak to the ensemble in the studio with a simple push of a button.

Fig. 51. Monitor controller for various studio monitors (front panel).

Fig. 52. Monitor controller for various studio monitors (rear panel).

Fig. 53. Monitor controller for various studio monitors (front panel).

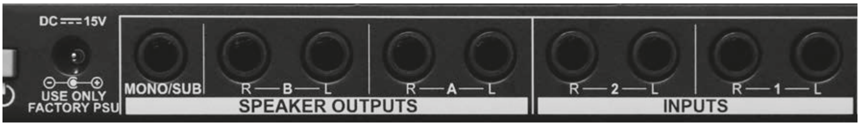

Fig. 54. Monitor controller for various studio monitors (rear panel).

2.7. Adaptation to room acoustics and studio monitor parameters

To adapt the monitors to local acoustic conditions, sound engineers frequently examine the room with a specifically generated soundwave referred to as noise. When the microphone is connected to a full-duplex sound card which simultaneously plays and records, room acoustics can be brought under scrutiny, just like the frequency response and sound reflection. This information, combined with practical knowledge on how to use audio data, may help identify not only the right inclination of the monitors but also their optimal placement in the room[77].

The monitors themselves can equalize high and low frequencies by boosting or cutting a selected frequency band – usually provided by the manufacturers on the back baffle of the speaker unit and oscillating around 2–6 dB, with a jump every 2 dB. The volume control in the form of a rotary potentiometer and the high-pass filter used when working with an auxiliary subwoofer are also located on the back. Anti-vibration tabs and special sponges for closing the bass-reflex openings are often included with the product.

Monitor parameters resemble those of a regular speaker. However, manufacturers tend to add supplementary information to specify the intended application and the class of the device. In general, the following parameters are provided[78]:

a) precision of low-frequency conversion, including diaphragm latency,
b) frequency response,
c) THD (total harmonic distortion)[79],
d) compression audibility,
e) the power of specific amplifiers used in the monitor,
f) phase distortion[80].

2.8. Setting monitors in the control room

The parameters provided by studio monitor manufacturers include the distance between two monitors. This information helps the user to correct a wide stereo base, which should not be further broadened to avoid sound distortion and directionality shifts. Excessive distance between speakers could disrupt the stereo base and produce unnatural sounds of particular instruments in the stereo panorama[81], which may lead to the hole-in-the-middle effect. On the other hand, placing the devices too close to each other

77 Borwick J. (2001). op. cit., p. 283–284.
78 Alten S. R. (2011). op. cit., pp. 42–50.
79 Whitaker J. C., Benson B. K. (2002). op. cit., pp. 5–34, 5–40; Self D. (2009). op. cit., pp. 74–75.
80 Whitaker J. C., Benson B. K. (2002). op. cit., pp. 5–34, 5–40.
81 Newell P. R., Holland K. (2007). op. cit., pp. 225–227.

could narrow down the stereo field (and thus make the stereophony inaudible).

Studio monitors have a minimum listening distance, measured from the ears of the sound engineer to the speaker unit. The gap often determines the range of frequencies correctly reproduced by the system. Note that since the speakers start to sound well only when you step away (like a traditional musical instrument), the identification of the right distance between the speakers and the engineer is essential[82].

2.8.1. *Monitor configuration*

The simplest setup which ensures optimal sound is to position your speakers at the three corners of an equilateral triangle (fig. 55), which automatically identifies the listening area and the sweet spot[83]. The isosceles triangle design should be avoided as it may produce a different frequency response[84].

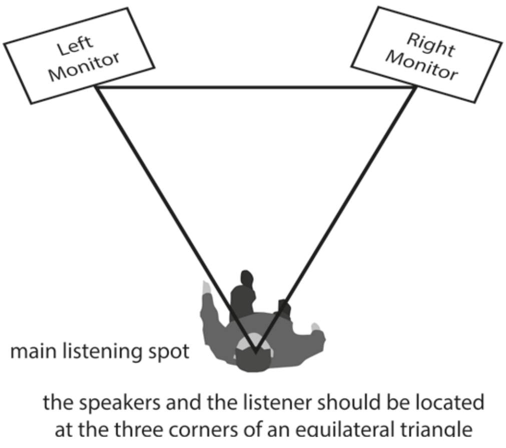

Fig. 55. Playback monitors in the equilateral triangle formation in the control room of a recording studio.

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82 Whitaker J. C., Benson B. K. (2002). op. cit., pp. 1–63, 1–64.

83 Newell P. R., Holland K. (2007). op. cit., pp. 202–204.

84 Alten S. R. (2011). op. cit., pp. 48–55; Holman T. (2008). *Surround Sound: Up and Running*, 2nd edition, Focal Press, Burlington, pp. 36–37; Savage S. (2011). *The Art of Digital Audio Recording: A Practical Guide for Home and Studio*, Oxford University Press, New York, pp. 16–17; Izhaki R. (2012). op. cit., pp. 85–86.

2.8.2. *Monitor set-up*

If the monitors sit on the engineer's mixing console, it is a nice touch to locate them on anti-vibration stands. Monitor units should be positioned at the level of the engineer's ears. If they are higher, slant them downwards to achieve equal reproduction of all constituent frequencies (when the speakers are fitted in wall niches or on dedicated stands). Monitors may also sit directly behind the controls of the mixing console[85]. However, such a solution leads to the unwanted comb filtering of lower frequencies. This is because the sound reflected from the console arrives later than the direct sound, which is the main reason for this anomaly[86].

2.8.3. *Monitor orientation*

Some studio monitors operate properly only in one position – horizontal or vertical. Therefore, changing their orientation on your own may prove disadvantageous and significantly affect the quality of the reproduced sound, disrupting its balance[87].

2.8.4. *Monitor orientation in relation to tweeters*

Another proposed monitor placement takes into account the position of tweeters. This placement indicates the suggested location of mid-range speakers and tweeters. When the latter are placed on the outside (to the sides), the listener experiences a wider stereo base – which may be a positive or a negative feeling[88]. The orientation of some monitors should not be changed from vertical to horizontal and vice versa, as it could degrade the stereo base and the sound. Attempts to achieve the best acoustic conditions in a particular room (by trial and error) in this manner should be made only by experienced users[89].

85 Owsinski B. (2006). *The Mixing Engineer's Handbook*, 2nd edition, Artistit-Pro Publishing – Thompson Course Technology, Boston, p. 73.
86 Alten S. R. (2011). op. cit., pp. 53–54; Newell P. R., Holland K. (2007). op. cit., p. 211; Newell P. (2008). *Recording Studio Design*, 2nd edition, Focal Press, Burlington, pp. 523–524; Holman T. (2008). op. cit., p. 51.
87 Whitaker J. C., Benson B. K. (2002). op. cit., pp. 1–64, 1–68; Owsinski B. (2006). op. cit., p. 74.
88 Newell P. R., Holland K. (2007). op. cit., pp. 198–205, 216–217.
89 Borwick J. (2001). op. cit., pp. 471–493.

2.8.5. *Settings of the console in relation to the monitors*

With proper calibration of the console, the monitors transmit the real sound image. It is a foregone conclusion that the devices should never be connected at random. Studio monitors and the console are the two most important devices of the entire signal path, so their calibration is highly recommended. Firstly, both devices need to be appropriately regulated to avoid overdrive in the signal path during volume adjustment. Secondly, both devices must be positioned so that the audio, when processed through the mixing console and the speaker units, reflects the real sound image without unwanted frequencies which could significantly affect music reception. Particular frequency bandwidths may be boosted or attenuated using a graphic equalizer, which usually includes a set of slide potentiometers. The tool should be employed for suppressing the bands and not exaggerating them to compensate for acoustic defects caused by flawed room acoustics. The graphic equalizer causes far-reaching sound changes, so it should be used with caution.

Chapter 3. Selected commercial standards for surround sound and audio coding standards

Multichannel audio is an evolution of stereophony. The technology relies on multiple speakers to give a sense of the surrounding space or introduce special effects in the playback. It allows for playing three or more independent channels concurrently. Every additional speaker (or channel) enhances spatialisation, which is why speakers are placed to the sides and the back of the listener.

Multichannel solutions complement multimedia presentations, computer games, surround music, and movies. In general, the technology is used in the cinemas, home theatres, gaming consoles, and dedicated audio sets equipped with a surround device. Multichannel audio can be recorded on special discs in the DVD-Audio (DVD-A) or Super Audio CD (SACD) standards. It offers a novel form of musical expression and unique quality of virtual space.

3.1. Multichannel playback systems

As you can see in the figures 56. and 57., the layouts of a multichannel playback system for cinema and home use are strikingly different. An auditorium is a far larger space, so adequate amplification is of the essence. Home studios use Hi-Fi standards based on professional cinema solutions but specifically adapted for home use.

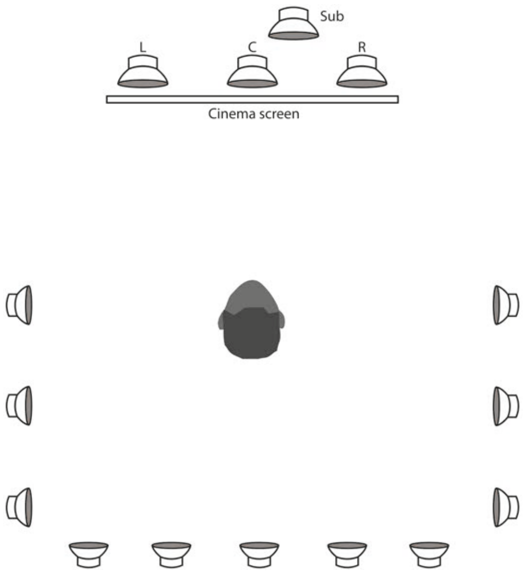

Fig. 56. Layout of one of many multichannel playback system for cinema use.

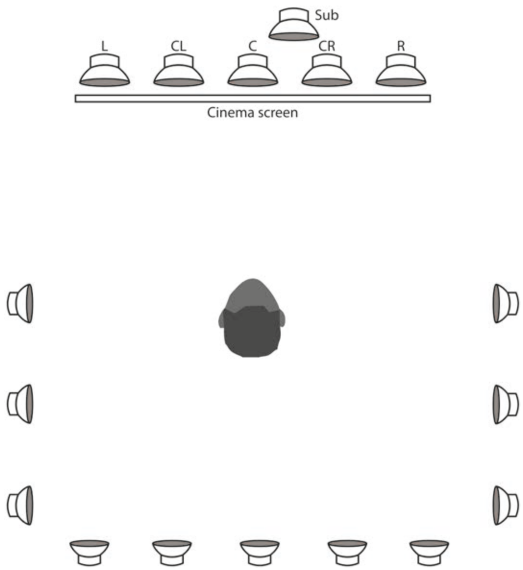

Fig. 57. A variation on the traditional multichannel playback system for cinema use, with an increased number of loudspeakers behind the screen.

3.1.1. *Quadraphonic playback*

The term refers to the speaker arrangement which composed of four speakers (fig. 58 a) – b)). Quadraphony did not catch on in musical productions because of playback issues involving the problems in the precise spatial localization of the audio (the sound is abruptly cut from one speaker and fed by another, instead of completing a smooth transition)[90].

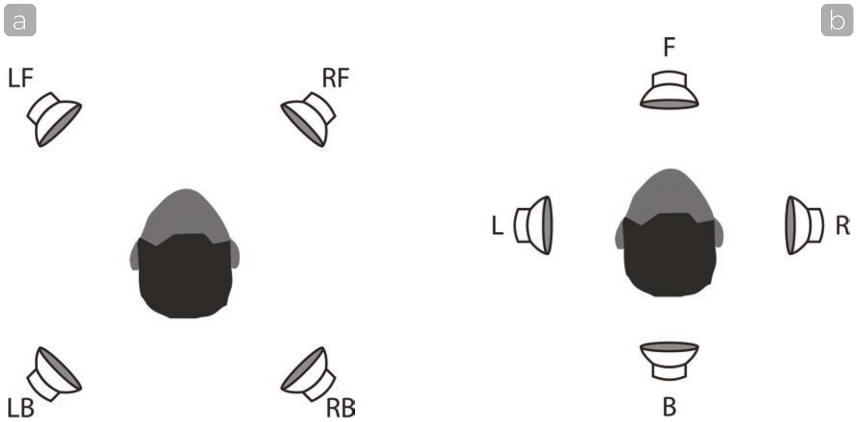

Fig. 58. a) and b) Positioning variants for speakers in quadraphonic monitoring.

3.1.2. *Dolby Surround*

The first commercial multichannel system used in analog technology. Dolby Surround includes four speakers – two in the front and two in the back – with the latter pair used only for transmitting surround effects[91].

3.1.3. *Dolby Pro Logic*

An analog surround system which is an extension of Dolby Surround. For proper operation, it requires plugging in five speakers,

[90] Slavik K. M., Weinzierl S. (2008). "Wiedergabeverfahren", in: Weinzierl S. (ed.). *Handbuch der Audiotechnik*, Springer, Berlin, pp. 612–613.

[91] Adams J. J., Wolenik R. (2002). *Build Your Own Home Theater*, 2nd edition, Newnes, Woburn, p. 121.

including stereo channels (left and right), one central channel, and two surround channels for special effects. However, they exhibit a boosted frequency response in the centre of the aural bandwidth. Another added feature is the central channel for speech conversion. With that addition, the audience feels that the actors are speaking directly to them. The desired effect is achieved by properly placing the speaker in the room. For instance, in a cinema, it will be situated behind the screen, whereas at home – above or under the TV set[92].

3.1.4. *5.1 System*

The system consists of five broadband speakers, including a central speaker, two front speakers and two rear speakers. One additional subwoofer completes the set. The system is generally used for the reproduction of sound. It comprises six independent audio channels[93] (fig. 59 and 60). 5.1 may be complemented with technologies such as Dolby Digital, Dolby Pro Logic II, DTS (discussed below).

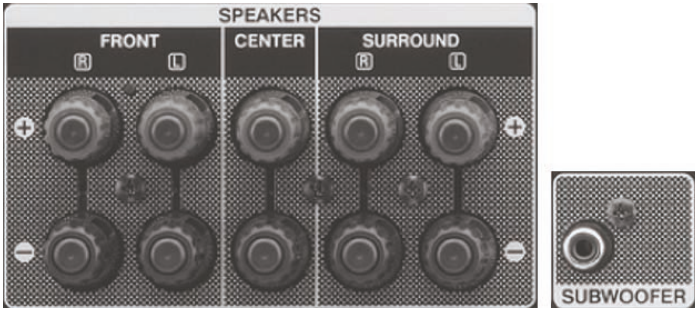

Fig. 59. A device for loudspeaker connection in the 5.1 surround sound system – a fragment of the back panel.

[92] Shepherd A. (2003). *Pro Tools for Video, Film, and Multimedia*, Muska & Lipman Publishing, Boston, p. 206; Briere D., Hurley P. (2009). *Home Theater for Dummies*, 3rd edition, John Wiley & Sons, Hoboken, p. 37; Friesecke A. (2007). *Die Audio-Enzyklopädie: ein Nachschlagewerk für Tontechniker*, K. G. Saur Verlag, München, pp. 755–756.

[93] Shepherd A. (2003). op. cit., pp. 206–208; Gupta R. G. (2006). *Television Engineering and Video Systems*, Tata McGraw-Hill Publishing Company Limited, New Delhi, pp. 386–387.

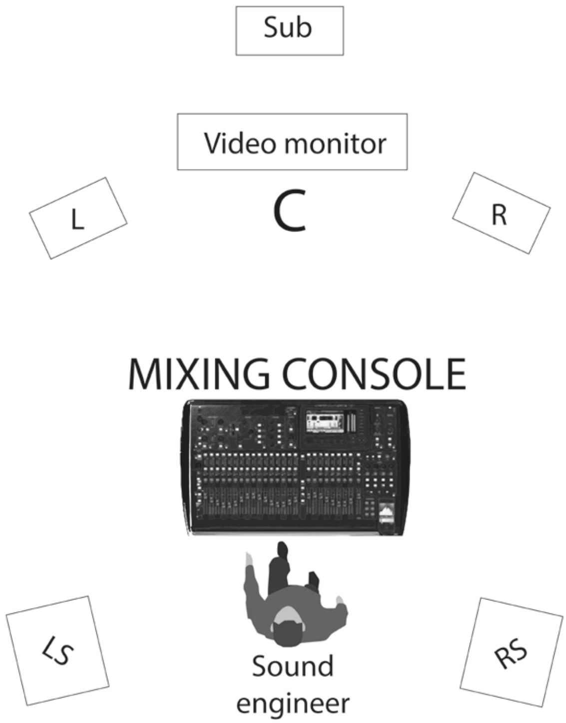

Fig. 60. Surround sound system placement in the control room of a recording studio relative to the sound engineer.

3.1.5. *Dolby Digital*

A digital surround sound system designed in Dolby laboratories and the direct successor of Dolby Pro Logic. Dolby Digital materially differs from its predecessor as a digital device, which can be used to reproduce digitally recorded sound in superior quality. Furthermore, in comparison to analog systems, it limits audio signal loss during conversion with the use of digital transducers. The system contains a pair of channels (Left Surround and Right Surround). In addition, rear columns have frequency response characteristics which correspond to the accepted model of human hearing (expansion in contex to the previous system). As a result, the sound augments the realism of any scene it accompanies. When compared to the previous-generation system, the drivers of all columns have better acoustic parameters, ensure a higher quality of sound conversion, and enrich the signal with LFEs (Low Frequency Effects) thanks to the separate subwoofer channel[94].

3.1.6. *Dolby Pro Logic II*

Another version of Dolby Pro Logic which can handle six separate audio channels recorded in analog (5.1). The system eliminates the biggest flaw of its predecessor by adding in the rear the full-range surround speakers which convert discrete audio channels. The particular audio encoding requires the use of a dedicated decoder which correctly plays all the six audio channels. However, this requirement is met by modern tuner amplifiers[95]. Additionally, Dolby Pro Logic II is compatible with larger sound processing systems such as 6.1 and 7.1.

3.1.7. *Digital Theatre System*

A digital surround sound system technologically comparable to Dolby Digital since both come equipped with five channels and a subwoofer (fig. 61 and 62). On digital devices, Digital Theatre System is often indicated by the acronym – DTS. DTS and Dolby Digital are competitive and incompatible systems[96].

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94 Shepherd A. (2003). op. cit., pp. 206–207; Rumsey F., McCormick T. (2006). op. cit., pp. 486–488; Farkas B., Govier J. (2003). *Use Your PC to Build an Incredible Home Theater System*, APress Media, Berkeley, p. 74; Friedman G. (2012). *The Complete Guide to Sony's NEX-7 Mirrorless Camera: The Friendly Manuals with Professional Insights*, The Friedman Archives Press, Plymouth, pp. 330–331; Kerins M. (2011). *Beyond Dolby (Stereo): Cinema in the Digital Sound Age*, Indiana University Press, Bloomington – Indianapolis, pp. 13–14; Jaiswal R. C. (2009). *Audio-Video Engineering: With Important Colour Figures*, Nirali Prakashan, Pune, pp. 4.11–4.12.

95 Angeli D. (2009). *Pro Tools for Film and Video*, Focal Press, Burlington, p. 107; Friesecke A. (2007). op. cit., pp. 760–761; Rushing K. (2006). *Home Theater Design. Planning and Decorating Media-Savvy Interiors*, Quarry Books, Gloucester, p. 33.

96 Lundstörm L. I. (2006). *Understanding Digital Television. An Introduction to DVB Systems with Satellite, Cable, Broadband, and Terrestrial TV Distribution*, Focal Press, Amsterdam, Boston, Heidleberg, London, New York, Oxford, Paris, San Diego, San Francisco, Singapore, Sydney, Tokyo, pp. 255–256; Millward S. (2007). *Fast Guide to Cubase 4*, PC Publishing, Thetford, pp. 369–370; Kerins M. (2011). op. cit., pp. 13–14.

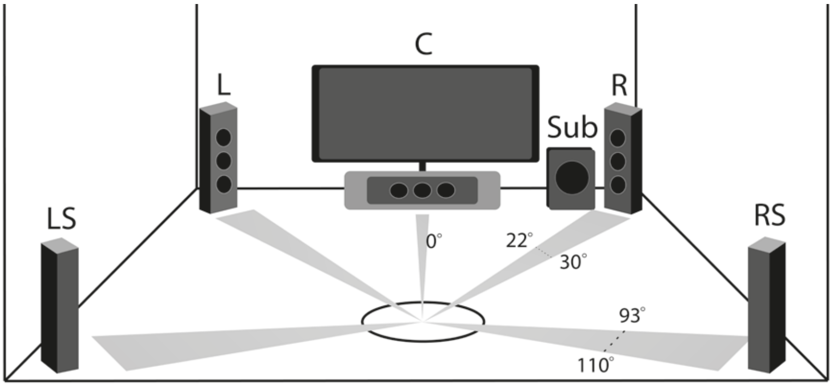

Fig. 61. Loudspeaker configuration in the 5.1 surround sound system.

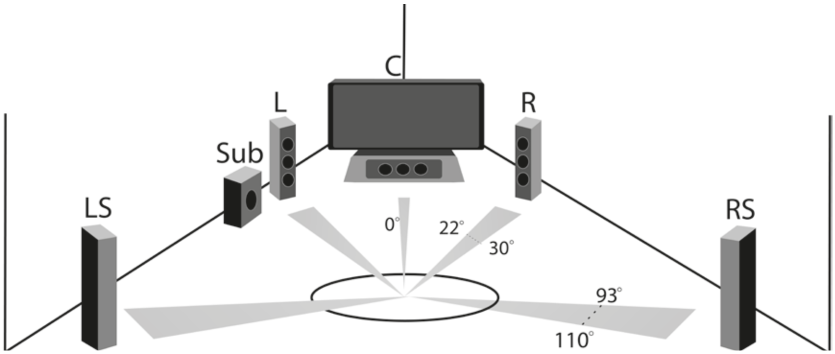

Fig. 62. Loudspeaker configuration in the 5.1 surround sound system in a room corner.

3.1.8. *6.1 System*

The 5.1 system boosted with an extra rear channel. 6.1 feeds audio to six full-range speakers and a single subwoofer. It comprises two front channels (Left and Right), one central channel, two surround channels (Left Surround and Right Surround), one rear bass channel, and one LFE channel[97].

[97] Parekh R. (2006). *Principles of Multimedia*, Tata McGraw-Hill Publishing Company Limited, New Delhi, p. 265; Gookin D. (2010). *PCs for Dummies: Windows® 7 edition*, John Wiley & Sons, Hoboken, p. 164.

3.1.9. *Dolby Digital EX*

An upgrade of the basic Dolby Digital standard. The spatial distribution is identical to the layout of the 6.1 system with an additional central rear channel placed behind the listener. This construction allows for more accurate sound localization within the soundscape when compared to Dolby Digital[98]. Dolby Digital EX can be paired with the 5.1 and 7.1 systems, but it is recommended for 6.1 due to their full compatibility.

3.1.10. *DTS-ES*

An evolution of the traditional DTS technology enriched with an extra LFE channel. *DTS-ES* can be paired with the 5.1 and 7.1 systems. It is recommended for 6.1 due to their full compatibility[99] (fig. 63).

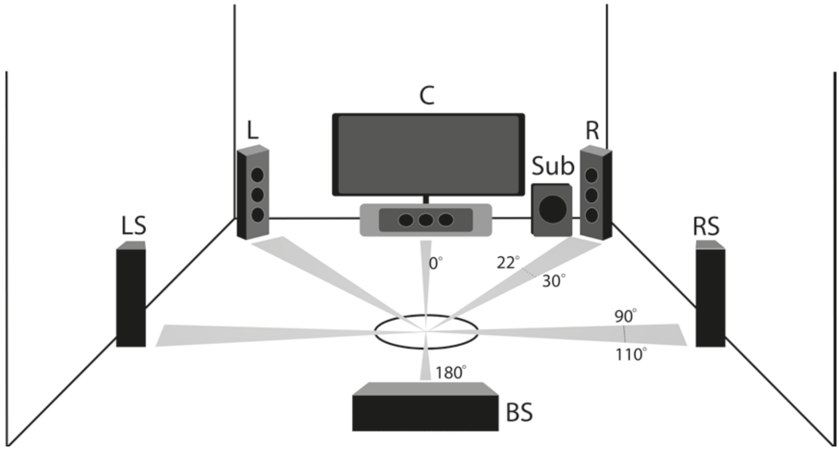

Fig. 63. Loudspeaker configuration in the 6.1 surround sound system.

98 Rushing K. (2006). op. cit., p. 33.

99 Sams Technical Publishing Engineering Staff. (2004). *The Savvy Guide to Home Theater: Savvy advice for designing and purchasing your home theater!*, Indy-Tech Publishing, Indianapolis, p. 38; Briere D., Hurley P. (2009). op. cit., p. 40; Farkas B., Govier J. (2003). op. cit., p. 79; Stapelkamp T. (2007). *DVD-Produktionen: gestalten – erstellen – nutzen*, Springer, Berlin, Heidelberg, New York, p. 533; Rushing K. (2006). op. cit., p. 33.

3.1.11. *7.1 System*

The 6.1 system complete with a new channel. 7.1 provides eight discrete speakers at the minimum (fig. 64 and 65). Every channel can be an independent medium of music data. Surround 7.1 resembles the 5.1 system in its use of the similar cables for connecting speakers with the control unit. However, while 5.1 directs spatial effects and rear speaker effects into two channels located behind the listener, the 7.1 system distributes the surround sound and effects across four channels: two surround channels (Surround Left and Surround Right) and two rear channels for spatialisation (Left and Right). The system is used in advanced devices for surround sound reproduction encountered in cinemas and home theatres[100].

3.1.12. *Dolby Pro Logic IIx / Dolby Pro Logic IIz*

Directly complement the 7.1 system with the option of switching to 6.1. Dolby Pro Logic IIx may be used in home theatres and computer games for a high-quality surround sound of significant depth. The system is used in the cinemas due to its accuracy in the representation of virtual space. Above all, it enriches the listening environment with a new type of surround effect. Dolby Pro Logic IIx generates a smooth, natural sound field for more realistic audio. In turn, Dolby Pro Logic IIz brings an even greater depth of spatial effects with different front speakers[101].

3.1.13. *Dolby Digital Plus*

Offers advanced surround sound. An audio codec designed for use in HD equipment, including Blu-ray. It may be used for surround sound generation in cinemas, computer games, and home theatres. In comparison with Dolby Digital, it stands out with its new coding algorithms for surround sound processing and the coverage of the entire aural bandwidth. Dolby Digital Plus is compatible with the 5.1 and 7.1 systems[102].

100 Gookin D. (2010). op. cit., p. 164; Millerson G., Owens J. (2009). *Television Production*, 14th edition, Focal Press, Burlington, p. 251.
101 See: http://www.dolby.com/us/en/consumer/technology/home-theater/dolby-pro-logic-iix.html [accessed: 26.06.2022].
102 See: http://www.dolby.com/us/en/consumer/technology/home-theater/dolby-digital-plus.html [accessed: 24.06.2022]; Briere D., Hurley P. (2009). op. cit., pp. 41–42.

3.1.14. *Dolby TrueHD*

Employs a lossless HD audio format. Dolby TrueHD is often described as the best approximation of the advanced equipment found in a recording studio. Dolby TrueHD supports the 7.1 system in HD, carrying up to eight full-range 96 kHz / 24-bit audio channels and six full-range 192 kHz / 24-bit channels. Since it can handle up to 16 discrete channels which may be used in the future, the format is still being developed[103].

3.1.15. *DTS-HD*

A surround system allowing for alternative spatial positioning of the speakers. It can deliver sound at sampling rates up to 192 kHz – only in the case in DTS-HD Master Audio – and the resolution of 24 bits for excellent quality studio reproduction, with 16 and 20 bits[104]. Recordings in DTS-HD Audio contain two radically different data streams. They are:

a) the differential (residual) signal which varies from the signal labelled as the core,
b) the differential signal that is different from the main signal.

The differential signal is the component of the original soundtrack created during when the signal is encoded (directly, bit for bit).

At the input, the encoder splits the original audio signal in two. The first component contains the DTS-HD core, and the other, backwardly compatible with the former, is used for comparative purposes[105]. The residual audio signal is encoded losslessly at a different bitrate, and the track is combined with the signal from the DTS core[106]. DTS-HD allows obtaining eight channels with a sampling frequency of 96 kHz and a 24-bit resolution, or a maximum sampling frequency of 192 kHz in the two-channel mode.

[103] See: http://www.dolby.com/us/en/consumer/technology/home-theater/dolby-truehd.html [accessed: 26.06.2022].
[104] Rumsey F., McCormick T. (2006). op. cit., p. 489; Friesecke A. (2007). op. cit., pp. 762–763.
[105] See: http://www.cinematic.pl/wiedza/dts-hd.html [accessed: 25.06.2022]; Briere D., Hurley P. (2009). op. cit., p. 44.
[106] Réveillac J. M. (2018). *Musical Sound Effects. Analog and Digital Sound Processing*, Wiley, Hoboken, pp. 57–58; Briere D., Hurley P. (2009). op. cit., p. 44.

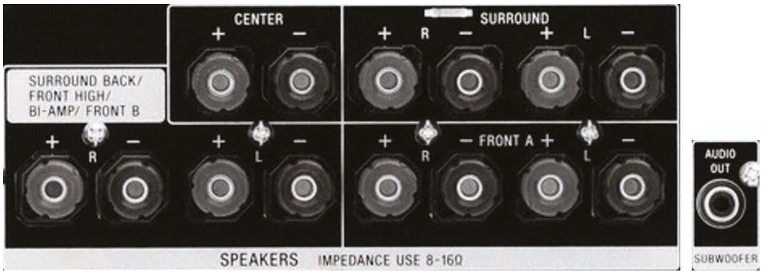

Fig. 64. A device for loudspeaker connection in the 7.1 surround sound system – fragment of the back panel.

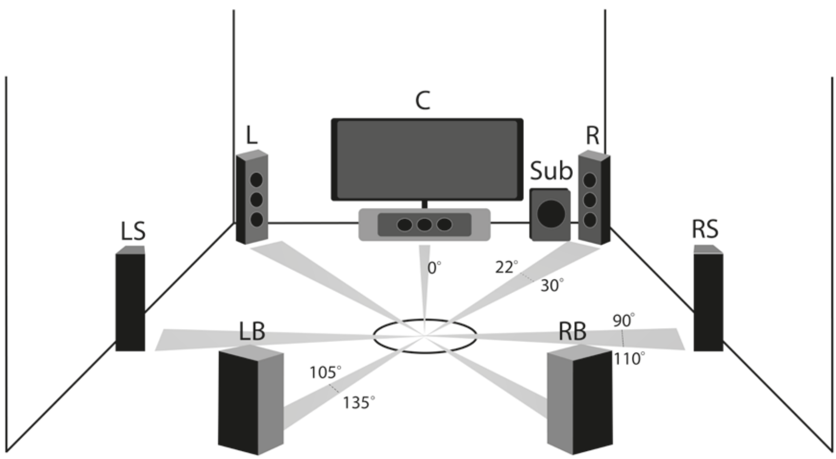

Fig. 65. Loudspeaker configuration in the 7.1 surround sound system.

3.1.16. *10.2 System*

A surround sound system handling twelve discrete speakers (channels) (fig. 66). Seven channels are located at the front, including: Left Wide, Left Height, Left, Centre, Right, Right Height, Right Wide, three surround channels: Left Surround, Right Surround, Back Surround; two LFE channels – LFE Left and LFE Right[107]. Note that the 5.1 system offers the bare minimum of spatialisation. The 10.2 system

107 Kerins M. (2011). op. cit., p. 195; Rumsey F., McCormick T. (2006). op. cit., p. 522.

uses multiple front speakers and only one surround system since the human aural perception is far more precise in the front than in the back.

An additional two front overhead speakers, located at an angle of 45 degrees to the audience (in front of the listener and above their head), add depth to the sound field. The entire arrangement guarantees the accuracy of sound localization, whereas two low-frequency channels enhance the depth[108]. The 10.2 became a departure point for another surround sound system which added yet another two surround speakers – the 12.2. It is worth adding that in contrast to the 5.1 or 7.1 systems, neither 10.2 nor 12.2 is available in a consumer version[109]. In addition, the 10.2 format requires a brand-new design of the main device, capable of distributing sound (spatialisation) among twelve discrete speakers (or fourteen in the case of the 12.2 system). With the advancements in consumer electronics, such systems may be launched in the market in the future.

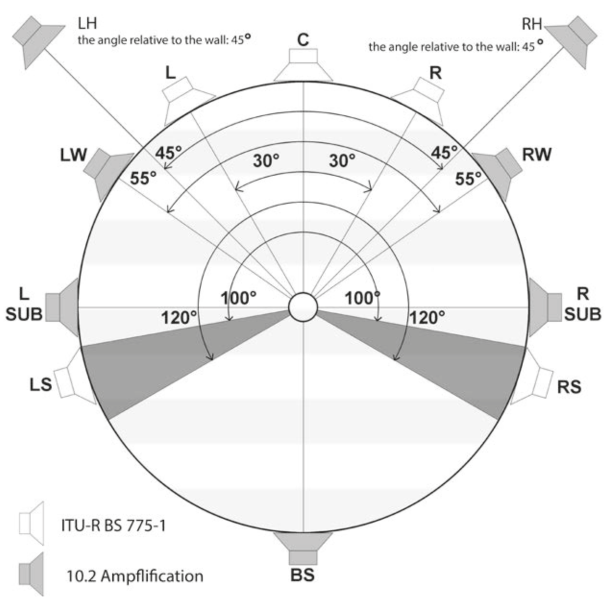

Fig. 66. Loudspeaker configuration in the 10.2 surround sound system.

108 Yang D. T., Kyriakakis C., Kuo C. C. J. (2004). *High-fidelity Multichannel Audio Coding*, Hindawi – New York – Cair, pp. 18–19.

109 Kerins M. (2011). op. cit., pp. 195–196.

3.1.17. *22.2 System*

A surround sound system also referred to as Hamasaki 22.2 from the name of its founder. The 22.2 system was developed for the cinemas and high-definition television in the 4K or 8K resolution. The technique requires the arrangement of speakers in three separate layers. The upper layer consists of nine speakers, the middle layer – ten speakers tilted relative to the upper layer. The bottom layer comprises three speakers, two at the front in room corners, at a distance greater than the cinema screen width, and one in the centre of the screen. The remaining two speakers are LFE channels located to the left and right of the screen (fig. 67 and tables 2–6). To ensure high-quality spatial simulation, the height gap between layers should not exceed 1 m. The 22.2 system is currently the most demanding standard for recording and processing because the audio is transmitted over 24 distinct channels[110].

[110] Nakayama Y., Nishiguchi T., Sugimoto T., Okumura R., Imai A., Iwaki M., Hamasaki K., Ando A., Nishida Y., Mitani K., Kanazawa M., Kitajima S. (2007). "Live Production and Transmission of Large-Scale musical tv program using 22.2 Multichannel Sound with Ultra High Definition Video", in: *IBC 2007 Conference Publication*, http://citeseerx.ist.psu.edu/viewdoc/download?doi=10.1.1.605.7635&rep=rep1&type=pdf [accessed: 24.06.2022], pp. 253–259; Okano F., Kanazawa M., Mitani K., Hamasaki K., Sugawara M., Seino M., Mochimaru A., Doi K. (2004). "Ultrahigh-Definition Television System with 4000 Scanning Lines", in: 2004 NAB BEC Proceedings, http://car.france3.mars.free.fr/Formation%20INA%20HD/HDTV/HDTV%20%202007%20v35/SXRD%20divers2/437_Okano.pdf [accessed: 25.06.2022], p. 440; Hamasaki K., Hiyama K., Nishiguchi T., Ono K. (2004). "Advanced Multichannel Audio Systems with Superior Impression of Presence and Reality", *Audio Engineering Society*, Convention Paper 6053, Presented at the 116th Convention, 2004 May 8–11 Berlin, http://www.aes.org/tmpFiles/elib/20200314/12756.pdf [accessed: 26.06.2022], pp. 3–5; Hamasaki K., Komiyama S., Okubo H., Hiyama K., Hatano W. (2004). "5.1 and 22.2 Multichannel Sound Productions Using an Integrated Surround Sound Panning System", *Audio Engineering Society*, Convention Paper 6226, Presented at the 117th Convention 2004 October 28–31 San Francisco, http://www.aes.org/tmpFiles/elib/20200314/12883.pdf [accessed: 26.06.2022], p. 2; Hamasaki K., Hiyama K., Okumura R. (2005). "The 22.2 Multichannel Sound System and Its Application", *Audio Engineering Society*, Convention Paper 6406, Presented at the 118th Convention 2005 May 28–31 Barcelona, http://www.aes.org/tmpFiles/elib/20200314/13122.pdf [accessed: 24.06.2022], pp. 2–3; Hamasaki K., Nishiguchi T., Hiyama K., Okumura R. (2006). "Effectiveness of height information for reproducing presence and reality in multichannel audio system", *Audio Engineering Society*, Convention Paper 6679, Presented at the 120th Convention 2006 May 20–23 Paris, http://www.aes.org/tmpFiles/elib/20200314/13483.pdf [accessed: 25.06.2022], pp. 4–6; Hamasaki K., Nishiguchi T., Okumura R., Nakayama Y. (2007). "Wide Listening Area with Exceptional Spatial Sound Quality of a 22.2 Multichannel

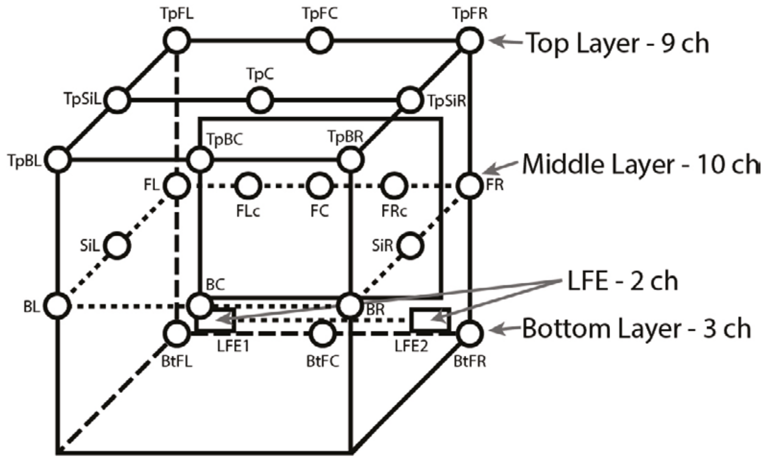

Fig. 67. Loudspeaker configuration in the 22.2 multichannel playback system for cinema use.

Table 2. Labels and spatial positions of speakers in the bottom layer (under the ceiling)[111]

Label	Spatial position of the speaker
TpFL	top (ceiling), front, left room corner
TpFC	top (ceiling), front, centrally above the cinema screen
TpFR	top (ceiling), front, right room corner

Sound System", *Audio Engineering Society*, Audio Engineering Society Convention Paper 7037, Presented at the 122nd Convention 2007 May 5–8 Vienna, http://www.aes.org/tmpFiles/elib/20200314/14022.pdf [accessed: 24.06.2022], pp. 3–5; Nishiguchi T., Nakayama Y., Okumura R., Sugimoto T., Imai A., Iwaki M., Hamasaki K., Ando A., Kitajima S., Otsuka Y., Shimaoka S. (2007). "Production and Live Transmission of 22.2 Multichannel Sound with Ultrahigh-definition TV", *Audio Engineering Society,* Convention Paper 7137, Presented at the 122nd Convention 2007 May 5–8 Vienna, http://www.aes.org/tmpFiles/elib/20200314/14122.pdf [accessed: 24.06.2022], pp. 9–10; Hamasaki K. (2011). "The 22.2 Multichannel Sounds and Its Reproduction at Home and Personal Environment, Audio Engineering Society", AES 43rd International Conference, Pohang, 2011 Sep. 29–Oct. 1, http://www.aes.org/tmpFiles/elib/20200314/16130.pdf [accessed: 26.06.2022], pp. 2–4; Millerson G., Owens J. (2009). op. cit., pp. 241–242.

111 Millerson G., Owens J. (2009). op. cit., p. 242.

Label	Spatial position of the speaker
TpSiL	top (ceiling), in the middle of the left wall
TpC	top (ceiling) at the intersection of room diagonals
TpSiR	top (ceiling), in the middle of the right wall
TpBL	top (ceiling), rear, left room corner
TpBC	top (ceiling), rear, in the middle of the back wall
TpBR	top (ceiling), rear, right room corner

Table 3. Labels and spatial positions of speakers in the middle layer (over the listener's head but in the lower half of the room)[112]

Label	Spatial position of the speaker
FL	front, left room corner
FLc	front, to the left behind the screen, in the middle between FL and FC
FC	front, centrally behind the cinema screen, at the intersection of screen diagonals
FRc	front, to the right behind the screen, in the middle between FC and FR
FR	front, right room corner
SiL	left, in the middle of the left wall
SiR	right, in the middle of the right wall
BL	rear, left room corner
BC	rear, in the middle of the back wall
BR	rear, right room corner

[112] Ibidem.

Table 4. Labels and spatial positions of speakers in the bottom layer (floor)[113]

Label	Spatial position of the speaker
BtFL	bottom (floor), front, left room corner
BtFC	bottom (floor), front, centrally in front of the cinema screen
BtFR	bottom (floor), front, right room corner

Table 5. Labels and spatial positions of LFE channels (floor)[114]

Label	Spatial position of the LFE channel
LFE1	bottom (floor), front, left side of the room, next to BtFL
LFE2	bottom (floor), front, right side of the room, next to BtFR

Table 6. The delay of individual speakers in the 22.2 system[115]

Channel	Delay	Channel	Delay	Channel	Delay	Channel	Delay
1–2	0	1–3	–35.2	1–4	–27.3	1–5	–27.3
1–6	–35.2	1–7	–43.0	1–8	–31.3	1–9	–54.7
1–10	–54.7	1–11	–43.0	1–12	–46.9	1–13	–15.6
1–14	–7.8	1–15	–7.8	1–16	–11.7	1–17	–39.1
1–18	–27.3	1–19	–27.3	1–20	–27.3	1–21	–27.3
1–22	–23.4	1–23	–27.3	1–24	31.3		

113 Ibidem.
114 Ibidem.
115 Nakayama Y., Nishiguchi T., Sugimoto T., Okumura R., Imai A., Iwaki M., Hamasaki K., Ando A., Nishida Y., Mitani K., Kanazawa M., Kitajima S. (2007). op. cit., p. 258.

3.1.18. *Periphonic sound system*

A method for encoding and playing multi-dimensional sound[116]. This technology employs some solutions of ambisonics[117], although it operates in a different manner and enables the listener to receive completely different perceptual impressions (pictures) by applying multi-channel sets of equivalent loudspeakers[118].

It is noteworthy that this system is not used by sound engineers because the processed sound picture, played in this system, is not consistent with the real creation of a spatial field with electroacoustic equipment. This is about distorting a natural acoustic field, lack of realism, which manifests itself as spatial aliasing, understood as improper identification of a sound signal. Distortions caused by incorrect production methods manifest themselves as phase changes which, in music recordings, are errors from the production perspective[119].

Owing to ambisonics, it is possible to precisely grasp, record and play a natural music event[120], whereas periphonics goes much further beyond the boundaries of ambisonics, i.e. mathematics, physics and naturalness of sound stimuli. Spherical harmonic sound field decomposition is regarded as a specific type of sound material in the work of composers or creators of spatial sound installations, called sound designers. Two distinct approaches should be identified, as it is a supreme goal to obtain periphonic sound because it enables one to build a space around a spectator – on each side – owing to which, ultimately, a listener can be surrounded by sound. This technology is used by composers of electronic and electroacoustic music. It is the object of various studies in scientific

[116] Scaini D., Arteaga D. (2014). "Decoding of Higher Order Ambisonics to Irregular Periphonic Loudspeaker Arrays", in: *AES 55th International Conference*, August 27–29, Helsinki, pp. 1–8.

[117] Zapała R. (2012). *Obszary i struktury materiału muzycznego w kompozycji "Skaner" na orkiestrę symfoniczną, elektronikę i soundscape*, unpublished doctoral dissertation written under the scientific supervision of L. Zielińska, Wydział Twórczości, Interpretacji i Edukacji Muzycznej, Akademia Muzyczna w Krakowie, Kraków, p. 36.

[118] Cf. Hollerweger F. (2006). *Periphonic sound spatalization in multi-user virtual environments*, unpublished master's thesis written under the scientific supervision of R. Hööldrich, Center for Research in Electronic Art Technology, Institute of Electronic Music and Acoustics, Graz.

[119] Anderson J. (2010). "Technika ambisonics. Bardzo krótkie wprowadzenie do innowacyjnej teorii", *Glissando*, 16, p. 50.

[120] Ibidem, p. 49.

centres around the world, but periphonics is an experimental system associated with the projection of spatial sound.

The periphonic system allows for obtaining sound in two horizontal axes: right-left, back-front, and a vertical one: up-down[121] – which is responsible for creating 3D sound. Its structure is based on applying various numbers of loudspeakers (there is no uniform standard), e.g. on two, three or more layers (heights). Each layer allows for placing loudspeakers on the circumference of a circle or perimeters of various regular polygons (loudspeakers are placed at vertices of polygons or in the middle of the sections – between the vertices). The number of loudspeakers can be suited to the acoustic conditions of the room in which the sound material is to be played.

The sound material is produced with at least four condenser microphones set up in appropriate directions toward the source of the sound being recorded or with specialised microphones, whose construction and characteristics are presented in subchapter 1.3.2. There are also other methods for producing periphonic sound using many more microphones, which improves the perceptual experience when listening to such sound. Due to the fact that there is no uniform standard, periphonics cannot be described and categorised precisely because its characteristics can include: a different number of loudspeaker layers, a different number of loudspeakers used for playing the sound and microphones for sound recording, diverse use of microphones, diverse microphone characteristics and various spatial figures based on which the loudspeakers are set up, etc.

[121] Trębacz E. (2010). “Pełna immersja: rzeczywistość czy utopia? Technologie przestrzenne z punktu widzenia artysty”, *Glissando*, 16, p. 45.

Chapter 4. Connectors and connections

This subchapter discusses connectors used for analog and digital signal transmission in audio devices. Since such devices and connections can be encountered on a stage and in a recording studio, it is important to give an overview of the topic.

Audio interference created during sound transmission over electrical cables is commonly reduced with the use of balanced sockets (inputs and outputs) coupled with the plugs and balanced wiring. However, when working with electroacoustic devices, electrical interference cannot be eliminated entirely. The problem grows more conspicuous with increasing distance when multiple cables located in close proximity transmit different electrical signals.

Unbalanced connections require a cable with two conductors: one is connected to the ground, while the other carries the audio signal (fig. 68 a) and b)).

Balanced connections necessitate a cable containing three conductors, two of which carry the same acoustic signal, sent in between them in reverse polarity. The third conductor, usually grounded, is a shield which protects the other two conductors from interference (fig. 69 a) and b)).

The audio signal is transferred with its polarity reversed. In this way, the phenomenon of concurrent signal attenuation can be exploited for cancelling out unwanted sound artefacts, should any interference arise on either conductor. As the interference cancels itself out, the correct signal is passed through. This effect is achieved with the right input system – present in a mixing console, for instance – for proper attenuation and isolation of unwanted anomalies[122].

[122] Butler T. (1994). op. cit., p. 147.

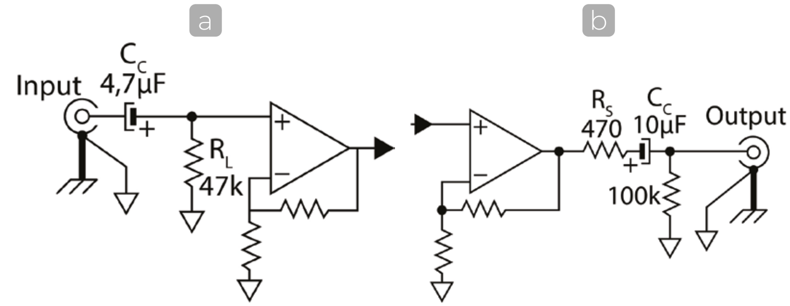

Fig. 68. a) and b) Example of unbalanced
better audiblyright.

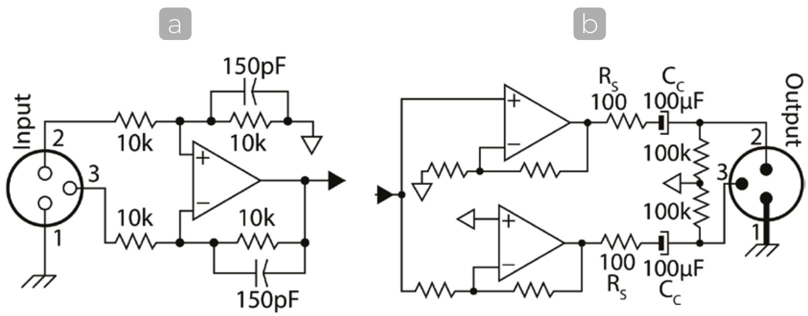

Fig. 69. a) and b) Balanced XLR input (to the left)
and XLR output (to the right).

4.1. The DI box (Direct Injection Box)

Is an electroacoustic device intended for balancing the audio signal (fig. 70 a) and b)). Professional studio and stage setups make use of only balanced connections to minimize various types of disturbances caused by the electrical network imperfections or the construction of a given musical instrument. Some instruments, such as the electric guitar, electroacoustic guitar, acoustic guitar with an electroacoustic transducer, bass, digital piano or synthesizer, have unbalanced outputs. These outputs could lead to capturing unwanted disturbance because of:

- long distances from the sound engineer (in stage applications),
- a large number of many electrical receivers connected to the electrical network (in studio applications),

- other unforeseen circumstances.
  In a nutshell, a DI box balances the audio signal to preserve its technical quality irrespective of the non-musical factors[123].

To balance the signal, a DI box sends it in two versions – regular and phase-inverted – to a mixing console or a recording device. Therefore, the connection requires a separate cable for the transfer of the additional, phase-inverted signal. Both signals reach the receiver (for instance, a mixing console) in a single cable, and the reversed signal is inverted back. Once in phase, the signals are summed. Both (!) contain disturbances. The primary signal is amplified by the other – in phase thanks to the double inversion. Meanwhile, the antiphase interference cancels itself out (goes silent) and is eliminated[124].

There are two types of DI boxes – active and passive. For proper operation, active devices must be powered by a battery placed inside the unit or by phantom power from the mixing console (or another generator). While not all active DI boxes may have the battery slot, all will operate on phantom power. Since passive DI boxes have no power supply, the name "passive", indicates no driving current flowing through. When comparing both types of DI boxes, the active devices must be unequivocally shown as superior to their passive counterparts, as they offer far better filtering and protection from extraneous, unwanted disturbances thanks to the power supply. As the active units can eliminate the artefacts produced in the audio signal far more effectively than passive ones. Contrarily, passive DI boxes coupled with long cables are more susceptible to disturbance due to the lack of a power supply[125].

DI boxes usually come with two switches, one of which is labelled as GND (ground). This comes in handy if you encounter any problems with the electrical network or the power supply of digital

[123] Slone J. J. (2002). *The Basics of Live Sound: Tips, Techniques & Lucky Guesses*, Hal Leonard, Milwaukee, p. 31; Middleton P., Gurevitz S. (2008). *Music Technology Workbook: Key Concepts and Practical Projects*, Focal Press, Burlington, pp. 293–294; Bartlett B., Bartlett J. (2009). *Practical Recording Techniques: The Step-by-step Approach to Professional Audio Recording*, 5th edition, Focal Press, Burlington, pp. 125–126; Rumsey F., McCormick T. (2009). op. cit., pp. 371–373; Hurtig B. (1988). *Multi-Track Recording for Musicians*, Alfred Publishing Co., Van Nuys, p. 66.

[124] Slone J. J. (2002). op. cit., p. 31; Middleton P., Gurevitz S. (2008). op. cit., pp. 293–294; Bartlett B., Bartlett J. (2009). op. cit., pp. 125–126; Rumsey F., McCormick T. (2009). op. cit., pp. 371–373; Hurtig B. (1988). op. cit., p. 66.

[125] Rumsey F., McCormick T. (2009). op. cit., pp. 371–373.

instruments (switching power supplies). It serves to create additional grounding and mitigate the disturbances. The other switch, labelled PAD, is used to dampen the signal, which may be excessively loud (distortion) when transmitted to the mixing console from the DI box. The switch attenuates the signal by a specific volume, which can differ for every model, so the exact number of decibels will always be provided next to the "PAD" label. Note that some devices have additional buttons, such as the repeat phase inversion, two dampeners of different volumes, or the fusion of two signals into one (only in stereo DI boxes).

a b

Fig. 70. a) DI box front panel and b) DI box rear panel.

4.2. XLR (commonly referred to as Cannon – from the name of James H. Cannon)

A balanced latch locks a connector with three pins (fig. 71 a) – b), 72), used for plugging in microphones or as an AES / EBU connector (but with a cable other than the microphone cable). XLRs can transmit signals over vast distances (a few dozen meters) with minimal losses. Balanced connections have a two-conductor cable surrounded with a single shield. The voltage is equivalent on both conductors, but with opposite signs (hence the signals are perfectly reversed). Note that both conductors are located in close proximity. The small distance may cause interference, which is eliminated in the same manner. This results in excellent resistance to interference, which accounts for the use of XLRs in a wide array of (studio and stage) installations[126].

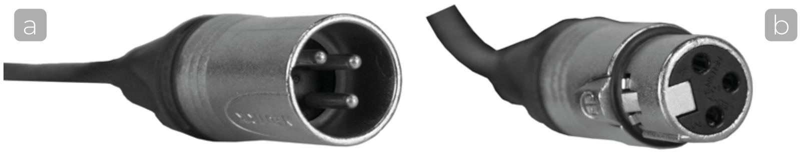

Fig. 71. a) and b) Male (to the left) and female (to the right) XLR connectors mounted on cables.

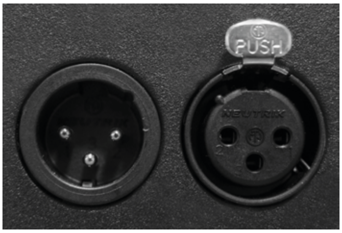

Fig. 72. Male (to the left) and female (to the right) XLR sockets mounted in an electronic device.

4.3. TS / TRS (colloquially known as the big Jack – 6.3mm in diameter, micro Jack – 3.5mm in diameter, and micro Jack – 2.5mm)

The TS are unbalanced connectors for monophonic signal transmission (fig. 73 a)). Due to its susceptibility to interference, a TS cable should not be very long. TS connectors are used for instrumental cables and studio equipment. In such applications, a single main

[126] Butler T. (1994). op. cit., pp. 151–153; Bailey A. (2001). *Network technology for digital audio*, Focal Press, Oxford, pp. 124–125; Alexander P. L., Sheridan J. (2001). *How to Do a Demo Quality Recording in Your Bedroom*, 2nd edition, Hal Leonard, Milwaukee, p. 103; Eargle J. (2003). *Handbook of Recording Engineering*, Klauwer Academic Publishers, Boston – Dordrecht – London, p. 93; Bartlett B., Bartlett J. (2014). op. cit., pp. 6–7.

cable with a shield makes the signal highly prone to interference. Any noise that penetrates the shield will be regarded as a signal, reinforced by the power amplifier, and fed to the speakers. Therefore, to reduce the risk of interference, only high quality cables should be used. The TRS connectors (fig. 73 b) and c)) can carry unbalanced signals (susceptible to interference) and balanced signals (resistant to interference)[127].

Fig. 73. a) – c) TS 6.3 mm connector (monophonic) – to the left, TRS 6.3 mm connector (stereophonic) – in the centre, and TRS 3.5 mm connector (stereophonic) – to the right.

Apart from the types discussed above, one may also encounter the Combo connector which is a combination of XLR and TS (fig. 74). A Combo socket is designed to accommodate a cable with either XLR or TS/TRS pins, but not both types at the same time.

Fig. 74. The Combo socket for the connection of an XLR or TS / TRS 6.3 mm plug.

[127] Rumsey F., McCormick T. (2009). op. cit., pp. 356–357, 362; Butler T. (1994). op. cit., pp. 151–153; Alexander P. L., Sheridan J. (2001). op. cit., pp. 101–102; Trubitt R. (2002). *Mackie Compact Mixers*, 2.1 edition, Hal Leonard, Milwaukee, pp. 165–166.

4.4. RCA (Radio Corporation of America; colloquially known as the cinch)

The connector carries an unbalanced signal (fig. 75 a) and b)). Often used as recorder input and output in mixing consoles. When installed on a coaxial cable, it can be utilized to transfer the digital S / PDIF signal[128].

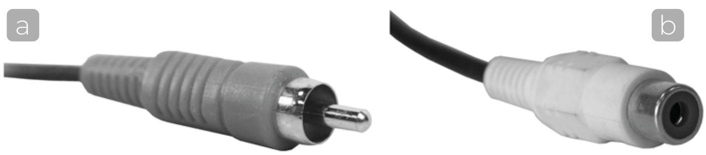

Fig. 75. a) and b) Male (to the left) and female (to the right) RCA connectors mounted on cables, often referred to as chinches.

4.5. Speakon

A connector designed chiefly with speaker cables in mind. Its solid construction and quality workmanship make it suitable for high-voltage devices (fig. 76 a) and b)). Speakons come with a safeguard that prevents accidental disconnection of the cable[129].

Fig. 76. a) A plug mounted on a cable and a socket installed in a Speakon electronic device, b) Speakon connector mounted on cables.

[128] Kefauver A. P., Patschke D. (2007). *Fundamentals of digital audio*, A-R Editions, Middleton, p. 170; Bailey A. (2001). op. cit., pp. 83–85; Leonard J. A. (2001). *Theatre Sound*, Routledge, New York, London, p. 29; Thompson D. M., (2005). op. cit., p. 163.

[129] Boyce T. (2014). *Introduction to Live Sound Reinforcement: The Science, the Art, and the Practice*, Friesen Press, Victoria, pp. 114–115.

4.6. Toslink (Toshiba LINK)

A connector used for carrying a digital audio stream with a fibre-optic cable, in compliance with the ADAT or S / PDIF protocols (fig. 77 a) – d)). The audio is sent in digital format, as red light with a wavelength of 660 nm[130].

Fig. 77. a) – d) Toslink fibre-optic connector (top-left corner) with sockets closed with protective anti-dust caps (centre of the upper row), sockets open with the caps off (top-right corner), and sockets with an automatic anti-dust block (centre of the lower row).

4.7. AES / EBU (Audio Engineering Society / European Broadcasting Union)

The audio is sent as an ordinary balanced signal (XLR connector) (fig. 78). However, a special cable allows for digital audio transfer at a sampling rate of 48 kHz[131].

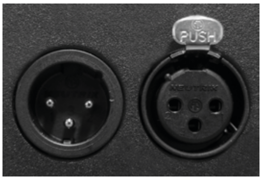

Fig. 78. Digital AES / EBU sockets (identical to XLR on the outside).

130 Kefauver A. P., Patschke D. (2007). op. cit., pp. 170–174; Derry R. (2003). *PC Audio Editing: Broadcast, Desktop, and CD Audio Production*, Focal Press, Burlington, p. 40.

131 Ballou G. (2008). *Handbook for Sound Engineers*, 4th edition, Focal Press, Burlington, p. 189; Touzeau J. (2009). *Home Studio Essentials*, Course Technology Cengage Learning, Boston, p. 135; Huber D. M., Runstein R. E. (2010). op. cit., p. 215.

4.8. S / PDIF (Sony / Philips Digital Interface Format)

Connectors used for signal transmission in this format are RCA connections (fig. 79 a) and b)) with a 75 Ω coaxial cable. S / PDIF allows for using up to 15m of cable. The connector of choice for advanced applications is the Toslink[132].

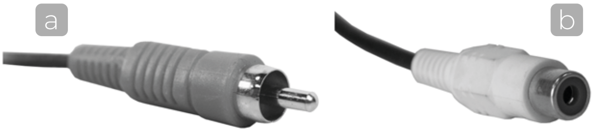

Fig. 79. a) and b) Male and female digital S / PDIF connectors mounted on cables (identical to RCA on the outside).

4.9. ADAT (Alesis Digital Audio Tape)

The name was coined from a digital recorder manufactured by Alesis, which recorded data (sound) on eight discrete channels using VHS tapes[133]. Currently, the ADAT Lightpipe uses the (fibre-optic) Toslink connector (fig. 80 a) – d)). ADAT is highly appreciated among sound engineers as it makes the transmitted signal impervious to electromagnetic interference. ADAT serves to connect devices and offers unidirectional transfer over (depending on the setup[134]

a) eight audio channels at 24-bit quality and a sampling frequency of 48 kHz – standard option
b) four audio channels at 24-bit quality and a sampling frequency of 96 kHz,

[132] Kefauver A. P., Patschke D. (2007). op. cit., pp. 122–123; Touzeau J. (2009). op. cit., pp. 135–136; Huber D. M., Runstein R. E. (2010). op. cit., p. 216; Kilts S. (2007). *Advanced FPGA Design: Architecture, Implementation, and Optimization*, John Wiley & Sons, Hoboken, p. 107.

[133] Parekh R. (2006). *Principles of Multimedia*, Tata McGraw-Hill Publishing Company Limited, New Delhi, p. 232.

[134] Bailey A. (2001). op. cit., p. 174; Touzeau J. (2009). op. cit., pp. 136–137; Huber D. M., Runstein R. E. (2010). op. cit., p. 218

c) two audio channels at 24-bit quality and a sampling frequency of 192 kHz[135].

Fig. 80. a) – d) ADAT plug and sockets (identical to Toslink on the outside).

4.10. MADI (Multichannel Digital Interface)

A connection technology which is based on serial transmission over a coaxial cable (RCA) or a fibre-optic cable (Toslink) (fig. 82 a) and b), 81 a) and b)). It can carry 28, 56, or 64 channels concurrently, depending on the setup[136]. The sampling frequency depends on the number of channels and amounts to 64 up to 96 kHz (48 kHz for 64 channel)[137].

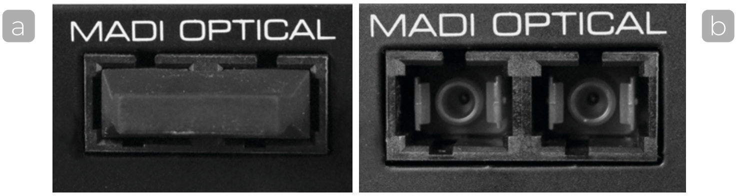

Fig. 81. a) and b) Optical sockets MADI protected from dust with a dedicated rubber cap (left) and optical sockets MADI with the caps off (right) to enable a fibre-optic cable connection.

135 Strong J. (2012). *Home recording for musicians for dummies*, 4th edition, Wiley, Indianapolis, p. 56; Huber D. M., Runstein R. E. (2010). op. cit., p. 218.

136 Kefauver A. P., Patschke D. (2007). op. cit., p. 169.

137 Bailey A. (2001). op. cit., pp. 172–173; Self D. (2009). op. cit., pp. 575–578; Touzeau J. (2009). op. cit., pp. 137–138; Huber D. M., Runstein R. E. (2010). op. cit., pp. 217–218; Morgan C., Hoffner R. (2007). "Digital Audio Standards and Practices", in: Williams E. A., Jones G. A., Layer D. H., Osenkowsky T. G. (eds.). *National Association of Broadcasters Engineering Handbook*, 10th edition, Focal Press, Burlington, p. 213.

Fig. 82. a) and b) Coaxial MADI sockets on the cable (to the left) and installed in an electronic device (to the right).

4.11. Word Clock

Word Clock is a special signal (not audio/MIDI) for the synchronization of multiple digital devices connected in a digital network. (fig. 83) The name refers to the fact that one cycle of the Word Clock corresponds to the duration of a single word – one audio sample. The source of the Word Clock signal is a master device (one per system, selected by the user). Its receivers are slave devices that can adjust their timing signal generators to the Word Clock signal transmitted from the master device. The Word Clock signal is transmitted over a 75 Ω coaxial cable tipped with a BNC connector. The higher frequency of the clock ensures greater accuracy of its performance[138].

Word Clock should not be confused with the SMPTE (Society of Motion Picture and Television Engineers) timecode. Word Clock serves exclusively for the generation of one time and a constant bitrate to avoid errors in digitally connected devices. The SMPTE timecode involves actual data (metadata) on the transmitted files. SMPTE is optional and sent differently than Word Clock. It is used in movie productions and show control applications, which

[138] Huber D. M., Runstein R. E. (2010). op. cit, pp. 222–223; Derry R. (2006). *PC Audio Editing with Adobe Audition 2.0: Broadcast, Desktop and CD Audio Production*, Focal Press, Burlington, p. 225; Franz D. (2003). *Producing in the Home Studio with Pro Tools*, 2nd edition, Berklee Press, Boston, p. 135; Gibson B. (1999). op. cit., pp. 189–191; Shepherd A. (2003). op. cit., pp. 60–62; Campbell R. J. (2013). *Pro Tools 10 Advanced Music Production Techniques*, Course Technology Cengage Learning, Boston, pp. 205–216.

require time coordination of various elements constituting the visual production[139].

The synchronization signal may also be transmitted with other communication protocols, which use different types of connectors and connections, such as S/PDIF, AES/EBU, ADAT, Ethernet[140]. The selection of a communication protocol depends on the user's equipment (not all devices come with the required sockets) and the type of signal for synchronization. Word Clock is chosen for the synchronization of audio and image, whereas the other protocols serve to synchronize audio-only signals.

Fig. 83. A word clock connector on a panel of an electroacoustic device.

[139] Shepherd A. (2003). op. cit., p. 62; Sethi A. (2005). *Multimedia Education: Theory And Practice*, International Scientific Publishing Academy, New Delhi, p. 62; Casabona H., Frederick D. (1988). *Advanced MIDI Applications. Computers,* Time Codes, *and Beyond. The Keyboard magazine library for Electronic Musician*, Alfred Publishing Co., Van Nuys/GPI Publications, Cupertino, pp. 18, 22–25.

[140] Shepherd A. (2003). op. cit., p. 60; Roback S. (2004). *Pro Tools 6 for Macintosh and Windows*, Peachpit Press, Berkeley, pp. 440–441.

4.12. Connections based on the Internet Protocol (fig. 84 a) and b))

Table 7. Basic technical information for audio connections based on Internet Protocol

Network type (system name)	Solution developer	Network structure	Standard	Data protocol	Layers
A-Net Pro16 / Pro64	Aviom	Independent	Reserved	Reserved	1
Audia	Biamp	CobraNet	Reserved	Reserved	2
AudioRail	AudioRail	Independent	Reserved	Reserved	1
Barix	Barix	Independent	Reserved	Reserved	No data
BASIS	QSC	CobraNet	Reserved	Reserved	2
CobraNet	PeakAudio	Independent	Reserved	Reserved	2
D.A.I.S.	Audio-Service	Independent	Reserved	Reserved	No data
Dante	Audinate	Independent	Open, Real-time Transport Protocol	Real-time Transport Protocol / User Datagram Protocol	3 (Internet Protocol / User Datagram Protocol / Real-time Transport Protocol)
Ethersound	Digigram	Independent	Reserved	Reserved	2

Network type (system name)	Solution developer	Network structure	Standard	Data protocol	Layers
iDR-Series	Allen&Heath	Independent	Reserved	Reserved	1
Livewire	Axia	Independent	Open, Real-time Transport Protocol	Real-time Transport Protocol	3 (Internet Protocol / User Datagram Protocol / Real-time Transport Protocol)
Media-Matrix	Peavey	CobraNet	Reserved	Reserved	2
MI 2000	Audiac	Independent	Reserved	Reserved	No data
NetCira	Fostex	Ethersound	Reserved	Reserved	2
Nexus	Salzbrenner Stagetech	Independent	Reserved	Reserved	Bus system
Nova Net	Lawo	Independent	Reserved	Reserved	Bus system
RAVE	QSC	CobraNet	Reserved	Reserved	2
REAC	Roland	Independent	Reserved	Reserved	2
Studer	Studer	Independent	Reserved	Reserved	No data
SymNet	Symetrix	CobraNet	Reserved	Reserved	2
Vadis	Klotz Digital	Independent	Reserved	Reserved	Bus system
RockNet 100	Riedel	Independent	Reserved	Reserved	1

Network type (system name)	Solution developer	Network structure	Standard	Data protocol	Layers
RockNet 300	Riedel	Independ-ent	Reserved	Reserved	1
AVB	IEEE	Independ-ent	Open, Real-time Transport Protocol	IEEE 1722	2
Ravenna	ALC Networx	Independ-ent	Open	Real-time Trans-port Proto-col / User Datagram Protocol	3
Hyper-MAC	Sony Pro-Audio Lab	Independ-ent	No data	No data	No data

Table 8. Basic tehnical information for audio connections based on Internet Protocol

Network type (system name)	Transfer	Compatibility with the Ethernet network and switch	Maximum number of channels at the transfer rate of 100 Mbit/s	Latency (ms) at the transfer rate of 100 Mbit/s	Audio format (Bit / kHz)
A-Net Pro16 / Pro64	Cat-5	No	64	326	24 / 48

Network type (system name)	Transfer	Compatibility with the Ethernet network and switch	Maximum numbel of channels at the transfer rate of 100 Mbit/s	Latency (ms) at the transfer rate of 100 Mbit/s	Audio format (Bit / kHz)
Audia	Cat-5 / Optical fibre	Yes	No data	No data	No data
AudioRail	Optical fibre	No	32	5	24 / 96
Barix	No data	No data	No data	No data	No data
BASIS	Cat-5 / Optical fibre	Yes	No data	No data	No data
CobraNet	Cat-5 / Optical fibre	Yes	64	1330 / 2660 / 5330	24 / 96
D.A.I.S.	Optical fibre	No data	No data	No data	No data
Dante	Cat / Optical fibre	Yes	48	150 / 500 / 1000 / 5000	32 / 96
Ether-sound	Cat-5 / Optical fibre	Partial	64	125	24 / 192
iDR-Se-ries	Cat-5	No data	No data		24 / 48
Livewire	Cat / Optical fibre	No data	43 (26 in the Livestream mode)	< 1000	No data

Network type (system name)	Transfer	Compatibility with the Ethernet network and switch	Maximum numbel of channels at the transfer rate of 100 Mbit/s	Latency (ms) at the transfer rate of 100 Mbit/s	Audio format (Bit / kHz)
Media-Matrix	Cat-5 / Optical fibre	Yes	No data	No data	No data
MI 2000	Optical fibre	No data	No data	No data	No data
NetCira	Cat-5 / Optical fibre	Partial	64	No data	No data
Nexus	No data	No data	No data	No data	No data
Nova Net	No data	No data	No data	No data	No data
RAVE	Cat-5 / Optical fibre	No data	No data	No data	No data
REAC	Cat-5	Yes	40	375	24 / 96
Studer	No data	No data	No data	No data	No data
SymNet	Cat-5 / Optical fibre	Yes	No data	No data	No data
Vadis	No data	No data	No data	No data	No data
RockNet 100	Cat / Optical fibre	No	80	400	32 / 96

Network type (system name)	Transfer	Compatibility with the Ethernet network and switch	Maximum number of channels at the transfer rate of 100 Mbit/s	Latency (ms) at the transfer rate of 100 Mbit/s	Audio format (Bit / kHz)
RockNet 300	Cat / Optical fibre	No	160	400	32 / 96
AVB	Cat / Optical fibre	AVB Switches	No data	250	Consistent with IEC 61883
Ravenna	Cat / Optical fibre	Yes	No data	< 1000	Variable depending on the frequency
Hyper-MAC	Cat-5e / Optical fibre	No data	No data 384 – 200 Mbit/s	62.5 – 100 Mbit/s 41.66 – 200 Mbit/s	24 / 192 kHz

Table 9. Basic technical information for audio connections based on Internet Protocol

Network type (system name)	Nominal rate in Mbit/s	Network topology	Compatibility with other network traffic	Redundancy
A-Net Pro16 / Pro64	100	Peer to peer / Star	No	No
Audia	100	Any	Yes (VLAN)	Spanning Tree Protocol / Ring
AudioRail	100	No data	No data	No data
Barix	No data	No data	No data	No data
BASIS	100	Any	Yes (VLAN)	Spanning Tree Protocol / Ring
CobraNet	100	No data	Yes (VLAN)	Spanning Tree Protocol / Ring
D.A.I.S.	No data	No data	No data	No data
Dante	100 / 1000	Any	Yes	Spanning Tree Protocol / Ring
Ethersound	100 / 1000	Daisy-Chain	No	Spanning Tree Protocol / Ring
iDR-Series	No data	No data	No data	No data
Livewire	100 / 1000 / 10G	Any	Yes	No data
MediaMatrix	100	Any	Yes (VLAN)	Spanning Tree Protocol / Ring

Network type (system name)	Nominal rate in Mbit/s	Network topology	Compatibility with other network traffic	Redundancy
MI 2000	No data	No data	No data	No data
NetCira	100 / 1000	Daisy-Chain	No	Spanning Tree Protocol / Ring
Nexus	No data	No data	No data	No data
Nova Net	No data	No data	No data	No data
RAVE	100	Any	Yes (VLAN)	Spanning Tree Protocol / Ring
REAC	100	No data	No data	No data
Studer	No data	No data	No data	No data
SymNet	100	Any	Yes (VLAN)	Spanning Tree Protocol / Ring
Vadis	No data	No data	No data	No data
RockNet 100	400	Ring	Yes	Ring
RockNet 300	400	Ring	Yes	Ring
AVB	100 / 1000 / 10G	Any	Yes	Spanning Tree Protocol
Ravenna	100 / 1000 / 10G	Any	Yes	Spanning Tree Protocol / Ring
HyperMAC	100 / 1000 / 10G	Any	No data	No data

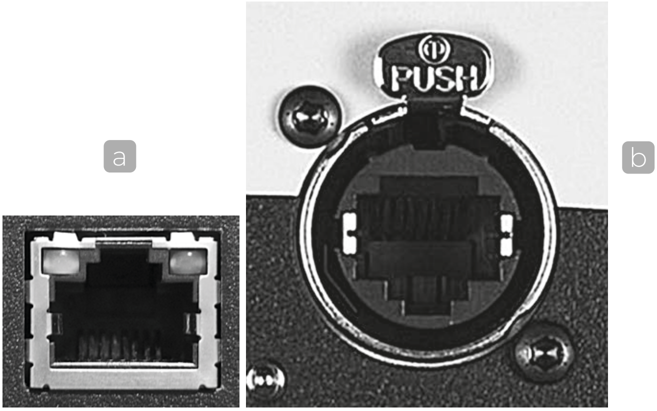

Fig. 84. a) and b) An EtherSound socket installed in an electronic device.

Chapter 5. Production methods for stereo recordings

5.1. Miking techniques

Sound engineers and music producers often rack their brains over the choice of technical solutions for recoding audio materials to achieve the best possible sound. To compound their predicament, recordings are sometimes made in a non-dedicated room, such as a lecture hall or a church, rather than a professional studio.

This chapter presents the most common microphone techniques. The use of various methods allows recording the same material in a multitude of ways, accentuating different acoustic characteristics of the sound source and the room.

The microphones and the method are selected before the recording starts. All consequences of employing the selected techniques must be predicted in advance, considering the acoustic conditions of the room. Before commencing work, you need to create a plan which takes into account multiple factors such as:

a) quality of the power supply if the recording is not held in a professionally adapted room (the risk of electrical interference related to the operation of the lifts, air-conditioning, or mechanical devices),
b) selection of the right equipment for both the performer and the room,

c) level of noise in the room (for instance, the proximity of busy streets),
d) the type of music that will be registered,
e) acoustic conditions during the rehearsal preceding the recording,
f) the amount of equipment necessary (depending on the number of band members),
g) movement onstage, transformations in the arrangement of the musicians, the choir, etc.,
h) acoustics (including the timbre) of the recording space (a concert hall, a church, a school gym, a soundproofed room).

If the recording is made in a room with interesting acoustics, microphones may be placed further away from the source to record effects such as the unique reverberation. However, in the event of any interfering factors such as a busy street nearby, the focus should fall on recording direct sound (with microphones situated near the sound source) despite excellent ambience. The reverb effect should be added later with a digital device.

5.1.1. *Mono recordings*

It is common knowledge that monophonic recordings are the bedrock of modern sound engineering (fig. 85). The use of mono devices in music productions has a long history as they allowed for sound recording and archiving on various media. Mono recording may be made with one microphone, a dozen, or even more than twenty devices. The essential point to produce a mono recording from multiple devices is to correctly set the pan controls. To that end, the pan controls of all microphones on your mixing console or your sequencer software should be set to the central position. Modern stereo microphones actually consist of two mono microphones, whereas multichannel devices – of several mono microphones in a proper spatial arrangement. Nowadays, stereo recordings are often checked for engineering and technical mistakes (such as wave interference) when the material is changed from mono to stereo. A well-made monophonic material can be a future stereo recording. Likewise, a proper stereo recording should also be playable in mono, because many single-speaker radio receivers still rely on monophonic sound reproduction. However, while a proper monophonic recording will sound well on a stereo set, a stereo production will not necessarily prove its worth in mono. That is why sonic compatibility is vital to limit imperfections. With the technical advancements in recording and sound engineering, monophonic microphones came to be applied in various stereo or multichannel configurations. The use of a particular configuration is referred to as a recording technique, practice, or method. Therefore, the application and correct placement of monophonic microphones to produce a stereo or multichannel recording is a matter of crucial importance.

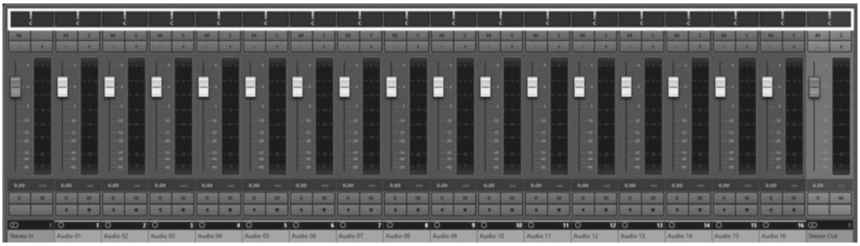

Fig. 85. The pan control set to the centre for multiple microphones implies a monophonic recording (marked with the white rectangle).

5.1.2. *The orchestra angle*

Also referred to as the extension angle of the orchestra, is not a miking technique but a general method for microphone placement relative to a monumental sound source such as the orchestra. The angle is measured from a stereo pair of microphones to the corners of the band or the orchestra. The calculation follows the formula presented in the figure 86[141].

141 See: https://lossenderosstudio.com/glossary.php?index=o [accessed: 24.06.2022].

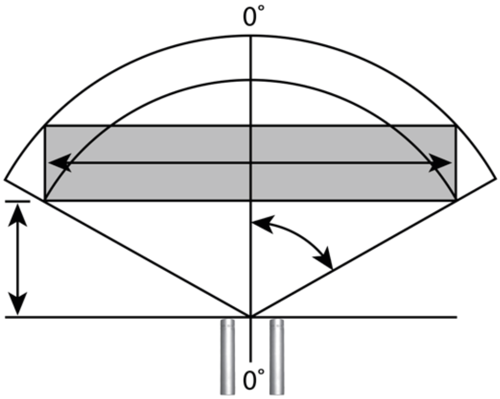

Fig. 86. The orchestra angle.

Stereo Recording Angle (SRA) is a technique developed by Michael Williams for positioning microphones relative to the sound source. It allows for setting the microphone angles, distances, etc. so that the recording played through speakers can reproduce a virtual acoustic image for the listener while using only two audio channels – left and right (fig. 87 – 91)[142].

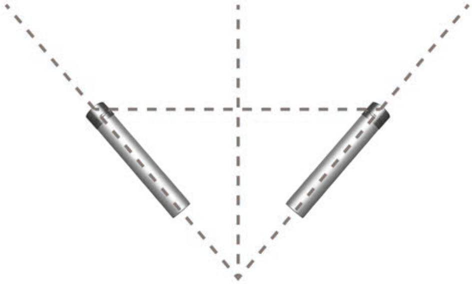

Fig. 87. Stereophonic positioning of microphones without angular offset.

142 Williams M., Dû G. L. *Multichannel sound recording Microphone Array Design (MMAD)*, https://microphone-data.com/media/filestore/articles/MMAD-10.pdf [accessed: 25.06.2022].

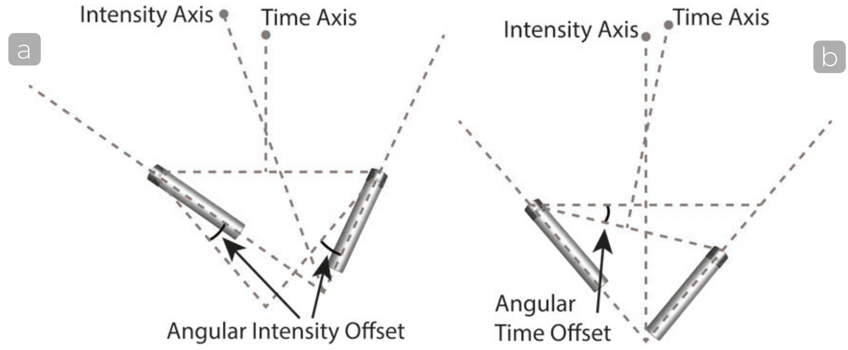

Fig. 88. a) The angular offset between the microphones makes one of them receive more direct sound, and the other – more reflected sound. This positioning creates a negative angular offset, b) the angular offset between the microphones makes one of them receive more direct sound, and the other – more reflected sound. This positioning creates a positive angular offset.

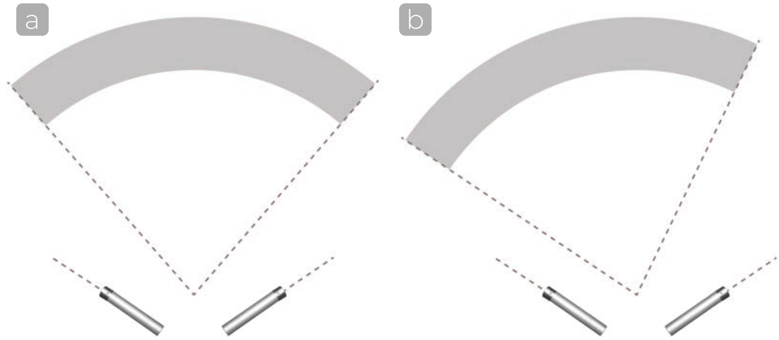

Fig. 89. a) The Stereophonic Recording Angle is between the axes of the microphones[143], b) the Stereophonic Recording Angle is offset by negative 15°, which makes it align with the axis of the left microphone.

143 Ibidem; Williams M., Dû G. L. (1999). "Microphone Array Analysis for Multichannel Sound Recording", *Audio Engineering Society*, Convention Paper Presented at the 107th Convention 1999 September 24–27 New York, http://www.mmad.info/Collected%20Papers/Multichannel/4997%20New%20York%201999%20(31%20pages).pdf [accessed: 24.06.2022], pp. 12–19.

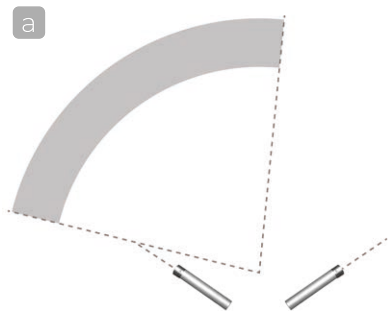

Fig. 90. a) The Stereophonic Recording Angle is offset by negative 35°, which positions one microphone for receiving direct sound, b) the Stereophonic Recording Angle is wider than that between the microphones, which lets them both receive more direct sound.

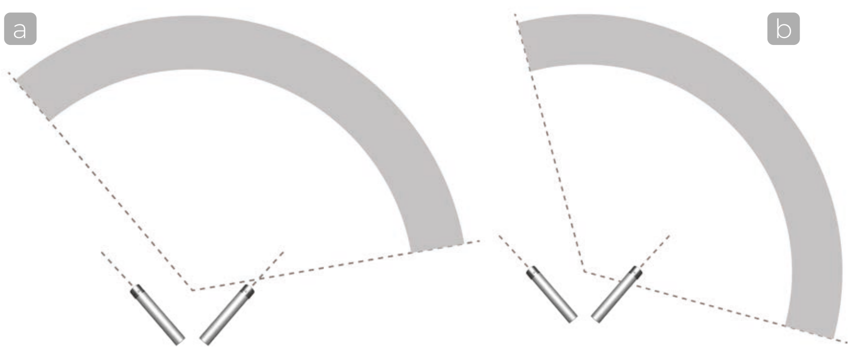

Fig. 91. a) The Stereophonic Recording Angle is offset by positive 20°, which lets the right microphone receive direct sound, while the axis of the left one is aligned to the sound source[144], b) the Stereophonic Recording Angle is offset by a positive 45°, which lets the right microphone receive direct sound almost from the centre of the sound source, while the left microphone receives sound reflecting off the walls.

144 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (1999). op. cit., pp. 12–19.

5.1.3. *The X-Y technique*

The X-Y technique uses two cardioid microphones, frequently placed on a single stand at the distance of about 5–10 cm from one diaphragm to another (fig. 92 and 93). The microphones pointed at the musicians in extreme positions (in a musical ensemble, for instance) form an angle of 90–110 degrees. The angle automatically determines the distance between the microphones and the musicians (the stage). On the one hand, a narrow angle limits the stereo base, which could produce unnatural sound. On the other, if the angle is too wide, the sounds from the centre (for instance, from the stage) may be left out from the recording, causing the so-called hole-in-the-middle effect[145].

X-Y recordings pick up mostly direct sound coming straight from the source. The share of the reflected sound is far smaller, so the ambient acoustics are only partially captured[146].

The X-Y technique is one of the most straightforward practices of audio recording in stereo. The sole requirement is the proper placement of the microphones. There is no need to continually move the devices relative to the source or to meticulously balance out the channel volume and timbre with a mixing console. When recording small ensembles, it is a good idea to add subtle digital (computer) reverb to the mix[147].

There are many arguments in favour of the XY method:

a) pickup of room acoustics depends on the size of the whole ensemble (as the distance from the microphones grows with the size of the group, more reflected sound is added to the recording),
b) natural sound of the entire ensemble (such as a choral group),
c) fair recreation of the panorama,
d) simplicity of the production process,
e) natural sonic features of the recorded sound sources,
f) only two microphones required,
g) effortless maintenance of balance between the left and the right channel.

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[145] Owsinski B. (2005). op. cit., p. 60; Edstrom B. (2011). *Recording on a Budget: How to Make Great Audio Recordings Without Breaking the Bank*, Oxford University Press, New York, pp. 78–79; White G. D., Louie G. J. (2005). *The Audio Dictionary*, 3rd edition, University of Washington Press, Seattle, London, p. 200; Holman T. (2008). op. cit., pp. 83–84.

[146] Strong J. (2012). op. cit., pp. 165–166.

[147] Whitaker J. C., Benson B. K. (2002). op. cit., pp. 4–38.

The method has also several shortcomings:

a) in large instrumental ensembles, microphones need to be suspended high up in the air to cover all rows or musicians – i.e. sound sources (choir, orchestra) – at the risk of committing a grave engineering error of capturing the first row the loudest, with the volume progressively receding in the following rows[148],
b) the recorded signal is the sum and the result of multiple sound sources, so the adjustment of individual sources in the recording is impossible (you cannot tweak the timbre, the panorama, the proportions, or the placement of individual instruments),
c) ambient acoustics are recorded together with extraneous and unwanted sounds (such as an ambulance with a blaring siren).

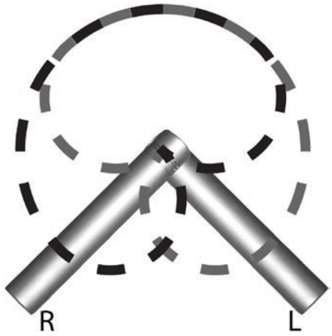

Fig. 92. The layout of microphones in the X-Y technique.

[148] Sztekmiler K. (2003). op. cit., p. 132.

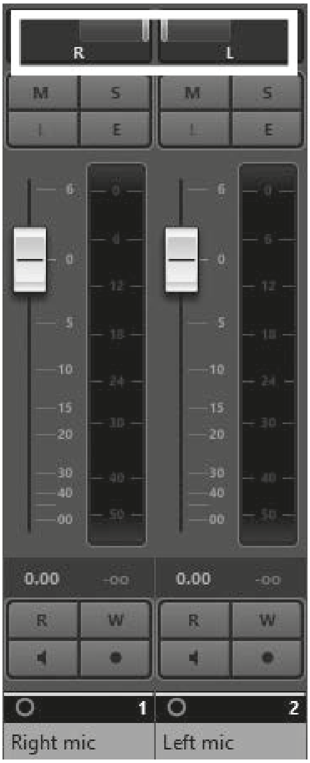

Fig. 93. Proper microphone panning used in the X-Y technique (marked with the white rectangle).

5.1.4. *The A / B method*

The A / B method allows for sound recording with two properly placed omnidirectional microphones (fig. 94 and 95). Both devices are situated in front of the stage or above it and spaced approximately 3–5 m apart (or a different distance depending on the conditions of the band and where the recording takes place). Audio pickup depends on the size (the number of performers) of the ensemble (e.g. choir)[149].

[149] Edstrom B. (2011). op. cit., p. 79.

The sounds reaching the left and right microphone will differ, which may cause phase issues[150]. The phase offset of the waveforms is caused by the differences in the time required for the sounds to travel to the microphones; in other words, the waveforms may reach the devices with a delay[151].

Pros of the A / B method:

a) the opportunity to capture ambient sound in the recording,
b) simplicity of the production process,
c) only two omnidirectional microphones required,
d) a very wide stereo base.

Cons of the A / B method:

a) poor spatial reproduction of moving sound sources (e.g. a passing car or a theatre actor walking onstage, etc.) due to the so-called bleeding when the sound switches from one channel to another (for instance, issues with recording theatre plays)[152],
b) the necessity to use high-quality microphones,
c) phase differences in the recorded material, which may negatively affect the sound,
d) no mono compatibility.

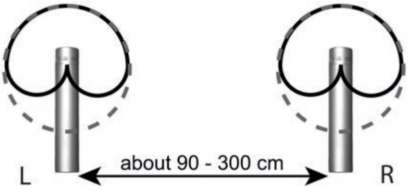

Fig. 94. The layout of microphones in the A / B technique.

150 Boyce T. (2014). op. cit., p. 265.
151 Owsinski B. (2005). op. cit., pp. 59–60; Boyce T. (2014). op. cit., p. 265.
152 Sztekmiler K. (2003). op. cit., pp. 133–134.

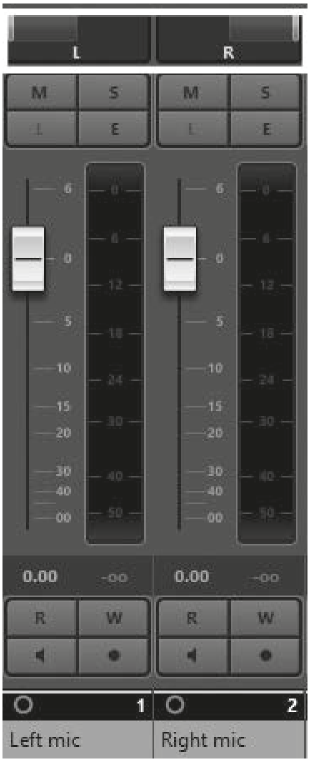

Fig. 95. Proper microphone panning used in the A / B technique (marked with the white rectangle).

5.1.5. *The A / B + M technique*

The idea behind this technique is to add one more microphone to the traditional A / B setup to avoid deviations in the panorama (fig. 96 and 97). The panorama is a tool for deciding where individual instruments should reside within the stereo image. It ensures enhanced separation of the instruments, which manifests as clear and pronounced sound. The pan control allows for setting the position of the apparent sound source between the speakers. However, this arrangement disregards the time-of-arrival differences of the music reaching the right and left ear of the listener. The resulting spatial effect is not equivalent to the real and natural direction of sound propagation in, for instance, a selected indoor space where a person is present. Moving the pan control to the centre position will create the impression of listening "from the forehead", between the left and right speaker. If the recording includes multiple tracks, and each has the pan control switched to the centre position, the end result is a monophonic production[153].

The stereo base width refers to the distance between the left and right speaker. Too narrow (close) placement of speaker units often causes unnatural squeezing of the sound field. The recording sounds unnatural because many acoustic sources overlap in a minimal sound field, resulting in auditory masking and compromised separation of acoustic sources. In contrast, placing the speaker units (studio monitors) too far apart leads to the rupture of the stereo base, less coherent sound, and the so-called hole-in-the-middle effect. All because the equipment cannot generate a sound field which focuses the audio in the right place – for instance, in front of the engineer[154].

The A and B microphones are separated by the third microphone with omnidirectional characteristics. The panorama of the middle receiver is placed at the centre of the mixing console.

The method has the same pros and cons as the A B technique. Still, the addition of another microphone allows for preserving the stereophonic space, base, and – to no small extent – the balance of the soundscape[155].

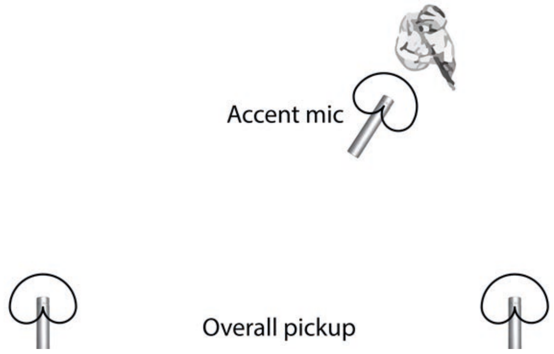

Fig. 96. The layout of microphones in the A / B + M technique.

[153] Ibidem, p. 43.
[154] See: http://www.zsl.gda.pl/egzaminy/akustyka_pomieszczen.pdf [accessed: 25.06.2022].
[155] Sztekmiler K. (2003). op. cit., p. 134.

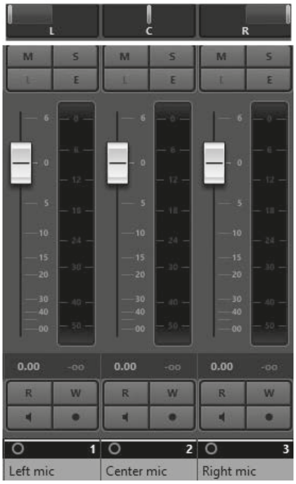

Fig. 97. Proper microphone panning used in the A / B + M technique (marked with the white rectangle).

5.1.6. *The X-Y + M technique*

This technique is used only when the recording of the entire sound source (e.g. a choir) needs to be completed with an additional voice or a solo instrument. The extra (third) cardioid device, usually referred to as a close mic, is placed near the performer. The sound engineer sets the pan to the centre and sums the signals. A particular recording may include many more close mics than one, depending on the ensemble and the creativity of the sound engineer.

In order to record the signal in phase, the close mics must pick up audio with a suitable delay. Therefore, one needs to measure the distance in meters from the close mic to the X-Y configuration. The formula for the delay that should be added to the close mics is as follows:

t (time) **=** **L** (distance in meters) **x 3 ms**

If the sound engineer does not possess an audio delay (fig. 98 and 99) unit, the angle between the microphones can be increased to more than 110 degrees to purposefully create the hole-in-the-middle effect. The hole may be then filled by the sound processed by the close mic. Although not a recommended solution, it undeniably helps remedy particular phase issues when recording.

In comparison with the traditional variant, this technique offers additional benefits such as the correction of the timbre of a solo voice or instrument, fair mixing, proper placement of solo instrument within the recording (by the role played at the given moment)[156].

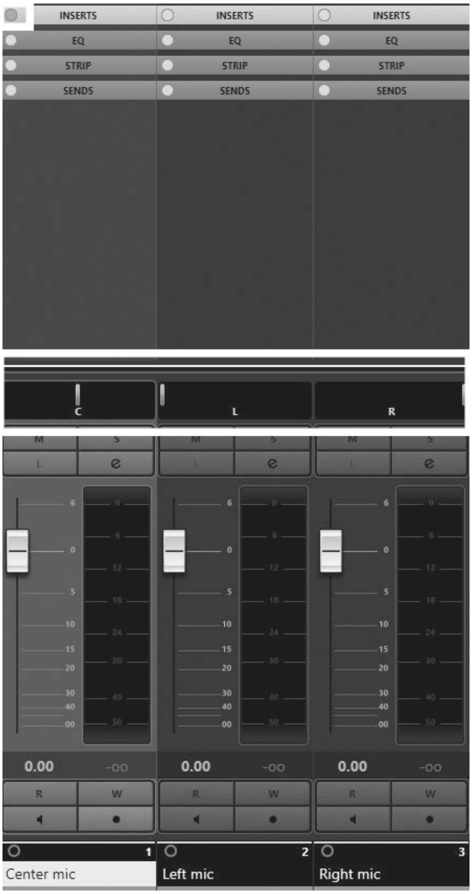

Fig. 98. Proper microphone panning used in the X-Y + M technique (marked with the white rectangle). A special delay line was used for the inserts to avoid the phase difference of the sounds reaching the central microphone.

156 Ibidem, pp. 134–136.

Fig. 99. An alternative microphone panning used in the X-Y + M technique (marked with the white rectangle). The central microphone track has a digital signal delay of 180 ms to eliminate phase issues of sounds reaching the central microphone. Consequently, the delay line is unnecessary as the delay is ensured differently.

5.1.7. *The X-Y technique + ambience*

This technique assumes that the material will be recorded indoors with four microphones. The traditional X-Y configuration is employed on one side of the room, next to the sound sources. Meanwhile, the other side (opposite to the sound source) is recorded with two extra omnis. Mixing the signal sampled by four receivers creates a sense of a far larger space. By adjusting the volume of both pairs, the ambient sound combined with the source audio can be creatively tweaked. The final result is a wide-range spatial effect, not unlike those obtained with devices for digital reverb / echo[157].

Pros of the XY + ambience technique:

a) the right localization of sound sources if the material is well-mixed,
b) good potential for adjustments in the scope of timbre and the duration of the spatial effect,
c) capture of the natural room reverberation.

Cons of the XY + ambience technique:

a) microphones are often placed in the audience, where they may be bumped or kicked; such disturbances add unwanted artefacts that are hard to mask in the mix[158],
b) as many as four high-quality condenser microphones required,

[157] McGuire S., Pritts R. (2008). *Audio Sampling. A practical guide*, Focal Press, Amsterdam, Boston, p. 90.
[158] Sztekmiler K. (2003). op. cit., p. 137.

c) pickup of additional noise created by room acoustics or external sources.

5.1.8. *ORTF / NOS*

For a new quality in audio recording, reduce the distance between two cardioid microphones, just like in the A / B technique, and space them some 17–30 cm apart (fig. 100). This combination of the A / B and X-Y techniques has sparked two new recording techniques called ORTF and NOS, respectively[159]. The application of both methods relies on the particular microphone arrangement (combination). Additionally, the microphones should point at the musicians in extreme positions at the left and right side of the stage[160]. Pros of the combined ORTF / NOS technique:

a) modifiability – the inclination of the receivers can be adjusted,
b) better stereo image when compared to the A / B technique,
c) proper sound localization owed to the X-Y method,
d) the simplicity of installation – only two microphones required,
e) the recording usually does not manifest phase differences in mono.

Cons of the combined methods ORTF / NOS technique:

a) the timbre and proportions between instruments cannot be adjusted,
b) the reverb is recorded together with additional sounds (artefacts)[161].

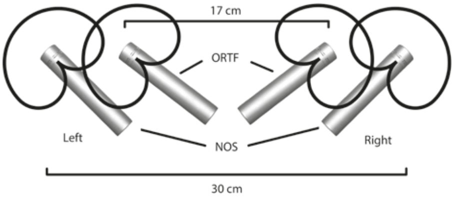

Fig. 100. The layout of microphones in the ORTF / NOS technique.

159 Ballou G., Ciandelli J. (2015). *Microphones*, in: Ballou G. (ed.). *Handbook for Sound Engineers*, 5th edition, Focal Press, Burlington, p. 652.

160 Holman T. (2008). op. cit., p. 83; Pedersen K., Grimshaw-Aagaard M. (2019). *The Recording, Mixing, and Mastering Reference Handbook*, Oxford University Press, New York, p. 148.

161 Sztekmiler K. (2003). op. cit., pp. 150–151.

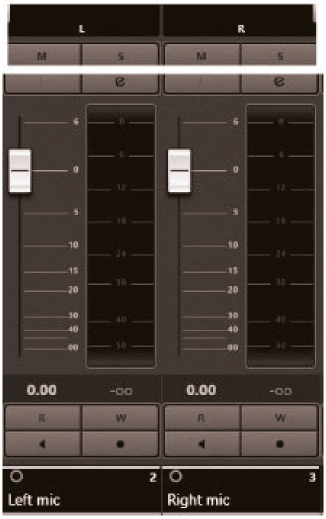

Fig. 101. Proper microphone panning used in the ORTF / NOS technique (marked with the white rectangle).

5.1.9. *The Blumlein technique*

The method uses two figure-8 microphones placed in a single point (often face-to-face, one on top of another) (fig. 102 and 104). The axes of both devices are perpendicular to each other and adjusted to pick up sound from the fringes of the stage or musicians located in extreme positions[162] (fig. 103).

Pros of the Blumlein technique include:

a) natural proportions of the ensemble,
b) a wide stereo base,
c) fair recreation of room acoustics.

Cons of the Blumlein technique include:

[162] McGuire S., Pritts R. (2008). op. cit., p. 92; Whitaker J. C., Benson B. K. (2002). op. cit., pp. 4–38, 4–39; Edstrom B. (2011). op. cit., p. 79; Kearns R. E. (2017). *Recording Tips for Music Educators: A Practical Guide for Recording School Groups*, Oxford University Press, New York, p. 40; Zotter F., Frank M. (2019). *Ambisonics. A Practical 3D Audio Theory for Recording, Studio Production, Sound Reinforcement, and Virtual Reality*, Springer, Cham, p. 2.

a) no adjustments of the timbre of individual instruments possible by parameter regulation,
b) the recording of extraneous noise produced by room acoustics or external sources,
c) no adjustments of the proportions between the volume of individual instruments possible[163].

Fig. 102. The layout of microphones in the Blumlein technique.

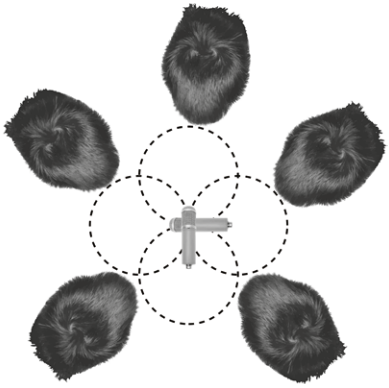

Fig. 103. The layout of microphones in the Blumlein technique.

[163] Sztekmiler K. (2003). op. cit., p. 138.

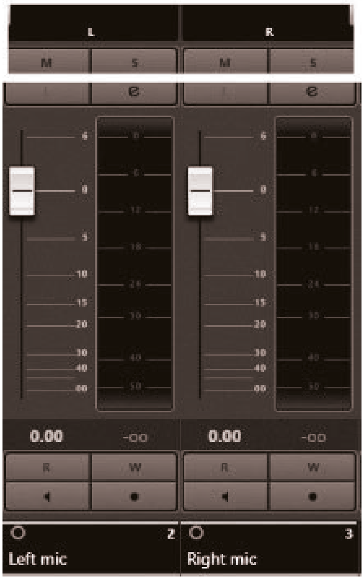

Fig. 104. Proper microphone panning used in the Blumlein technique (marked with the white rectangle).

5.1.10. *The MM technique*

The MM technique uses up to several dozen mono microphones (generally one receiver per voice or instrument) to register audio on a multitrack device. The method allows for far-reaching editing. The engineer can adjust parameters such as sound proportions, the timbre of individual instruments, the pan, special effects, spatial effects, or even the pitch if some fragment has been performed out of tune. Since the microphones are located in close proximity to the sound source, room acoustics are insignificant in the recording.

The MM technique requires expertise and practical skills on the part of the engineer since every part of the material is recorded in mono. To achieve a stereo recording, the instruments need to be correctly placed within the pan by recreating such setups as a traditional orchestral structure.

Positioning the microphones too close to each other is a mistake because the instruments resound only at a distance. An insufficient gap can lead to audio distortions. For instance, instead of capturing the entire cello, the microphone may pick up only the sound of the moving bow or only the nearest fragment of the soundbox. The microphone should be positioned to record the entire instrument and all of its vibration. The receiver and the instrument are usually spaced 30–200 cm apart, with the exact distance depending on the size and the bandwidth of the latter. Increased gap gives a warmer and better timbre. However, it could also result in crosstalk, or the multilateral spills of sound to other tracks, which makes mixing considerably harder.

The relationships described above can have both positive and negative consequences, depending on the engineer's goal and the desired style of the recording. However, the precise placement of microphones relative to the sound source and crosstalk between devices (adjacent or otherwise) are both of considerable importance.

"Entering" the material creates a fair share of both opportunities and issues. The MM recordings are made at a loss of the natural room reverb (due to the small distance from the microphone). The timbre of individual instruments can be slightly distorted. In addition, all the recordings are monophonic, without a panorama. That is why the engineer's skills become a vital issue.

Pros of the MM technique:

a) the amount of interference caused by room acoustics is insignificant,
b) discrete (multitrack) recording possible,
c) possible adjustments of the timbre of individual instruments, possible alterations in complex relationships between volumes, and full modifiability of the panorama and space,
d) opportunity to use lower-quality microphones,
e) no phase difference (mono compatibility),
f) cutting, copying, and pasting individual instrumental parts possible.

Cons of the MM technique:

a) skills in precise microphone placement required,
b) the risk of losing the authenticity of the sound,
c) a large number of microphones required,
d) the risk of disturbing the natural timbre of individual instruments and the sound proportion of the entire ensemble (by flawed adjustments),
e) a multitrack mixing console (or several consoles) required,

f) long time needed for mixing console adjustments,
g) unnatural, artificial, and dry sound unless digital reverb effect is added,
h) constant corrections on the mixing console needed for the audio response of the microphone[164].

5.1.11. *The MS technique*

As the name implies, the technique uses two signal sources: M (in German *Mitte* means "middle") and S (in German *Seite* means "side"). Such a solution allows for using either one specialty microphone created precisely for such purposes or two different receivers. In the latter case, the first microphone has a cardioid pickup pattern (a supercardioid is also an option) and captures the music from the front in mono. The other, with a figure-8 pickup pattern, records the ambience and the sound from the sides. Both microphones are located at the same spot. Two points to remember are to utilize a special matrix for distributing the signal from the figure-8 device into the right and left channels (the so-called pairing) and to reverse the phase of one of the signals[165] (fig. 105–107).

When using the MS technique, you can smoothly change between a mono and stereo recording with a special switch or a potentiometer, which allows for sound field adjustments and quick adaptation of the microphones to the acoustic conditions in the room[166].

The most important pros of the MS technique:

a) adaptation of ambient acoustics in the recorded room,
b) electronic control over the mono / stereo base,
c) relatively simple installation.

Cons of the MS technique:

a) a dedicated device required to abstract the stereo signal (unless present in the mixing console),
b) extra sounds recorded together with the acoustic background of the room[167].

[164] Ibidem, pp. 140–141.
[165] Holman T. (2008). op. cit., pp. 82–83.
[166] Whitaker J. C., Benson B. K. (2002). op. cit., pp. 4–39, 4–40.
[167] Sztekmiler K. (2003). op. cit., p. 141.

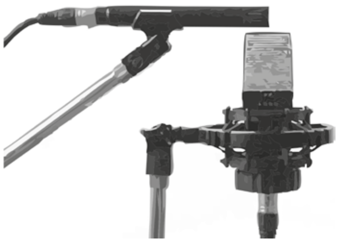

Fig. 105. The layout of microphones in the MS technique.

Fig. 106. MS microphone positioning – top view.

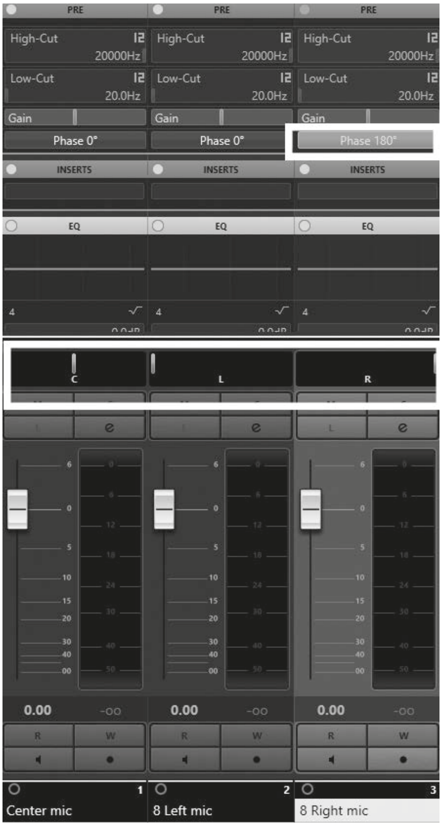

Fig. 107. Proper microphone panning used in the MS method including the necessary phase reversal by 180 degrees in one of the channels (marked with the white rectangle).

5.1.12. *MS 180°*

A miking technique which evolved from the traditional MS setup and requires two cardioids placed next to or atop one another at the closest possible distance (fig. 108). It is frequently employed to complement the sound of ambisonic microphone setups. Conversely, it does not find application on its own in recordings such as XY, AB, or ORTF[168].

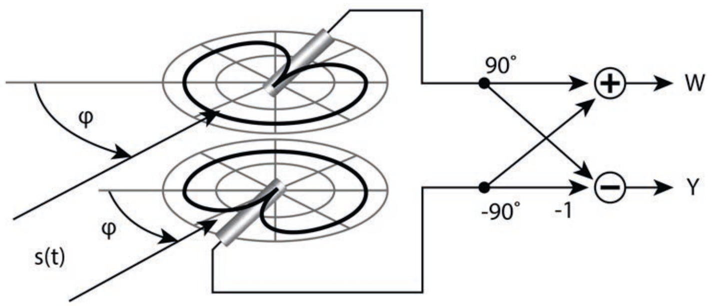

Fig. 108. The layout of microphones in the MS 180° technique.

5.1.13. *Native 2D FOA recording*

A miking technique whose acronymic name stands for First Order Ambisonic. It uses an array of three microphones positioned at the closest possible distance. The first microphone, an omni placed between two figure-8 devices, picks up the sound from the front. The figure-8 above the omni is positioned to process audio from the front and the rear, whereas the bottom figure-8 is positioned to process audio from the right and the left side (fig. 109). This microphone layout allows recording and mixing the musical material in keeping with the taste and the concept of the sound engineer[169].

168 Zotter F., Frank M. (2019). op. cit., pp. 4–5.
169 Ibidem, pp. 5–6.

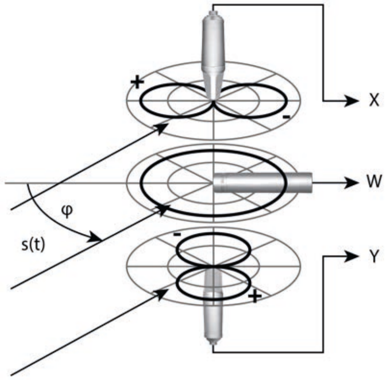

Fig. 109. The layout of microphones in the Native 2D FOA recording technique.

5.1.14. *2D FOA (1)*

One of the ambisonic variants of miking techniques. Its structure consists of four cardioids placed at the closest possible distance (fig. 110). One of the devices is positioned to process audio from the front. The microphone located atop it, rotated by 180°, captures audio from the rear, whereas the two other devices, located lower, are rotated by 90° to the left and right, accordingly, to record the left and right channels[170].

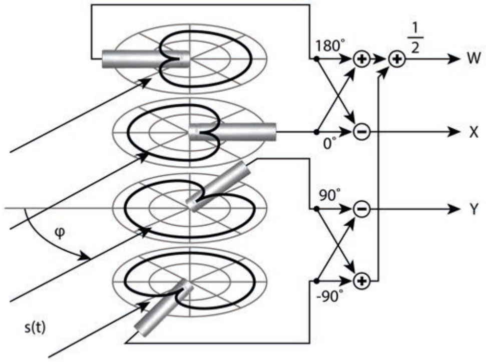

Fig. 110. The layout of microphones in the Native 2D FOA (1) technique.

[170] Ibidem, p. 7.

5.1.15. *2D FOA (2)*

Another ambisonic variation whose design comprises only three cardioid microphones placed at the closest possible distance from each other (fig. 111). The bottom microphone converts audio from the front, and the other two – rotated by 120° to the left and right relative to the bottom device – ensure the left and right channel of the recording[171].

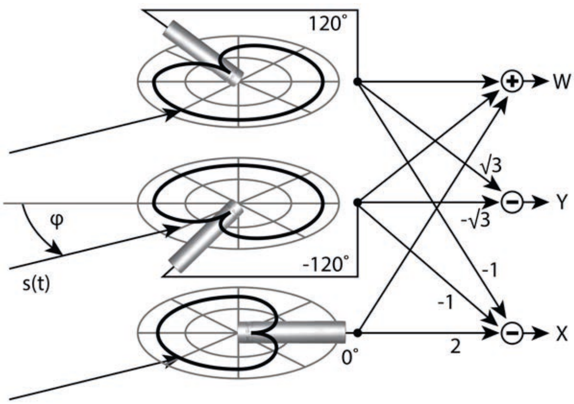

Fig. 111. The layout of microphones in the Native 2D FOA (2) technique.

5.1.16. *INA 3 (German: Ideale Nieren Anordnung or ICA 3 Ideal Cardioid Arrangement)*

A technique invented by Günther Theile, uses three cardioid microphones (fig. 112) and underlies the surround miking technique called INA 5[172]. The technique uses three channels: left, right, and centre, with one of the microphones facing the sound source. The arrangement of the other two devices, including the angles and distances, is presented in the table below[173].

171 Ibidem, p. 7.

172 See: https://lossenderosstudio.com/article.php?subject=17 [accessed: 25.06.2022].

173 Dimpker Ch. (2013). *Extended notation. The depiction of the unconventional*, Lit Verlag, Zürich – Münster, p. 309.

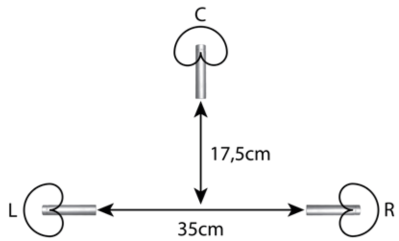

Fig. 112. The layout of microphones in the INA 3 technique.

Table 10. The arrangement of microphones in the ICA 3 technique, including the suitable angles and distances[174]

Total visual angle	Distance between microphones: L–C, C–R (distance a in figure 113)	Distance between L and C (distance *b* in figure 113)
100°	69 cm	126 cm
120°	53 cm	92 cm
140°	42 cm	68 cm
160°	32 cm	49 cm

Despite their close similarity, the setups differ in:
- the angles of rear microphones,
- the distance between the microphones constituting the setup.

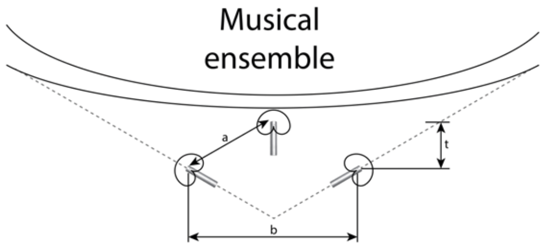

Fig. 113. An alternative layout of microphones in INA 3.

174 Ibidem, p. 309.

5.1.17. *Decca Tree / ABC Stereo*

A technique authored by Roy Wallace and Arthur Haddy in the London Decca Studios in the 1950s and frequently deployed in philharmonic halls to record large ensembles, laid the foundation for the Decca Tree Surround (fig. 114 and 115). The setup includes three omni microphones in a triangle formation, with the central receiver panned to the C position and side receivers (L and R) panned fully left and right. Such placement and panning can create a vast stereo image[175].

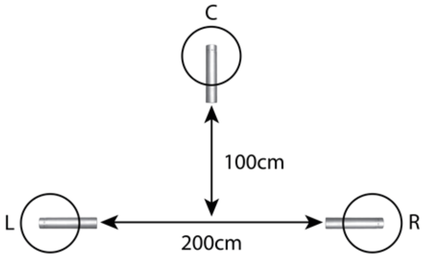

Fig. 114. The layout of microphones in the Decca Tree / ABC Stereo technique.

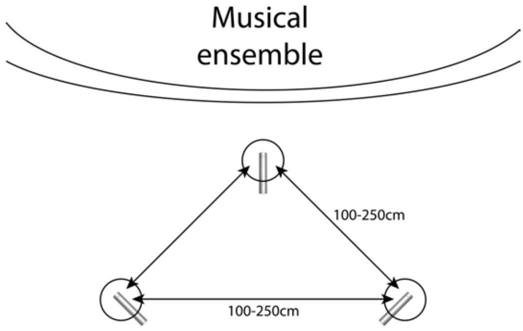

Fig. 115. An alternative layout of microphones in Decca Tree / ABC Stereo.

175 See: https://lossenderosstudio.com/article.php?subject=17 [accessed: 24.06.2022].

5.1.18. *OCT (Optimized Cardioid Triangle)*

A technique developed by Günther Theile and Helmut Wittek, underlies the OCT Surround setup. OCT uses three microphones: one central cardioid and two side super- or hypercardioids (fig. 116). The central device is panned to the C position, whereas side devices (L and R) are panned fully left or right. Such layout and panning can create a vast stereo image[176].

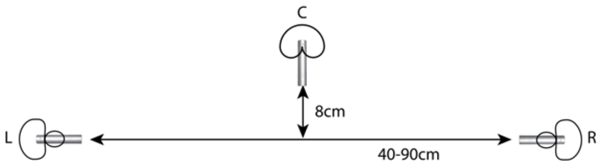

Fig. 116. The layout of microphones in the OCT technique.

5.1.19. *Omni +8*

Omni +8 has become the basis for the Omni +8 Surround technique. It uses a three-microphone array wherein a single figure-8 device features in the centre and a pair of omnidirectional devices – at the sides (L and R) (fig. 117). The omnis ensure a rich, broad sense of space and an accentuated bass response, whereas the figure-8 is a guarantee of a powerful, stabilized central (middle) channel. The distances between the microphones are not fixed and may be adjusted to achieve the optimal sound[177].

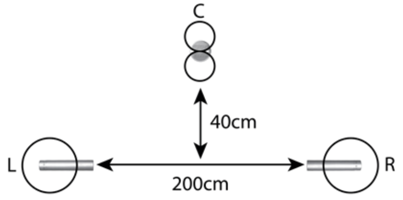

Fig. 117. The layout of microphones in the omni +8 technique.

176 Ibidem.
177 Ibidem.

5.1.20. *Omni Triad*

Invented by C. Robert Finea Mercury Studio, is a technique including three omni devices and used for orchestral recordings. The central microphone is placed in the middle, around 4.5 m in front of the orchestra, with the pan control set to C. Two extra microphones, left and right (L and R), are situated at an equal distance from the centre (fig. 118). The exact gap depends on the ensemble size and the desired sound image. When panning the left and right microphone, a vast stereo image may be obtained with the sound sources placed entirely in the left or right channel, accordingly. However, large distances between microphones may cause phase shifts and unsatisfactory sound, so instead of panning the left and right channels fully to the sides, you can position them 75%, 60% to the left / right, depending on the sound[178].

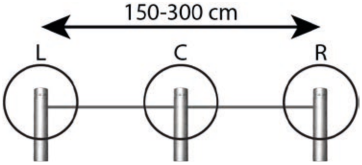

Fig. 118. The layout of microphones in the omni triad technique.

5.2. The recording of selected acoustic instruments, solo vocals and instrumental section

Before proceeding with the recording, you need to familiarize yourself with the recorded instrument (essential parameters include the size, chromatic range, its division into registers, or frequency response). Furthermore, it is vital to learn more about the audio itself (whether it is quiet, loud, spanning across the entire dynamic range, falling in the category of classical music or popular music, etc.). Finally, to record the instrument accurately and ensure the

[178] Ibidem.

best possible quality of the production, give special consideration to the position of the performer in the room.

The next step is the selection of suitable microphones. When making your choice, consider the frequency range of the instrument and the response of the receiver in question. For instance, instruments with a strident and powerful sound in higher registers (e.g. a trumpet) may be recorded with a ribbon microphone, which exhibits a fair pickup of loud and high-pitched sounds. Such an application will slightly dampen the higher registers of the trumpet in the recording, softening the tone in the higher range.

When selecting the microphone, you should also consider the pressure generated by the sound source. For instance, the bass drum of a drum set, emitting a high audio pressure, requires a different microphone than a soloist.

Microphone characteristics should also correspond to the recorded sound source. The selection of receivers is often dictated by the features of the recording room. The final choice depends to a large extent on the sound source, reverberation time in the room, the vision of the sound engineer, and their decision to include or exclude the influence of room acoustics. Another critical point is to pick the right miking technique and the diaphragm size of condenser microphones. Large-diaphragm microphones offer better coloration when compared to their small-diaphragm counterparts. The choice depends on acoustic conditions in the room, the amplitude of the acoustic background, and the impact of external phenomena on the recording process (for instance, the proximity of a big street). The right choice of the microphone and the recording technique is half the battle.

5.2.1. *The piano*

Keep in mind that placing the microphone too close to the strings may produce a sharper sound and the key action mechanism and the pedals could become audible in the mix. A microphone situated inside the box accentuates the central part of the bandwidth, whereas one located to the side highlights the higher registers. Positioning the microphones farther away from the source is the so-called golden mean in audio recording because the sound is picked up in full force together with the balanced, natural acoustic background of the recording room. Below you will find several example microphone configurations for a piano recording:

Configuration 1:

a) the X-Y technique,
b) close distance between the microphones and the half of the curved side of the piano microphones positioned 180 cm away from the piano,
c) microphones inclined at the same angle relative to the strings or the lid of the piano,
d) increased distance between the piano and the sound source to add more room reverb into the recording[179];

Configuration 2:

a) the use of a stereophonic technique (frequently X-Y to limit phase difference) over the head of the pianist,
b) microphones aimed at the soundboard,
c) a “mellow” sound that sits well in the mix,
d) increased distance between the piano and the sound source to add more room reverb into the recording[180];

Configuration 3:

a) two cardioid microphones,
b) one microphone placed at the back of the piano and aimed at the low-sounding strings,
c) the other microphone placed above the hammer mechanism, often in the centre[181];

Configuration 4:

a) two small-diaphragm condenser microphones,
b) the microphones spaced around 120 cm apart,
c) the microphones mounted at the height of around 240–270 cm and aimed at the soundboard,
d) microphones positioned 180 cm away from the piano,
e) the piano lid fully open[182];

179 Sztekmiler K. (2003). op. cit., p. 167.
180 Kefauver A. P. (2001). op. cit., pp. 142–143.
181 Owsinski B. (2005). op. cit., p. 172.
182 Ibidem, pp. 172–173.

Configuration 5:

a) the X-Y technique applied inside the soundboard, with the lid fully open,
b) microphones positioned over the strings, at the crossing of the strings of the low and high registers[183].

5.2.2. *The violin and the viola*

Because of the structure of these instruments, the vibrations in the violin and the alto are carried by the soundbox to the openings which emit the sound upwards and to the right.
Example configurations:

Configuration 1:

a) a single condenser microphone,
b) the microphone positioned above the instrument,
c) the distance between the instrument and the microphone: 180 cm,
d) the sound of the instrument recorded in full, with limited audibility of the bow[184];

Configuration 2:

a) a pair of microphones,
b) a ribbon microphone placed slightly above the instrument (a focus on direct sound pickup),
c) another microphone placed below the instrument to achieve complementary signals,
d) skilful mixing of the two signals[185].

5.2.3. *The acoustic guitar*

The acoustic guitar may also be recorded in multiple microphone arrangements:

Configuration 1:

a) a pair of condenser microphones,
b) the X-Y technique,

183 Bartlett B., Bartlett J. (2013). op. cit., pp. 135–136.
184 Huber D. M., Runstein R. E. (2010). op. cit., p. 167.
185 Owsinski B. (2005). op. cit., pp. 149–150.

c) one of the microphones aimed at the bridge, the other at the 12th fret area[186];

Configuration 2:

a) a pair of condenser microphones or a single microphone,
b) the main microphone positioned 20 cm away from the neck and the body of the instrument,
c) the other microphone (optional) positioned around 25 cm away; aimed at the end of the soundbox[187];

Configuration 3:

a) a single omnidirectional microphone,
b) the microphone positioned 10–20 cm away from the soundhole[188];

Configuration 4:

a) a pair of microphones (one dynamic device and one condenser device),
b) the dynamic microphone pointed at the body of the instrument,
c) the condenser microphone positioned at the height of the left ear of the guitarist, aimed at the 12th fret of the instrument[189].

5.2.4. *The classical guitar*

The techniques used for recording the acoustic guitar still apply. Additionally, recording the classical guitar requires considering its structure and the direction of the vibrations. Commonly noted differences, when compared to acoustic guitars, include the shape of the soundbox, the width of the neck, and the type of strings. A different structure creates an opportunity to use other microphone configurations. The sound produced by a classical guitar

[186] Ibidem, pp. 151–154.
[187] Ibidem.
[188] Ibidem, p. 152.
[189] Snoman R. (2009). *Dance music manual. Tools, Toys and Techniques*, 2nd edition, Focal Press/Elsevier, Burlington, pp. 171–172.

radiates downwards (to the floor), to the right (if the performer is right-handed).

Three example configurations:

Configuration 1:

a) one small-diaphragm condenser microphone,
b) the microphone positioned around 15–20 cm away from the soundhole,
c) the microphone aimed at the bridge[190];

Configuration 2:

a) a pair of condenser microphones (one small-diaphragm and one large- diaphragm device),
b) the small- diaphragm microphone located on a stand at the level of the guitarist's left ear, aimed at the 12th fret of the instrument,
c) the large-diaphragm microphone located at the same level but around 30 cm away from the strap lock[191];

Configuration 3:

a) a pair of condenser microphones,
b) one microphone located at a distance of 60–90 cm to the right of the guitarist, near the floor,
c) the other microphone aimed at the 12th fret of the instrument (at the level of the 12th fret)[192].

5.2.5. *The harp*

Two condenser microphones with a cardioid pickup pattern feature in front of the instrument, pointing at the strings on both sides. The distance between the microphones is around 100 cm, with the harp placed in the middle (at 50 cm from the left and right device)[193].

190 Owsinski B. (2005). op. cit., pp. 155–156.
191 Ibidem.
192 Clement V. (2004). *The complete studio guitarist. The guitarist's guide to session work and home recording*, Alfred Publishing, Van Nuys, p. 86.
193 Eargle J. (2005). *The Microphone Book. From Mono to Stereo to Surround. A Guide to Microphone design and Application*, Focal Press, Burlington, p. 233.

5.2.6. *The electric guitar and the bass guitar*

In terms of engineering complexities, both instruments can be treated in a similar fashion. However, note that the panorama can be altered with computer effects such as stereo (VST) only in the case of the electric guitar. The same cannot be applied to the bass because low-frequency sounds radiate omnidirectionally. For that reason, a bass guitar is recorded in mono, without effects, and its pan control remains in the centre position.

Both instruments can be recorded from the so-called line (the instrument plugged directly to the mixing console) or a music card. This technique will produce a dry but clear guitar sound without ambience, which can be a desired result or the exact opposite, depending on the sound engineer's intentions.

The sound is captured in full force using an amplifier. Amplifiers, as a product of multiple technologies, can give an entirely new sound to the instrument, which opens the way for recording the unique effect produced by the particular amplifier. Such recordings are made with various microphones positioned at the speaker of the amplifier. Keep in mind that in a studio, a single speaker may be recorded with a few different receivers to select the best variant or blend several options in a single mix. This concept does not assume the use of any uniform recording technique. That is because, with the diversity of microphone frequency response characteristics and sounds produced by boosters, a universal technique does not exist.

Remember that a microphone placed at the centre of the speaker (next to the dome) will register a fuller, more balanced frequency range, often with the audible middle registers. Such frequency radiation is the corollary of the structure of a cone speaker. In contrast, a microphone placed to the side will register a far darker, matte sound – which is not an error but the concept of the sound engineer[194].

5.2.7. *Leslie speaker (Hammond organ)*

Rotating speakers, which form part of the instrument, require a special approach differing from the established miking techniques. The Hammond organ may be recorded with either dynamic or condenser microphones. The description below refers to dynamic microphones, but the same layout is used for condenser devices.

[194] Slone J. J. (2002). op. cit., pp. 28–29.

Two dynamic microphones are placed on the left (left mic) and right (right mic) side of the casing, next to the special apertures for sound emission. One large-diaphragm dynamic microphone for bass (for instance, for the central drum in a drum kit) is located at the bottom to the front, next to the special apertures for sound emission[195]. Channel mixing involves panning the left mic to the left channel and the right mic to the right channel, whereas the sound picked up by the large-diaphragm microphone is panned to the central position. Additionally, its volume is adjusted to complement the character of the music (for instance, the need for a less or more intense bass response).

5.2.8. *The saxophone*

A cardioid condenser microphone sits 30–200 cm from the bell[196], depending on the desired sound image. The proper location is in front of the bell and not the musician. Close miking will produce the sound of a vast tonal spectrum, both sharp and analytical. A larger distance brings the need to flatten the instrument tonality as the recorded audio includes the direct wave and the wall, ceiling, and floor reflections. Recordings of this type can elicit the ambient sound recommended in some productions due to the character of the performed music.

5.2.9. *The trombone, the French horn and the trumpet*

The trombone, the French horn and the trumpet are all recorded in a similar style as for wind instruments. Every instrument requires a single cardioid condenser microphone. The recording device should be adequately angled relative to the bell to ensure that the microphone faces the instrument in a straight line. The distance between the bell and the microphone depends on the instrument:

- 50–100 cm for the trombone,
- 100–200 cm for the French horn,
- 30–100 cm for the trumpet[197].

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[195] Dowsett P. (2016). *Audio Production Tips. Getting the Sound Right at the Source*, Focal Press, New York, p. 464.

[196] Eargle J. (2005). op. cit., p. 232; ibidem, p. 453.

[197] Eargle J. (2005). op. cit., p. 233.

A small distance will produce the sound of a broad tonal spectrum, both sharp and analytical. However, the (potential) hiss of the air travelling through pipes and vents may become a nuisance (making the audio too analytical and aggressive). Therefore, for a satisfactory recording, it is suggested to space the device and the instrument further apart. As the distance increases, the recorded sound image flattens owing to reflections caused by the room (like in saxophone recordings).

5.2.10. *The transverse flute (Western concert flute)*

This instrument is recorded with a single condenser microphone exhibiting a cardioid pickup pattern. The placement varies depending on the desired sound:

- for orchestral sound, the microphone features approximately 50–60 cm above the flute, and the recording device is pointed more to the foot joint (a darker sound of the flute),
- for folk sound, the microphone features approximately 30–40 cm above the flute, and the recording device is pointed to the central part of the instrument (a lighter sound of the flute),
- for jazz and dominating sound, the microphone features approximately 30–40 cm above the flute, and the recording device is pointed to the embouchure hole (very bright sound, audible whizz and hiss of the air possible); pointing the microphone to the head (sub-optimally, to the embouchure hole) may partially eliminate the hiss of the air present during the recording[198].

5.2.11. *The recorder (the English flute) / the piccolo*

Recording these instruments requires a single condenser microphone exhibiting a cardioid pickup pattern placed above the instrument, at a distance of 30–60 cm. The microphone should be pointed at the central part of the instrument. The diaphragm needs to face the instrument in a straight line, and the microphone angle must resemble the angle of the recorded instrument[199].

[198] Dowsett P. (2016). op. cit., p. 449.
[199] Ibidem, pp. 449–450.

5.2.12. *The shaker*

The recording requires a single condenser microphone with a cardioid pickup pattern. The height of the microphone is adjusted to the level of the shaker, which is moved back and forth towards the microphone[200]. The instrument and the recording device are spaced approximately 30–50 cm apart depending on the desired spatial image.

5.2.13. *The vibraphone / the xylophone*

These instruments may be recorded with either of the three techniques using condenser microphones. The first of these, the A/B technique, requires cardioid microphones mounted on stands at the height of 50–100 cm from the bar of the instrument. The devices are placed at points corresponding to 1/3 and 2/3 of the instrument width.

The second technique is an additional pair in the A/B setup. In the case of the vibraphone, it captures the sound from below. To record the sound in full force, the signals from resonator tubes below the instrument are mixed with the first technique. Thus, this method uses a total of four cardioids[201].

The third technique, ORTF / NOS, is also situated 50–100 cm above the bars. The only difference is the placement of the microphones in the middle of the vibraphone. In accordance with the ORTF / NOS technique, they sit at a precisely defined angle[202].

5.2.14. *The drums*

There are many ways to record a drum kit. Panning is an interesting aspect, as kit components may be set between the left and right channel to match the auditory impressions of the drummer or the audience. The setup depends on the taste and the preference of the listeners or the engineer.

All of the microphones are cardioids except the cases listed below since:

- the toms are recorded with dynamic microphones,
- the cymbals are recorded with condenser microphones (small- or large-diaphragm),

200 Ibidem, p. 453.
201 Ibidem, p. 458.
202 Eargle J. (2005). op. cit., p. 224.

- the central drum is recorded with a large-diaphragm dynamic microphone and, optionally, a subsonic microphone (which looks like a combination of a speaker and one of the toms); he subsonic device serves to process the bands of a very low frequency and is not in common use),
- the hi-hat may be recorded with a dynamic microphone, although a small-diaphragm condenser microphone will ensure a better performance (fig. 119).

Possible miking setups for drums are presented below:

- every tom, the snare drum, the central drum, and the hi-hat all have their own microphones, the cymbals have two microphones, the so-called overheads, placed approximately 50–100 cm higher – the most common setup;
- every tom, the snare drum, and the central drum all have their own microphones, while the hi-hat does not; the cymbals have two microphones placed approximately 50–100 cm higher in a manner allowing for the conversion of the hi-hat sound; the microphones placed higher are overheads;
- the snare drum and the central drum have their own microphones, the toms have figure-8 devices placed in-between, the hi-hat may have its own microphone, but if it does not, the sound is recorded by the overheads; the cymbals have two microphones placed approximately 50–100 cm higher – this setup uses the smallest number of microphones but offers considerably limited options for sound editing and panning.

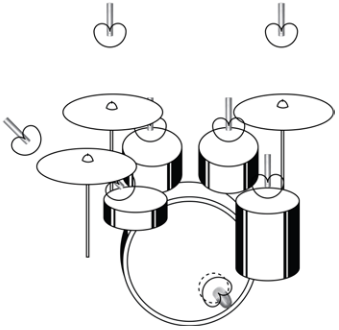


Fig. 119. Recording setup for a percussion set (the most common variant).

5.2.15. *Solo vocals*

The first step in voice recording is to realize that not every microphone will work for recording a particular voice. The limitations arise due to the construction of the microphones themselves and the vast variety in human voices, in terms of not only pitch but also timbre. Thus, a microphone may successfully capture one voice but fail to suit another. Keeping that in mind is essential as the voice plays a unique role in musical recordings. No matter how agile or melodious, the voice conveys the text comprehended by the listener. Therefore, the audience may subconsciously focus on the vocals more than on the instruments. Additionally, to maintain speech intelligibility, vocals are often set to sound louder than the accompanying musical arrangement during mixing. They may often dominate the entire mix. Therefore, excellent voice recording is a must.

A singing voice is captured by a single cardioid microphone with a pop filter (a circle several centimetres in diameter covered with the material mounted to the microphone stand) which serves to eliminate plosives such as “p” or “b” from the recording. The filter remains in close proximity to the microphone, at a distance of several centimetres. The vocalist’s lips are placed several centimetres apart from the filter. The height of the microphone may be adjusted and set to match the level of the singer’s lips. The microphone arrangement with the cable placed upwards (as in radio productions) or downwards (as in all other recording types) has no bearing on the quality and sound, as the device captures the proper soundwave.

The second method involves an array of cardioids. Every microphone may be different, producing characteristic and individual sound, with a pop filter placed in the front. In that case, the vocalist sings into all microphones simultaneously as they are located in proximity in a manner allowing for capturing direct sound from the singer’s lips. The devices tend to be spaced 2–10 mm apart to avoid direct contact but ensure optimal conversion of the singer’s voice. All microphones are panned to the central position, which produces a monophonic recording despite the use of multiple devices. The sound engineer selects the best-sounding microphones during playback and leaves them in the mix, while the others are switched off. At the next stage of the processing, the engineer mixes the loudness of individual microphones to achieve the best sound of the recorded vocals. This manoeuvre serves to expose the characteristic timbre of the voice because – regardless of the impression that the vocalist was singing to a single microphone – the audio is mixed from various recording devices, each having its individual sonic properties.

The third method of recording a singing voice uses two cardioids featuring at different distances from the vocalist. One microphone sits close to the singer's lips (like in the first method) and the other may be positioned in a straight line around 50–200 cm further from the former. It is worth remembering that to eliminate phase issues in the second microphone, the first one should be delayed based on the multiplication of the propagation speed by the distance provided in meters.

This setup may prove successful in recording a voice of remarkable weight, as the use of a single device may prove technically impossible. It is also applied for two superposed recordings (with the slightly modulated delayed signal overlapping the original one) to elicit a degree of room reverberation and the natural effect known in recording productions as the chorus.

5.2.16. *The instrumental section*

A musical production involving a larger instrumental ensemble requires extensive knowledge and experience from the engineer, for the difficulty level in such projects is exceptionally high (fig. 120). The sweet spot of optimal playback, where the ensemble sounds good despite the impact of room acoustics, is captured with a single stereo technique. Frequent choices include the X-Y configuration, around 10 cm behind the conductor, or the A / B configuration with the microphones spaced around 120 cm apart. This main pair is used to obtain a holistically uniform sound consistent with the vision of the conductor or the composer.

The extra microphones (close mics) situated directly at the instruments capture direct sound (sometimes not fully developed in terms of timbre) which needs to be mixed with the main technique. Additional microphones must be used carefully for they ensure the quality of the solos. Sound capture from up close does not always produce the desired effect. Potential risks involve the registering of strident tones, which later on need to be mixed with the rest of the material (the main technique) to obtain a sonically uniform material.

A more sizeable instrumental section should be recorded in a dedicated room with excellent acoustics and little background noise. In spaces with poor acoustics, the microphones spaced apart pick up unwanted artefacts and impair the overall impression by exposing various acoustic defects[203].

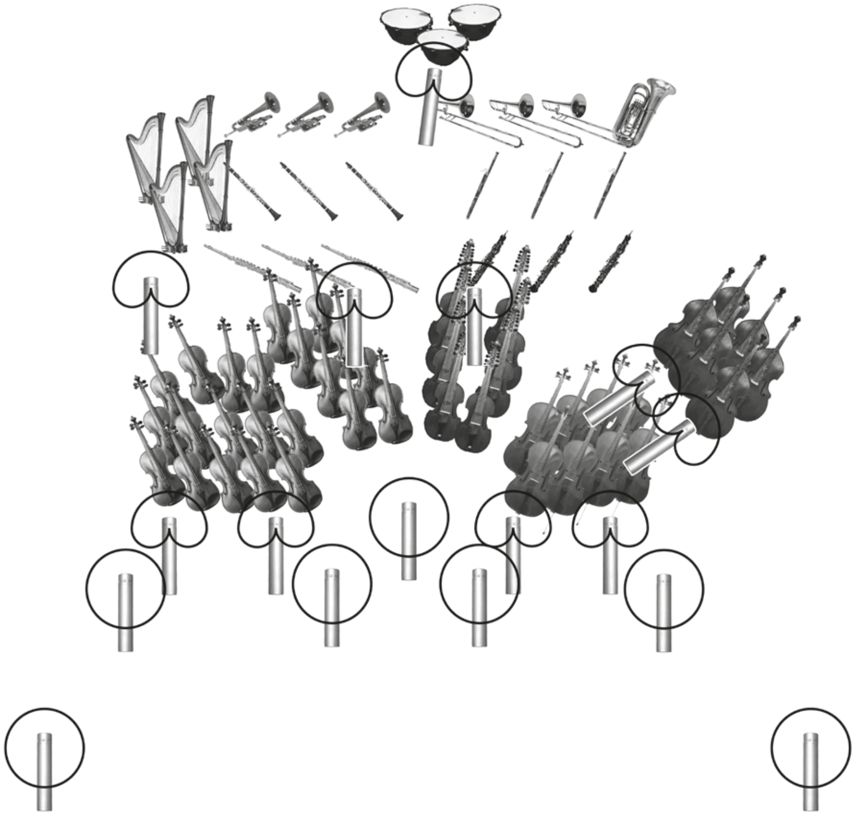


Fig. 120. Stereophonic recording of a symphony orchestra.

[203] Sztekmiler K. (2003). op. cit., pp. 139, 146–168.

Chapter 6. Production methods for surround recordings

Multichannel recordings allow for encoding audio with either multi- or mono- microphones. Those productions typically use a larger number of devices and special matrices for the distribution of signals which will later make up the final track. When making surround recordings, sound engineers stick to guidelines concerning the position of the microphones relative to each other and the sound source (just like in traditional techniques). The directionality of each receiver should be exploited for the best quality, which manifests as the full spatial image of the audio.

6.1. Multi-channel miking techniques

6.1.1. *IRT atmo-cross*

It comprises four microphones forming a square with the side length of 25–40 cm. The devices, angled at 30–45° relative to the right angle, should exhibit cardioid or omnidirectional sound pickup pattern (fig. 121). IRT atmo-cross is an evolution of other surround techniques and aims to add more space to the main setup. The arrangement allows for registering sound reflected from the walls in the form of delays[204].

[204] Rumsey F., McCormick T. (2006). op. cit., p. 512.

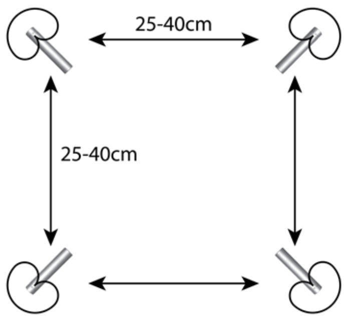


Fig. 121. The layout of microphones in the IRT atmo-cross technique.

6.1.2. *Hamasaki Surround / NHK Surround + Ambience Matrix*

It is the Hamasaki Surround / NHK Surround technique expanded with four cardioid or omni microphones forming a square with the side length of 100 cm (fig. 122). In contrast to the traditional Hamasaki Surround / NHK Surround, The Ambience Matrix technique is always located a few meters away from the sound source. As microphone pairs are oriented sideways, the space created by the delayed sound reflected from the walls is added to the production[205].

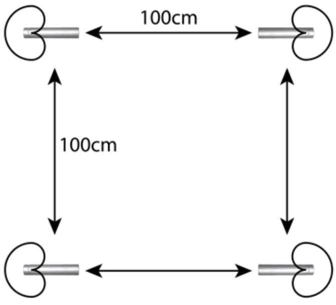


Fig. 122. The layout of microphones in the Hamasaki Surround / NHK Surround + Ambience Matrix technique.

205 Ibidem, p. 552.

6.1.3. *3D FOA*

Yet another ambisonic variation whose design uses four microphones placed at the closest possible distance from each other (fig. 123). A single omni device converts the surrounding signal, while the figure-8 located next to it (either on the left or right side) processes the downward (reflected from the floor) and upward (reflected from the ceiling) sound. The next figure-8 located atop the central pair is positioned to capture audio from the front and the rear. The last figure-8, situated at the bottom, records the left and right channel[206].

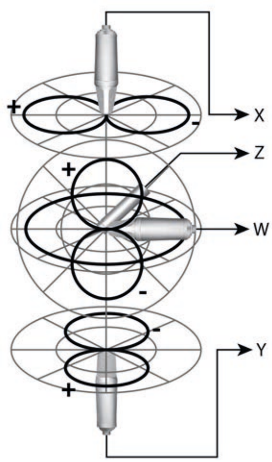


Fig. 123. The layout of microphones in the 3D FOA technique.

6.1.4. *Tetrahedral array*

It comprises four cardioid microphones (fig. 124 a) and b)). Keep in mind that many manufacturers now produce single devices housing four inbuilt microphone capsules, in a design comparable to a single device equipped with four smaller microphones mounted inside. The microphone layout used in the technique is presented in the figures 124 a) and b)[207].

206 Zotter F., Frank M. (2019). op. cit., p. 10.
207 Ibidem, pp. 11–12.

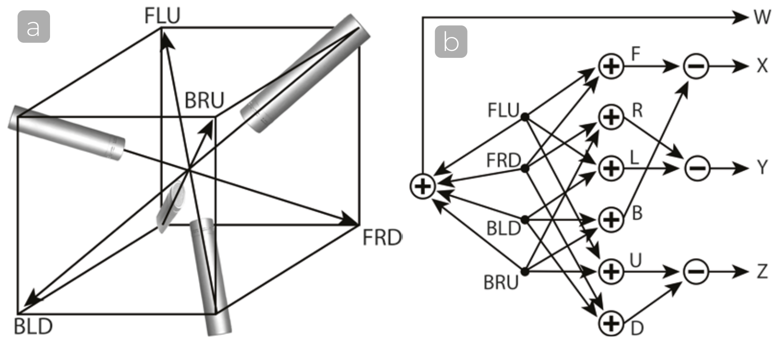


Fig. 124. a) The layout of microphones in the tetrahedral array technique and b) Technical parameters of microphone connections including signal phase shifts.

6.1.5. *Double X-Y*

A miking technique developed based on the X-Y stereo setup. The first and the second pair of X-Y microphones are spaced a few meters apart, with the exact distance dependent on the ensemble size and the desired auditory effect (fig. 125). This technique is applied for recording large ensembles, for instance, in philharmonic halls. Conversely, it is unsuitable for recording individual instruments, such as the flute, the acoustic guitar, or the piano. The L_L and R_R are fully panned to the left and right, accordingly, whereas the central channel is the sum of L_R and R_L minus 3 dB[208].

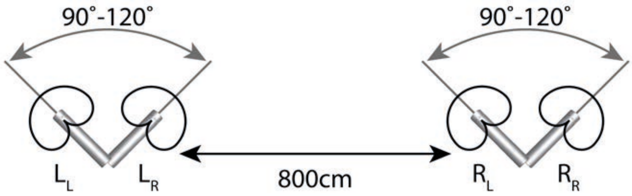


Fig. 125. The layout of microphones in the double X-Y technique.

[208] https://lossenderosstudio.com/article.php?subject=17 [accessed: 25.06.2022].

6.1.6. *Double A / B*

A miking technique developed based on the A / B stereo set-up, although it uses a five-microphone array of cardioids rather than omnidirectional devices. All microphones sit in a single file, usually across the entire width of the stage. Consequently, they should be spaced evenly (for instance, every 2.5 m), although the distance between the devices is not a fixed value(fig. 126). This technique is useful for recording large ensembles, for example in philharmonic halls. Conversely, it is unsuitable for recording individual instruments such as the trombone, the harp, the oboe, or the piano. The L_L + L channel is panned left, with the intensity of the L channel reduced by 3 dB, whereas the R_R + R channel is panned right, with the intensity of the R channel reduced by 3 dB, and the central channel – C + L + R, with L and R reduced by around 3 dB[209].

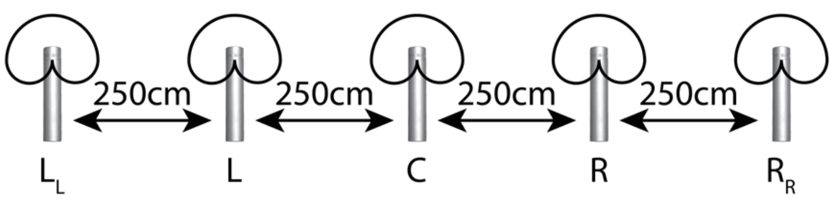


Fig. 126. The layout of microphones in the double A / B technique.

6.1.7. *Double MS*

This technique (which was already partially discussed) is based on the M / S method with an additional cardioid or omnidirectional microphone pointed to the back (fig. 127–130). Double MS uses two discrete M / S arrays spaced apart[210].

209 Ibidem.

210 Rumsey F. (2001). *Spatial Audio*, Focal Press, Oxford, pp. 200–201; Holman T. (2010). *Sound for Film and Television*, 3rd edition, Focal Press, Burlington, p. 74; Zotter F., Frank M. (2019). op. cit., p. 6; Bartlett B., Bartlett J. (2009). op. cit., p. 477–478.

Fig. 127. A microphone set for Double MS recordings.

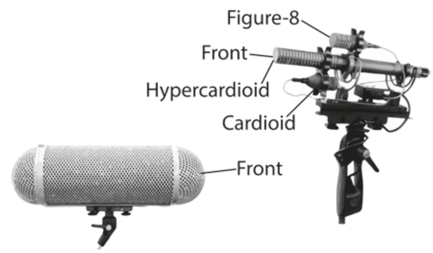


Fig. 128. The Double MS recording method using the shotgun microphone, most commonly deployed in the movies, and a special anti-wind cover.

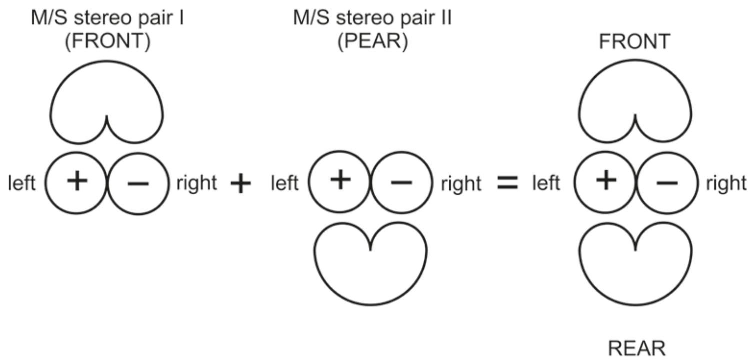


Fig. 129. Microphone directionality characteristics in Double MS.

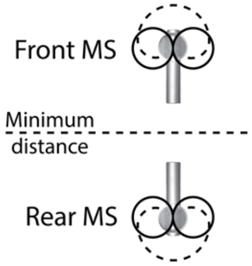


Fig. 130. An alternative layout of microphones in Double MS.

6.1.8. *IRT Cross / ORTF Surround (Institute of Radio Technology)*

This method uses four cardioid microphones in a double ORTF set-up (fig. 131–133). The names of the two variants rarely appear alongside each other, being frequently regarded as the same method due to their high similarity. The general assumptions of the technique may be further tweaked to resemble either the ORTF or the IRT set-up, depending on microphone positions and the distance between them[211].

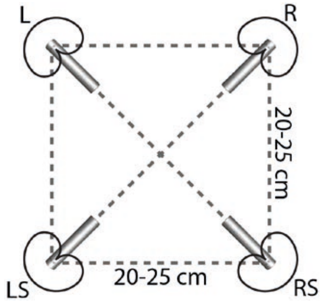


Fig. 131. The layout of microphones in the IRT Cross technique[212].

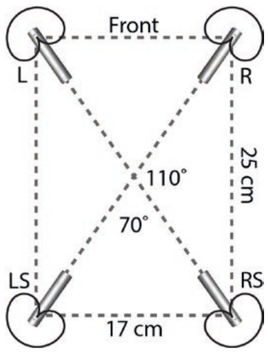


Fig. 132. The layout of microphones in the ORTF Surround (double ORTF) method[213].

211 Rumsey F., McCormick T. (2006). op. cit., p. 510; Owsinski B. (2005). op. cit., p. 216; Réveillac J. M. (2018). op. cit., pp. 62–63; Kassier R., Lee H. K., Brookes T., Rumsey F. (2005). "An Informal Comparison Between Surround-Sound Microphone Techniques", *Journal Audio Engineering Society*, Convention Paper 6429, Presented at the 118th Convention 2005 May 28–31 Barcelona, http://www.aes.org/tmpFiles/elib/20200210/13145.pdf [accessed: 25.06.2022], p. 5.

212 Wuttke J. (2005). "Surround Recording of Music: Problems and Solutions", *Journal Audio Engineering Society*, Convention Paper 6556, Presented at the 119th Convention 2005 October 7–10 New York, http://www.aes.org/tmpFiles/elib/20200210/13356.pdf [accessed: 26.06.2022], p. 7.

213 Ibidem.

Fig. 133. A microphone positioned according to the double ORTF technique. Devices of this type are frequently used in movie productions for the cinemas and the television. The long-filament filter is useful for ensuring a clean sound by the reduction of wind noise if the recording is made at the seaside, in high mountains, or other extraneous sounds.

6.1.9. *OCT Surround (Optimized Cardioid Triangle)*

A multi-channel miking technique which uses five or seven devices (fig. 134–135). The former configuration requires one cardioid device in the centre, two hypercardioid devices on the sides, and other two cardioid devices in the rear. The latter configurations need two extra omnidirectional microphones situated just behind the lateral hypercardioids. Both techniques can be mixed in a five-microphone array which replaces the lateral hypercardioids with a pair of omnis[214].

[214] Murphy J. J. (2016). *Production Sound Mixing: The Art and Craft of Sound Recording for the Moving Image*, Bloomsbury Academic Publishing, New York/London, p. 211; Friesecke A. (2007). op. cit., p. 435.

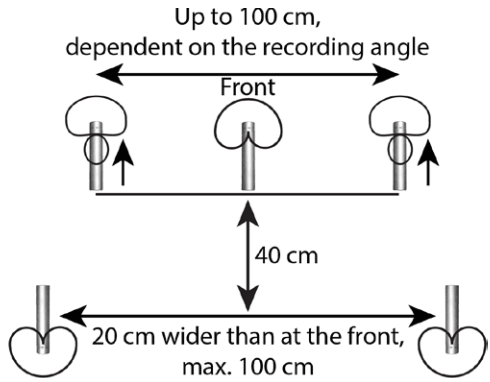


Fig. 134. The layout of microphones in the OCT Surround technique.

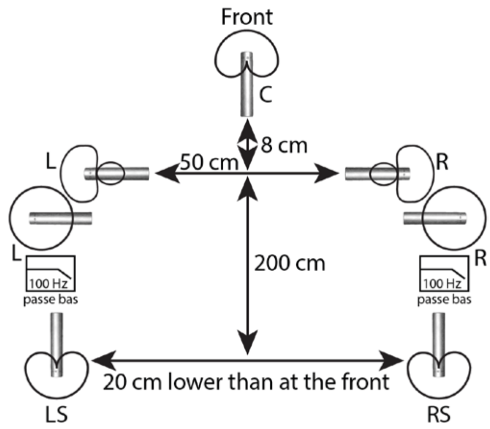


Fig. 135. An alternative layout of microphones
in the OCT Surround technique.

6.1.10. *OCT + IRT Cross*

A variant of the miking technique which relies on two other techniques operating concurrently in sync (fig. 136). In this case, OCT provides a so-called base for a stereo recording with the centre channel (L + C + R) (2.1) (discussed in detail on chapter 5.1.18), whereas IRT comprises four cardioid microphones forming a square, tilted at 90 degrees relative to each other and 45 degrees relative to the sound source axis[215].

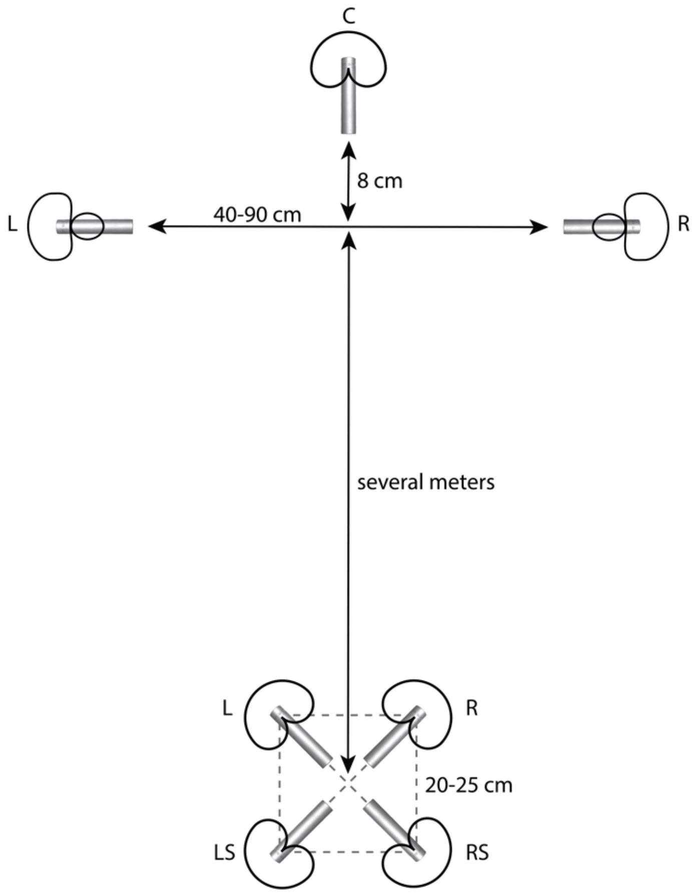


Fig. 136. An alternative layout of microphones in the OCT Surround technique.

[215] See: https://lossenderosstudio.com/article.php?subject=17 [accessed: 26.06.2022].

6.1.11. *Hamasaki Square*

The technique relies on four figure-8 microphones forming a square (fig. 137). Pay attention to the connections to avoid phase offset of the arriving signals[216].

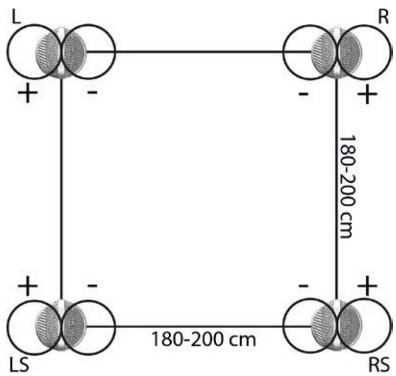


Fig. 137. The layout of microphones in the Hamasaki Square technique – top view.

6.1.12. *OCT + Hamasaki Square*

A variant of the miking technique which relies on two other techniques operating concurrently in sync (fig. 138). In this case, OCT provides a so-called base for a stereo recording with the centre channel (L + C + R) (2.1) (discussed in detail on chapter 5.1.18), whereas the spatial effect is created with the Hamasaki Square technique (discussed in detail on chapter 6.1.11)[217].

216 Murphy J. J. (2016). op. cit., p. 212; Friesecke A. (2007). op. cit., pp. 435–436, 79; Owsinski B. (2005). op. cit., p. 217; Kassier R., Lee H.-K., Brookes T., Rumsey F. (2005). op. cit. pp. 5–6.

217 See: https://lossenderosstudio.com/article.php?subject=17 [accessed: 25.06.2022].

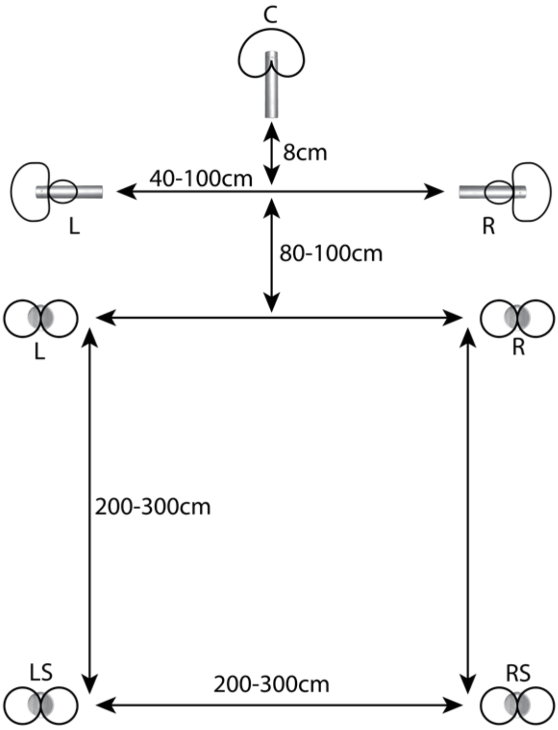


Fig. 138. The layout of microphones in the OCT + Hamasaki Square technique.

6.1.13. *INA 5 (German: Ideale Nieren Anordnung or ICA 5 Ideal Cardioid Arrangement)*

This technique uses five cardioid microphones aimed in the right directions (fig. 139 and 140). Due to the small distances between the devices, Ina 5 is regarded as a practical and convenient method[218].

[218] Friesecke A. (2007). op. cit., p. 435; Rumsey F., McCormick T. (2006). op. cit., pp. 506–507; Weinzierl S. (2008). op. cit., pp. 595–596; Alten S. R. (2012). *Audio basics*, Wadsworth Cengage Learning, Boston, p. 147.

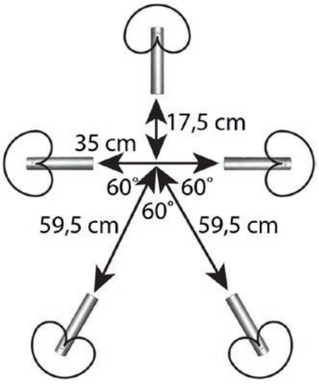


Fig. 139. The layout of microphones in the INA 5 technique – top view.

Fig. 140. Microphone configuration on the stand in INA 5.

6.1.14. *Polyhymnia Pentagon*

It was developed by Polyhymnia International (formerly: Philips Classics). Five omnidirectional microphones are positioned according to the speaker setup defined in Recommendation ITU-R BS775 (5.1 surround sound), where every individual microphone signal is fed to its corresponding loudspeaker, as required (fig. 141). Five microphones are in a circle formation, with the radius adjusted in keeping with the room dimensions. The central microphone and the surround are usually placed at an angle of 100–120 degrees, with 110 degrees being the prevalent choice[219].

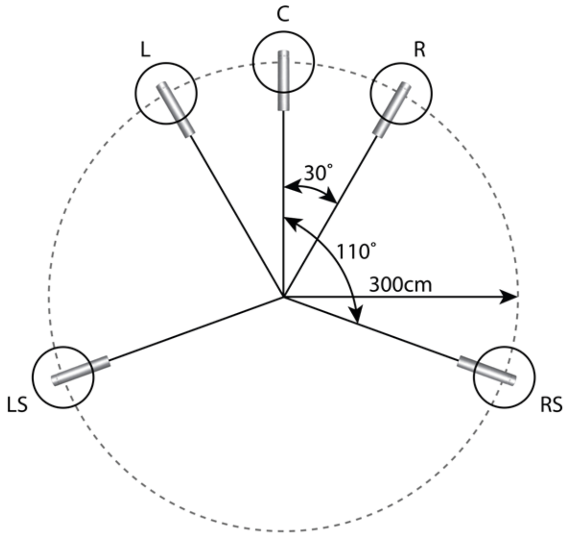


Fig. 141. The layout of microphones in the polyhymnia pentagon technique.

6.1.15. *Hamasaki Surround / NHK Surround*

It was developed in Japan by Kimio Hamasaki, an NHK research engineer. The technique characteristically uses a seven-microphone array and a baffle, with five cardioids and two omni devices (fig. 142). The baffle separates the left and right front channels, spaced 30 cm apart at an angle of 45 degrees. The two omni microphones utilize a low-pass filter set to 250 Hz. The rear cardioids are positioned at 135 degrees. Hamasaki Surround can create an intense sound with a powerful surround effect[220].

219 See: https://lossenderosstudio.com/article.php?subject=17 [accessed: 25.06.2022].
220 Ibidem.

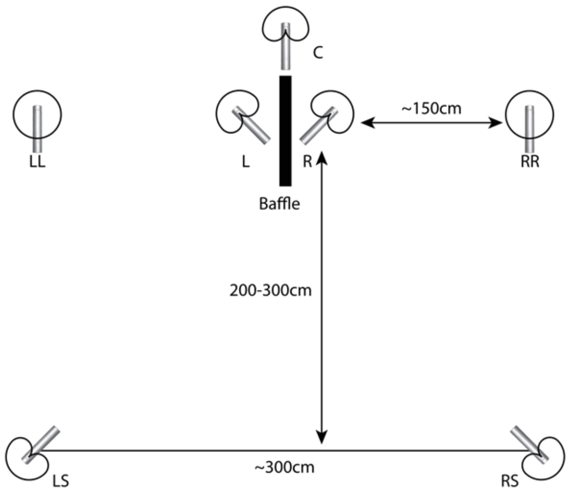


Fig. 142. The layout of microphones in the Hamasaki surround / NHK surround technique.

6.1.16. *Multichannel Microphone Array Design (MMAD)*

Many microphone techniques are based on MMAD. These setups range from quadraphonic to 7.1 surround (and larger, constantly developed systems), often utilizing several overlaid microphone techniques. This book presents only a dozen or so examples, since there are over 2000 (!) different setups applying Multichannel Microphone Array Design (fig. 143–152). Hence, the presentation of all the aspects of this technique is limited to a bare minimum. New variants of the technique can be created by varying:

- the distance between the front triplet microphones,
- the distance between the rear pair,
- the distance between the side pair,
- the distance between the front triplet and the rear pair,
- the angles between the front triplet,
- the angle between the rear pair,
- the angle between the side pair,
- the difference in angle between the front triplet and the back pair,
- the number of microphones used,
- directional characteristics of the microphones,
- the distance to the sound source,

- the size of the sound source (e.g. whether it is a choir, a small ensemble, or a solo performer).

The great number of existing Multichannel Microphone Array Design (MMAD) techniques is the result of minute granularity in the positioning of the microphones and their angles (as little as 2°), which constitute new variants. Some versions of this technique vary greatly, while other variants offer such small differences in sound as to be barely perceptible.

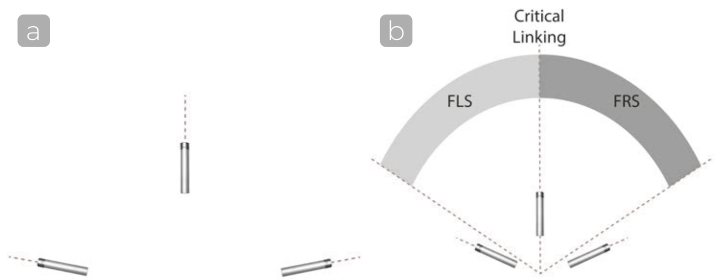


Fig. 143. a) Microphones positioned in a front triplet and b) the central microphone receives sound from the middle, dividing the source into two ostensible parts, while the angle between side microphones is larger than that of the sound source, which facilitates the recording of reflected (ambient) sound, with less intensity.

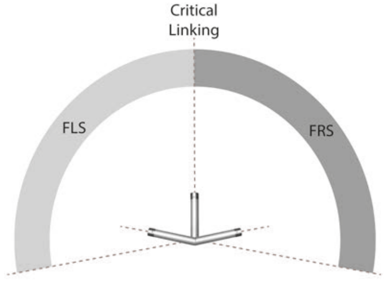


Fig. 144. The Stereophonic Recording Angle is wider than that between microphones, which makes the devices process more direct sound from the source which surrounds the setup, while the central microphone picks up the centre between the ostensible two parts of the source.

Table 11. Positioning variants of Front Triplet Coverage[221]

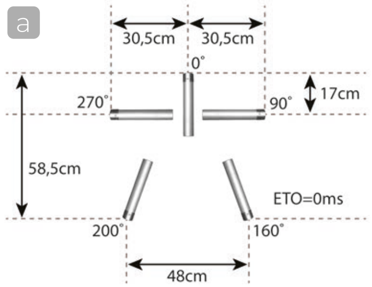
Front Triplet Coverage 72° +72°					
Microphone orientation	Distance between microphones	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage (backwards from the central microphone)	Microphone Position Time Offset (MPTO)	Figure illustrating the technique
90° (R)		+30.5 cm			Figure 145 a)
270° (L)	35 cm	–30.5 cm	17 cm	–15,6°	Figure 145 a)
80° (R)		+31 cm			
280° (L)	37 cm	–31 cm	20.5 cm	–6°	
72° (R)		+31.5 cm			
288° (L)	39 cm	–31.5 cm	23 cm	No offset	
60° (R)		+33 cm			
300° (L)	42.5 cm	–33 cm	26.5 cm	+9°	
50° (R)		+34 cm			
310° (L)	45 cm	–34 cm	29.5 cm	+15.5°	
40° (R)		+36.5 cm			Figure 145 b)
320° (L)	48.5 cm	–36.5 cm	31.5 cm	+20.9°	Figure 145 b)

221 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (2001). "The Quick Reference Guide to Multichannel Microphone Arrays. Part 1: using Cardioid Microphones", *Audio Engineering Society*, Convention Paper 5336 Presented at the 110th Convention 2001 May 12–15 Amsterdam, http://www.aes.org/tmpFiles/elib/20200503/9937.pdf [accessed: 26.06.2022], pp. 4–19.

Table 12. Positioning variants of Back Pair Coverage[222]

Lateral Pairs		Back Pair					
Lateral Segment Coverage	Electronic Time Offset (ETO)	Back Segment Coverage	Microphone orientation	Angle between microphones	Distance between microphones	Distance between the Back Pair microphones	Figure illustrating the technique
			160° (R)				Figure 145 a)
72°	No offset	72°	200° (L)	40°	48 cm	58.5 cm	Figure 145 a)
			155° (R)				
72°	No offset	72°	205° (L)	50°	45 cm	59 cm	
			144° (R)				
72°	No offset	72°	216° (L)	72°	39 cm	60 cm	
			135° (R)				
72°	No offset	72°	225° (L)	90°	34.5 cm	62 cm	
			130° (R)				Figure 145 b)
72°	No offset	72°	230° (L)	100°	32 cm	63.5 cm	Figure 145 b)

222 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.



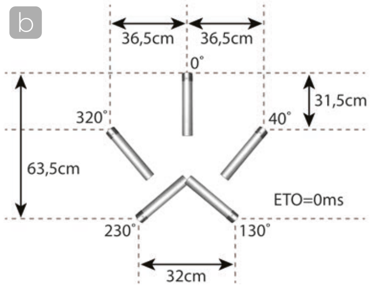


Fig. 145. a) and b) Variants of microphone positioning according to Multichannel Microphone Array Design, as described in tables 11 and 12.

Table 13. Positioning variants of Front Triplet Coverage[223]

Front Triplet Coverage 60° +60°					
Microphone orientation	Distance between microphones	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage (backwards from the central microphone)	Microphone Position Time Offset (MPTO)	Figure illustrating the technique
90° (R)		+42.5 cm			Figure 146 a)
270° (L)	46 cm	–42.5 cm	17 cm	–23°	Figure 146 a)
80° (R)		+43.5 cm			
280° (L)	48 cm	–43.5 cm	20.5 cm	–15°	

223 Williams M., Dû G. L. *Multichannel sound…*; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

72° (R)		+44.5 cm			
288° (L)	50.2 cm	–44.5 cm	23 cm	–7°	
60° (R)		+46 cm			
300° (L)	53 cm	–46 cm	26.5 cm	No offset	
50° (R)		+47.5 cm			
310° (L)	56 cm	–47.5 cm	29.5 cm	+7°	
40° (R)		+50 cm			Figure 146 b)
320° (L)	59 cm	–50 cm	31.5 cm	+12°	Figure 146 b)

Table 14. Positioning variants for the Back Pair[224]

Lateral Pairs		Back Pair					
Lateral Segment Coverage	Electronic Time Offset (ETO)	Back Segment Coverage	Microphone orientation	Angle between microphones	Distance between microphones	Distance between the Back Pair microphones	Figure illustrating the technique
			148° (R)				Figure 146 a)
97.5°	No offset	45°	212° (L)	64°	73 cm	46.5 cm	Figure 146 a)
			138° (R)				

224 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

97.5°	No offset	45°	222° (L)	84°	67.5 cm	48 cm	
			128° (R)				
97.5°	No offset	45°	232° (L)	104°	62 cm	49.5 cm	
			118° (R)				
97.5°	No offset	45°	242° (L)	124°	58 cm	51 cm	
			130° (R)				Figure 146 b)
98°	No offset	44°	230° (L)	144°	54 cm	53 cm	Figure 146 b)

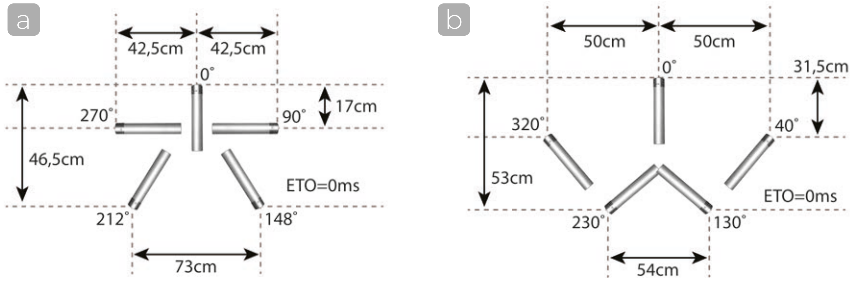


Fig. 146. a) and b) Variants of microphone positioning according to Multichannel Microphone Array Design, described in tables 13 and 14.

Table 15. Positioning variants for Front Triplet Coverage[225]

Front Triplet Coverage 50° + 50°					
Microphone orientation	Distance between microphones	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage (backwards from the central microphone)	Microphone Position Time Offset (MPTO)	Figure illustrating the technique
90° (R)		+58.5 cm			Figure 147 a)
270° (L)	61 cm	–58.5 cm	17 cm	–28.5°	Figure 147 a)
80° (R)		+58 cm			
280° (L)	62 cm	–58 cm	21 cm	–20°	
72° (R)		+59 cm			
288° (L)	63.9 cm	–59 cm	24 cm	–13°	
60° (R)		+61 cm			
300° (L)	66.5 cm	–61 cm	27 cm	–6°	
50° (R)		+63 cm			
310° (L)	69.5 cm	–63 cm	29.5 cm	No offset	
40° (R)		+65 cm			Figure 147 b)
320° (L)	72.5 cm	–65 cm	32 cm	+6°	Figure 147 b)

225 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

Table 16. Positioning variants for the Back Pair[226]

Lateral Pairs		Back Pair					
Lateral Segment Coverage	Electronic Time Offset (ETO)	Back Segment Coverage	Microphone orientation	Angle between microphones	Distance between microphones	Distance between the Back Pair microphones	Figure illustrating the technique
			140° (R)				Figure 147 a) – b)
114°	No offset	32°	220° (L)	80°	103 cm	44 cm	Figure 147 a) – b)

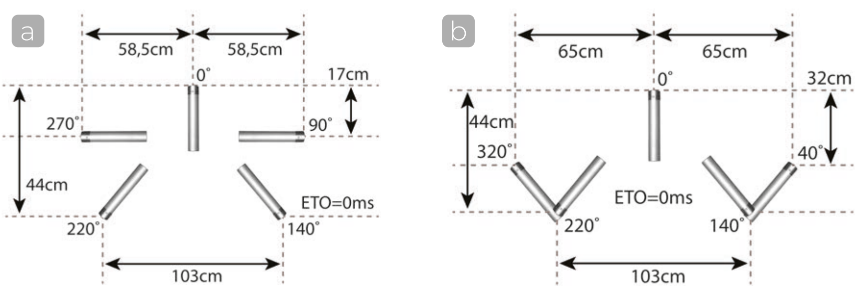


Fig. 147. a) and b) Variants of microphone positioning according to Multichannel Microphone Array Design, described in tables 15 and 16.

226 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

Table 17. Positioning variants for Front Triplet Coverage[227]

Front Triplet Coverage 90° +90°					
Microphone orientation	Distance between microphones	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage (backwards from the central microphone)	Microphone Position Time Offset (MPTO)	Figure illustrating the technique
90° (R)		+17.3 cm			Figure 148 a)
270° (L)	24.5 cm	–17.3 cm	17.3 cm	No offset	Figure 148 a)
80° (R)		+17.5 cm			
280° (L)	27 cm	–17.5 cm	20.5 cm	+9.2°	
72° (R)		+18 cm			
288° (L)	29.5 cm	–18 cm	23.5 cm	+18°	
60° (R)		+19 cm			
300° (L)	32.5 cm	–19 cm	26.5 cm	+24.5°	
50° (R)		+20 cm			
310° (L)	35.5 cm	–20 cm	29 cm	+30.5°	
40° (R)		+22 cm			Figure 148 b)
320° (L)	38.5 cm	–22 cm	31.5 cm	+35.5°	Figure 148 b)

227 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

Table 18. Positioning variants for the Back Pair[228]

Lateral Pairs		Back Pair					
Lateral Segment Coverage	Electronic Time Offset (ETO)	Back Segment Coverage	Microphone orientation	Angle between microphones	Distance between microphones	Distance between the BackPair microphones	Figure illustrating the technique
			140° (R)				Figure 148 a)
45°	–0.98 ms	90°	220° (L)	80°	27 cm	103 cm	Figure 148 a)
			140° (R)				
50°	–0.9 ms	80°	220° (L)	80°	32 cm	94 cm	
			135° (R)				
55°	–0.94 ms	70°	225° (L)	90°	36 cm	90 cm	
			135° (R)				
60°	–1.05 ms	60°	225° (L)	90°	45 cm	86.5 cm	
			130° (R)				
65°	–1.23 ms	50°	230° (L)	100°	55 cm	87.5 cm	
			120° (R)				Figure 148 b)
70°	–1.5 ms	40°	240°(L)	120°	68 cm	98 cm	Figure 148 b)

[228] Williams M., Dû G. L. *Multichannel sound…*; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

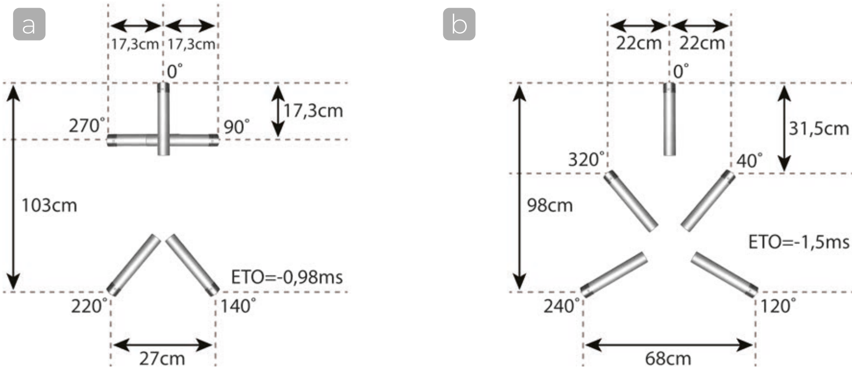


Fig. 148. a) and b) Variants of microphone positioning according to Multichannel Microphone Array Design, described in tables 17 and 18.

Table 19. Positioning variants for Front Triplet Coverage[229]

Front Triplet Coverage 80° +80°					
Microphone orientation	Distance between microphones	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage (backwards from the central microphone)	Microphone Position Time Offset (MPTO)	Figure illustrating the technique
90° (R)		+24 cm			Figure 149 a)
270° (L)	29.5 cm	–24 cm	17.5 cm	–9 °	Figure 149 a)

229 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

80° (R)		+24.5 cm			
280° (L)	32 cm	–24.5 cm	20.5 cm	No offset	
72° (R)		+25 cm			
288° (L)	34.5 cm	–25 cm	23.5 cm	+8°	
60° (R)		+26.5 cm			
300° (L)	37.5 cm	–26.5 cm	26.5 cm	+15°	
50° (R)		+27.5 cm			
310° (L)	40 cm	–27.5 cm	29 cm	+22°	
40° (R)		+29.5 cm			Figure 149 b)
320° (L)	43.5 cm	–29.5 cm	31.5 cm	+27°	Figure 149 b)

Table 20. Positioning variants for the Back Pair[230]

Lateral Pairs		Back Pair					
Lateral Segment Coverage	Electronic Time Offset (ETO)	Back Segment Coverage	Microphone orientation	Angle between microphones	Distance between microphones	Distance between the Back Pair microphones	Figure illustrating the technique
			140° (R)				Figure 149 a)
55°	–0.75 ms	90°	220° (L)	80°	27 cm	85.3 cm	Figure 149 a)
			140° (R)				
60°	–0.7 ms	80°	220° (L)	80°	32 cm	80 cm	

230 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

			135° (R)				
65°	–0.7 ms	70°	225° (L)	90°	36 cm	76.5 cm	
			135° (R)				
70°	–0.6 ms	60°	225° (L)	90°	45 cm	71 cm	
			130° (R)				
75°	–0.7 ms	50°	230° (L)	100°	55 cm	71 cm	
			120° (R)				Figure 149 b)
80°	–0.76 ms	40°	240°(L)	120°	68 cm	76 cm	Figure 149 b)

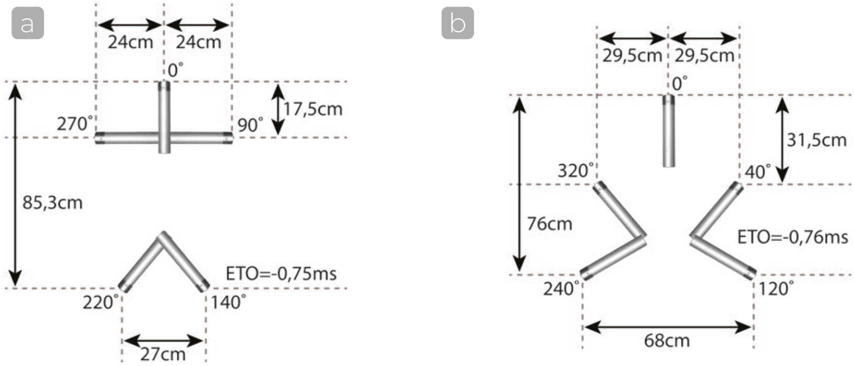


Fig. 149. a) and b) Variants of microphone positioning according to Multichannel

Table 21. Positioning variants for Front Triplet Coverage[231]

Front Triplet Coverage 72° +72°					
Microphone orientation	Distance between microphones	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage (backwards from the central microphone)	Microphone Position Time Offset (MPTO)	Figure illustrating the technique
90° (R)		+30.5 cm			Figure 150 a)
270° (L)	35 cm	–30.5 cm	17 cm	–15.6°	Figure 150 a)
80° (R)		+31 cm			
280° (L)	37 cm	–31 cm	20.5 cm	–6°	
72° (R)		+31.5 cm			
288° (L)	39 cm	–31.5 cm	23 cm	No Offset	
60° (R)		+33 cm			
300° (L)	42.5 cm	–33 cm	26.5 cm	+9°	
50° (R)		+34 cm			
310° (L)	45 cm	–34 cm	29.5 cm	+15.5°	
40° (R)		+36.5 cm			Figure 150 b)
320° (L)	48.5 cm	–36.5 cm	31.5 cm	+20.9°	Figure 150 b)

231 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

Table 22. Positioning variants for the Back Pair[232]

Lateral Pairs		Back Pair					
Lateral Segment Coverage	Electronic Time Offset (ETO)	Back Segment Coverage	Microphone orientation	Angle between microphones	Distance between microphones	Distance between the Back Pair microphones	Figure illustrating the technique
			140° (R)				Figure 150 a)
63°	+0.2 ms	90°	220° (L)	80°	27 cm	67.5 cm	Figure 150 a)
			140° (R)				
68°	+0.07 ms	80°	220° (L)	80°	32 cm	63.5 cm	
			135° (R)				
72°	No offset	70°	225° (L)	90°	39 cm	60 cm	
			135° (R)				
78°	–0.19 ms	60°	225° (L)	90°	45 cm	60.1 cm	
			130° (R)				
83°	–0.36 ms	50°	230° (L)	100°	55 cm	61.1 cm	
			120° (R)				Figure 150 b)
88°	–0.43 ms	40°	240°(L)	120°	68 cm	62 cm	Figure 150 b)

232 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

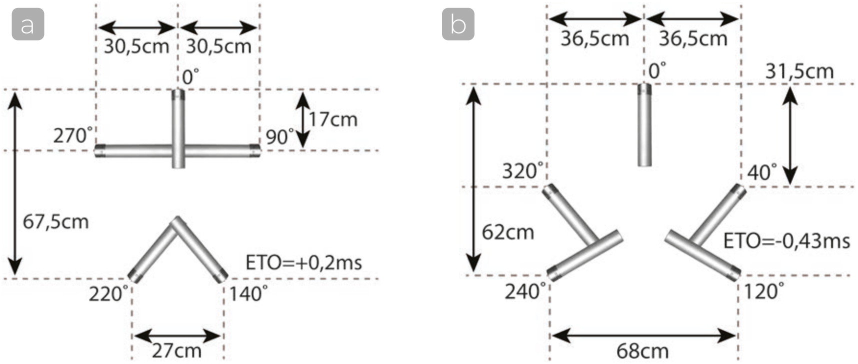


Fig. 150. a) and b) Variants of microphone positioning according to Multichannel Microphone Array Design, described in tables 21 and 22.

Table 23. Positioning variants for Front Triplet Coverage[233]

Front Triplet Coverage 60° +60°					
Microphone orientation	Distance between microphones	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage (backwards from the central microphone)	Microphone Position Time Offset (MPTO)	Figure illustrating the technique
90° (R)		+42.5 cm			Figure 151 a)
270° (L)	46 cm	–42.5 cm	17 cm	–23°	Figure 151 a)
80° (R)		+43.5 cm			
280° (L)	48 cm	–43.5 cm	20.5 cm	–15°	

233 Williams M., Dû G. L. *Multichannel sound...*; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

72° (R)		+44.5 cm			
288° (L)	50.2 cm	–44.5 cm	23 cm	–7°	
60° (R)		+46 cm			
300° (L)	53 cm	–46 cm	26.5 cm	No offset	
50° (R)		+47.5 cm			
310° (L)	56 cm	–47.5 cm	29.5 cm	+7°	
40° (R)		+50 cm			Figure 151 b)
320° (L)	59 cm	–50 cm	31.5 cm	+12°	Figure 151 b)

Table 24. Positioning variants for the Back Pair[234]

Lateral Pairs		Back Pair					
Lateral Segment Coverage	Electronic Time Offset (ETO)	Back Segment Coverage	Microphone orientation	Angle between microphones	Distance between microphones	Distance between the Back Pair microphones	Figure illustrating the technique
			140° (R)				Figure 151 a)
75°	+0.69 ms	90°	220° (L)	80°	27 cm	57 cm	Figure 151 a)
			140° (R)				

234 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

80°	+0.56 ms	80°	220° (L)	80°	32 cm	53 cm	
			135° (R)				
85°	+0.42 ms	70°	225° (L)	90°	36 cm	51 cm	
			135° (R)				
90°	+0.28 ms	60°	225° (L)	90°	45 cm	49 cm	
			130° (R)				
95°	+0.1 ms	50°	230° (L)	100°	55 cm	49 cm	
			120° (R)				Figure 151 b)
100°	–0.1 ms	40°	240°(L)	120°	68 cm	52 cm	Figure 151 b)

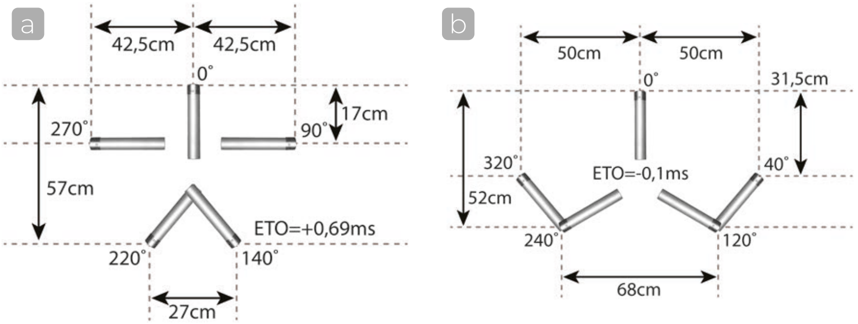


Fig. 151. a) and b) Variants of microphone positioning according to Multichannel Microphone Array Design, described in tables 23 and 24.

Table 25. Positioning variants for Front Triplet Coverage[235]

Front Triplet Coverage 50° + 50°					
Microphone orientation	Distance between microphones	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage	Distance from the central microphone capsule to the left and right microphone capsules in Front Triplet Coverage (backwards from the central microphone)	Microphone Position Time Offset (MPTO)	Figure illustrating the technique
90° (R)		+58.5 cm			Figure 152 a)
270° (L)	61 cm	–58.5 cm	17 cm	–28.5°	Figure 152 a)
80° (R)		+58 cm			
280° (L)	62 cm	–58 cm	21 cm	–20°	
72° (R)		+59 cm			
288° (L)	63.9 cm	–59 cm	24 cm	–13°	
60° (R)		+61 cm			
300° (L)	66.5 cm	–61 cm	27 cm	–6°	
50° (R)		+63 cm			
310° (L)	69.5 cm	–63 cm	29.5 cm	No offset	
40° (R)		+65 cm			Figure 152 b)
320° (L)	72.5 cm	–65 cm	32 cm	+6°	Figure 152 b)

235 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

Table 26. Positioning variants for the Back Pair[236]

Lateral Pairs		Back Pair					
Lateral Segment Coverage	Electronic Time Offset (ETO)	Back Segment Coverage	Microphone orientation	Angle between microphones	Distance between microphones	Distance between the Back Pair microphones	Figure illustrating the technique
			140° (R)				
90°	No solution	80°	220° (L)	80°	32 cm	–	
			135° (R)				
95°	No solution	70°	225° (L)	90°	36 cm	–	
			135° (R)				
100°	No solution	60°	225° (L)	90°	45 cm	–	
			130° (R)				Figure 152 a)
105°	+0.5 ms	50°	230° (L)	100°	55 cm	42.5 cm	Figure 152 a)
			120° (R)				
110°	+0.28 ms	40°	240° (L)	120°	68 cm	45.5 cm	
			140° (R)				Figure 152 b)
114°	No offset	32°	220°(L)	120°	103 cm	52.5 cm	Figure 152 b)

236 Williams M., Dû G. L. *Multichannel sound*...; Williams M., Dû G. L. (2001). op. cit., pp. 4–19.

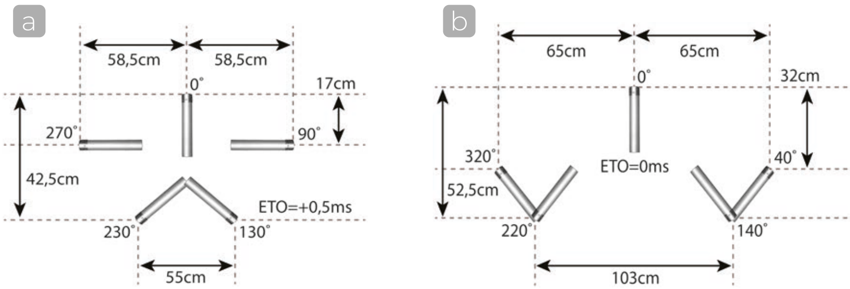


Fig. 152. a) and b) Variants of microphone positioning according to Multichannel Microphone Array Design, described in tables 25 and 26.

6.1.17. *Five channel surround / MMA (based on MMAD)*

A miking technique for surround sound, relying on five cardioids, supercardioids, or hypercardioids, depending on the production objectives and the engineer's taste (fig. 153). When compared to cardioids, super- or hypercardioids allow for an even better separation between speakers and a more precise spatial localization of the sound. One potential drawback of this arrangement is the risk of interrupting the precise sound localization (for instance, the audio may seem abruptly cut from one speaker and fed by another, instead of completing a smooth transition). The recommended distance between microphones ranges from 10 to 150 cm. The liberty in gap adjustment ensures comprehensive space imaging, depending on the requirements and needs of production[237].

[237] Alten S. R. (2011). op. cit., p. 413; Bartlett B., Bartlett J. (2009). op. cit., p. 477; Williams M., Dû G. L. (2000). "Multichannel Microphone Array Design", *Journal Audio Engineering Society*, Convention Paper, Presented at the 108th Convention 2000 February 19–22 Paris, http://www.aes.org/tmpFiles/elib/20200316/9181.pdf [accessed: 25.06.2022], pp. 11–21; Kassier R., Lee H.-K., Brookes T., Rumsey F. (2005). op. cit., p. 5.

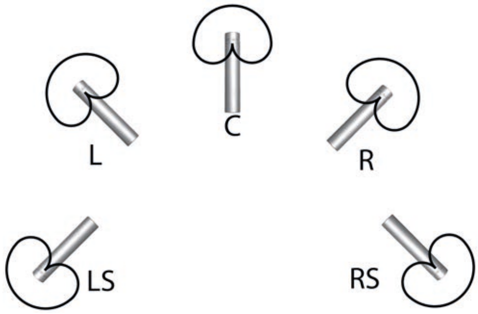


Fig. 153. The layout of microphones in the five-channel surround technique.

6.1.18. *Decca Tree Surround (1) (based on MMAD)*

A surround sound technique which, in the presented variant, uses five omni microphones (fig. 154). The distances from the frontal triplet (Decca Tree) to the sound source are adjusted to ensure optimal reproduction (the wider the sound source, the smaller the distance between devices). The rear surround microphones should not be too far behind the first row to avoid delay which distorts and degrades the sound. Although the certain directionality of the surround microphones may be preferred during surround recordings, the omni devices could interfere with the correct identification of the directionality. The problem can be mitigated with acoustic pressure equalizers (APE), which ensure a directional pattern at high frequencies and maintain omnidirectionality for low frequencies. Owing to the omni devices, Decca Tree Surround can create a balanced sound. Low frequencies are reproduced in a clear and pronounced manner, whereas the medium and high frequencies give the impression of a thick envelopment in sound. The technique has the downside of considerable crosstalk, as the omni devices are not separated with baffles to limit sound radiation[238].

[238] See: https://www.dpamicrophones.com/mic-university/immersive-sound-object-based-audio-and-microphones [accessed: 24.06.2022].

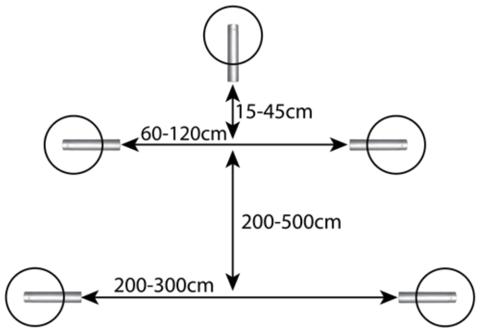


Fig. 154. The layout of microphones in the Decca Tree Surround (1) / Multichannel Microphone Array Design (MMAD) technique.

6.1.19. *Decca Tree Surround (2) (based on MMAD)*

A multichannel configuration which works best with six condenser microphones (fig. 155). The microphones indicated with number 1 (cardioid or supercardioid) form an X-Y stereo pair. Numbers 2 and 3 indicate an A / B configuration – subcardioid devices will be a good choice. Numbers 4 and 5 stand for ambient microphones – possibly subcardioid devices for registering room acoustics. Bear in mind that microphones 1, 2 and 3 focus on direct sound capture. Therefore, proper placement of all microphones is crucial to achieve a full and clear sense of the surrounding space[239].

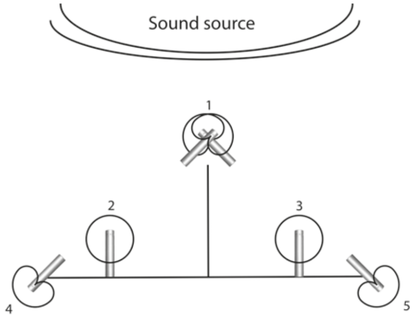


Fig. 155. The layout of microphones in the Decca Tree Surround (2) technique.

[239] Alten S. R. (2012). op. cit., p. 145; Alten S. R. (2011). op. cit., p. 412.

6.1.20. *The Wide Cardioid Surround Array*

Developed by Mikkel Naymand, is a technique which can – depending on the setup – capture or ensure (fig. 156):

- various sound of individual instruments, for instance during an orchestral recording,
- a strong focus on low frequencies,
- immersive spatial effects[240].

The distance between the microphones is not fixed and should be adjusted by the sound engineer. Too narrow spacing of the devices will cause interchannel interference, whereas too wide of a gap causes a non-uniform and incoherent sound. Importantly, the technique uses five identical microphones (the same model, with all devices from the same manufacturer) for a natural, uniform, and coherent sound. If the devices are spaced farther apart, you can use omnidirectional devices to directly sum the sound together with room ambience. The use of cardioids ensures better separation of the arriving sound, allowing to improve the accuracy of localization and front sound imaging while limiting the hum of the room. Surround microphones at the back should face the ceiling to intensify the spatial effects while increasing their distance to the frontal array will augment the sense of immersion during playback[241].

If you record a large ensemble or wish to produce additional spatial effects, you could try adding a factory-made pair of omnidirectional devices. The extra microphones can capture:

- more intense low frequencies,
- a richer sound more precisely localized in space[242].

The extra pair should be panned together with the front L / R microphones to achieve proper quality of the recording (avoid phase shifts of the arriving sound) and an intense, enveloping space[243].

240 See: https://www.dpamicrophones.com/mic-university/immersive-sound-object-based-audio-and-microphones [accessed: 25.06.2022].
241 Ibidem.
242 Ibidem.
243 Ibidem.

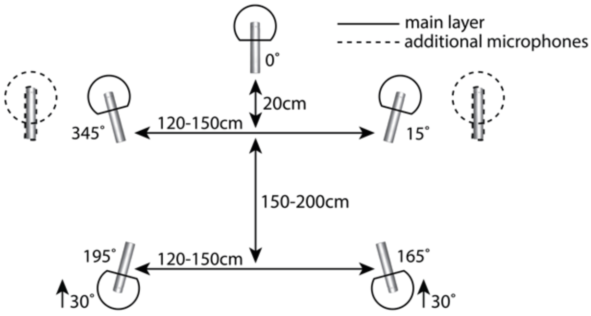


Fig. 156. The layout of microphones in the Wide Cardioid Surround Array technique.

6.1.21. *Fukada Tree*

This technique uses an array of seven microphones (fig. 157). Five devices with a cardioid pickup pattern are pointed in the right directions, whereas two others, having omnidirectional characteristics, serve to elicit an interesting timbre and spatial effect[244].

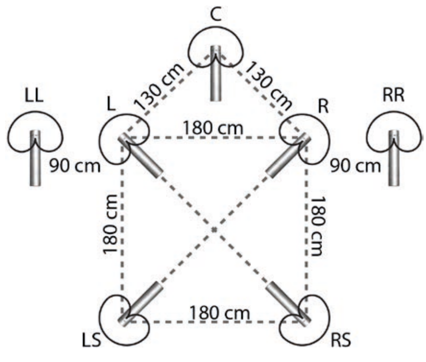


Fig. 157. The layout of microphones in the Fukada Tree technique – top view.

[244] Yun Y., Cho T. (2012). “The Study on Surround Sound Production beyond the Stereo techniques”, in: Kim T.-h. Mohammed S., Ramos C., Abawajy J., Kang B.-H., Ślęzak D (eds.). *Computer Applications for Web, Human Computer Interaction, Signal and Image Processing, and Pattern Recognition*, Springer Verlag, Berlin – Heidelberg, pp. 136–137; Rumsey F., McCormick T. (2009). op. cit., p. 511.

6.1.22. *Corey / Martin Tree*

A miking technique conceived by Jason Corey and Geoff Martin, which uses the total of five microphones: three subcardioids pointed to the front (the sound source) and two cardioids additionally facing the ceiling. The distances presented in the figure 158 may be adjusted depending on the ensemble size. The gap of 120 cm between front microphones (L and R) suits smaller ensembles, whereas 180 cm – larger groups. However, bear in mind that a wider gap may cause the incoherence of the three front channels. To remedy the problem of growing phase shifts, sound engineers must use their skills to strike the right length. The Corey / Martin Tree technique can create an intense sound with a powerful surround effect. It also has a variant called **Wide Cardioid Surround Array** (**WCSA**), invented by Mikkel Nymand, which uses only cardioid microphones[245].

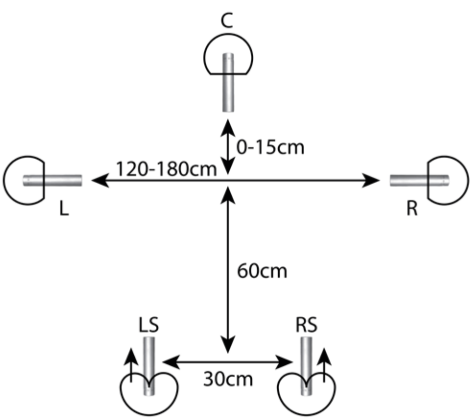


Fig. 158. The layout of microphones in the Corey / Martin tree technique.

[245] See: https://lossenderosstudio.com/article.php?subject=17 [accessed: 25.06.2022].

6.1.23. *A basic approach to direct / ambient surround*

A configuration of microphones in multichannel recordings. It may use an array of either five cardioid microphones or three cardioid and two omnidirectional microphones (fig. 159). The first three cardioid devices (left, central, and right), pointed towards the sound source, register direct sound. In contrast, two microphones placed in the rear (with cardioid or omnidirectional pickup pattern, depending on the variant) are ambient devices – left and right surround – which capture room acoustics. Bear in mind that the audio material should be mixed so that the direct sound together with the ambience portray the full space in the recording[246].

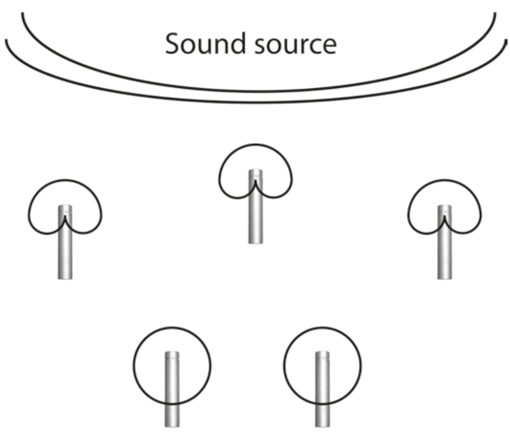


Fig. 159. The layout of microphones in the Basic Approach to Direct / Ambient Surround.

6.1.24. *Delor VR (Virtual Reality)*

A six-microphone technique for surround recordings, considered a variant of the Decca Tree method (fig. 160). Two omnidirectional microphones (left and right) are positioned to the front, while an ORTF pair or a similar configuration features in the centre. With the

[246] Alten S. R. (2012). op. cit., p. 145; Alten S. R. (2011). op. cit., p. 412.

frontal placement of the omni devices, room acoustics are largely captured by the front row. Surround microphones located at the back (usually at the end of the room) exhibit omnidirectional or cardioid directionality and serve to register the ambience. Note that the distance between the first and the second row of microphones depends on the size of the room. Furthermore, all the microphones must be properly placed to obtain a full and simultaneously clear sense of the surrounding space[247].

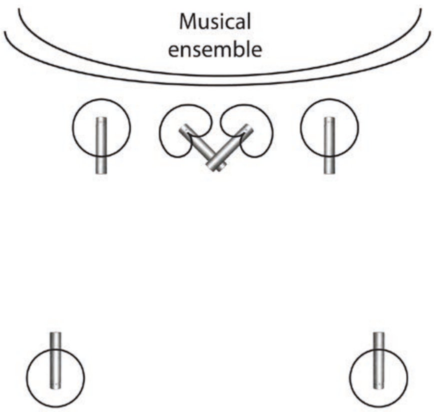


Fig. 160. The layout of microphones in the Delor VR technique.

6.1.25. *DMP (Digital Music Production)*

A technique for surround recordings which uses five microphones, usually cardioid ones, and resembles the Decca Tree method (fig. 161). The first two microphones (left, central, right) face the sound source and register chiefly the incoming direct sound. In turn, the two microphones at the back are surround, ambient devices for registering room acoustics, usually arranged in the ORTF configuration or a similar technique[248].

[247] Alten S. R. (2011). op. cit., p. 412.
[248] Ibidem, p. 413.

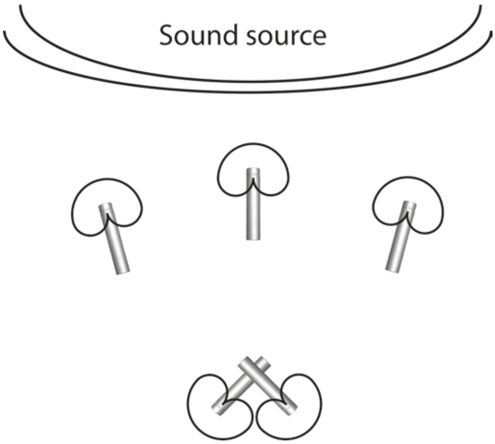


Fig. 161. The layout of microphones in the DMP technique.

6.1.26. *NHK (named after the Japanese Broadcasting Corporation)*

A seven-microphone configuration used in multichannel recordings (fig. 162). One of the cardioid microphones is placed in the centre, at the closest distance to the sound source. The next pair of omnidirectional microphones, left and right, are spaced wide relative to the performer in order to register both the source and the ambience. The other two microphones form a stereo pair, commonly in the X-Y configuration. Finally, surround microphones with cardioid characteristics feature in the rear. The entire array must be positioned with utmost precision (due to phase issues likely to occur at large distances) to achieve a full and clear sense of the surrounding space[249].

[249] Ibidem, p. 413; Bartlett B., Bartlett J. (1999). *On Location Recording Techniques*, Focal Press, Butterworth-Heinemann, p. 183; Bartlett B., Bartlett J. (2009). op. cit., p. 472.

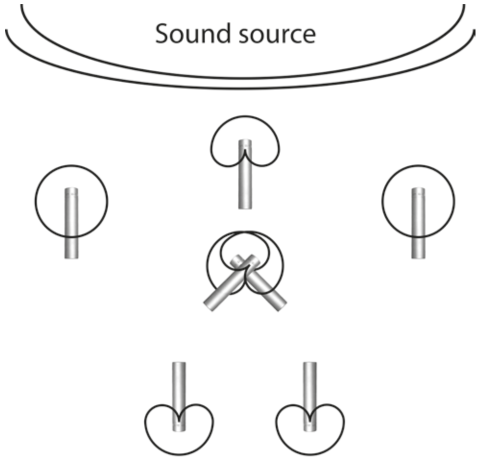


Fig. 162. The layout of microphones in the NHK technique.

6.1.27. *Woszczyk (named after Wiesław Woszczyk, founder of the technique)*

A multichannel configuration. The two front microphones are compact devices with a small-diaphragm transducer. They are commonly referred to as "pins" since they come equipped with a special clamp which can be attached to various surfaces. These omnidirectional to two devices are fastened flat, discrete acoustic baffles often positioned at different angles in line with the acoustic conditions in the room. Two rear microphones, used for creating the surround effect, are a pair of cardioid devices placed at an angle of 180 degrees relative to each other (fig. 163). Remember that rear microphones have reverse poles which will have to be adjusted on a mixing console or sequencer software[250] .

[250] Alten S. R. (2011). op. cit., p. 413; Bartlett B., Bartlett J. (2009). op. cit., pp. 475–477; Bartlett B., Bartlett J. (2007). op. cit., pp. 149–150.

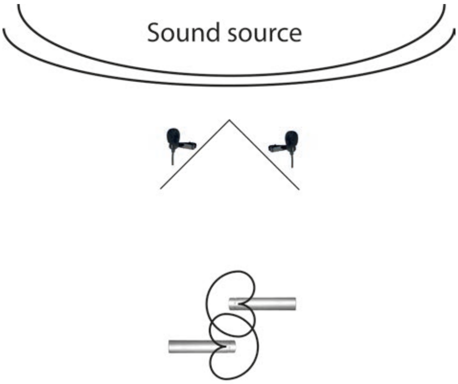


Fig. 163. The layout of microphones in the Woszczyk technique.

6.1.28. *Omni +8 surround*

A variant of the stereo technique called Omni +8 (discussed in detail on chapter 5.1.19), supplemented with two omni devices for spatial effects (fig. 164). The extra pair is located several meters ahead of the front Omni +8 pair, depending on the room size. If the surround microphones capture too much reflected sound (ambience), they may be replaced with devices exhibiting a figure-8 or cardioid pickup pattern[251].

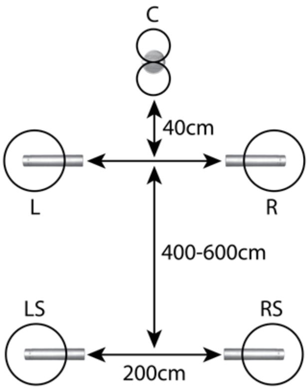


Fig. 164. The layout of microphones in the omni +8 surround technique.

[251] See: https://lossenderosstudio.com/article.php?subject=17 [accessed: 25.06.2022].

6.1.29. *Cardioid Trapeze / Trapezium, Theile Trapezoid*

It was developed by Günther Theile. The technique uses four cardioid microphones facing backwards and positioned at the corners of a trapeze, at an angle of 60 degrees relative to each other and a distance of 60 cm (fig. 165). This design was conceived specifically for pop concerts held live in Great Britain to avoid lateral reflections and create an ambient sound including the singing and clapping of the audience. The cardioid trapeze is often mixed with another technique used simultaneously onstage. The combination of both techniques aims to give a feeling of real participation in the concert when the material is played at home or elsewhere[252].

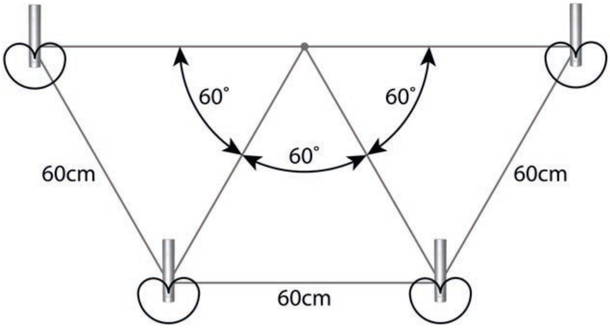


Fig. 165. The layout of microphones in the Cardioid Trapeze / Trapezium, Theile Trapezoid technique.

6.1.30. *Trinnov array*

A surround miking technique based on ambisonic methods. It uses eight omni-microphones (testing on devices exhibiting other pickup patterns is underway) whose arrangement follows from mathematical calculations (fig. 166). The use of Fourier-Bassel functions concerning spherical harmonics allows synthesizing microphone pickup patterns. A specifically designed Trinnov array stand has

252 Ibidem.

the dimensions of 20 x 25 cm and can house eight devices. Such positioning can ensure 360° surround sound and the conversion of recorded signals to stereo and mono. Additionally, the directional attenuation and the coverage (pickup) angle can be adjusted at will, which eliminates the problem of excessive reverberation in the room[253].

Fig. 166. The layout of microphones in the Trinnov array technique.

6.1.31. *Rosiński 7.1*

A system for surround sound recordings which uses seven microphones exhibiting cardioid, supercardioid, hypercardioid, and/or omnidirectional pickup patterns. The devices can be combined at will to achieve a technically correct recording (in phase, no comb filtration, etc.) and a sound image consistent with the sound engineer's intention. The system was established based on a combination and newly-conceived variants of the following techniques: IRT Cross, Double ORTF, OCT, INA 3, Decca tree / ABC stereo. Its structure is presented in the figures 167 a) and b).

[253] Wuttke J. (2005). op. cit., pp. 9–10.

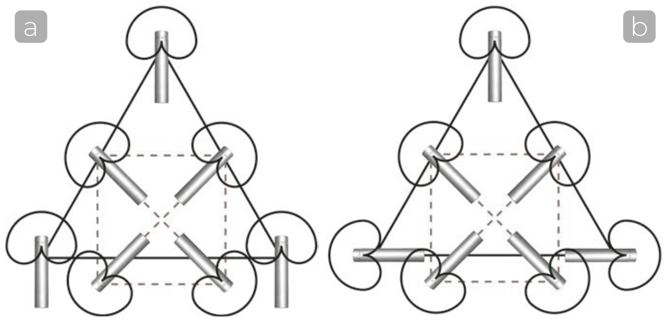


Fig. 167. a) The layout of microphones in the Rosiński 7.1 technique and b) An alternative layout of microphones in Rosiński 7.1.

6.1.32. *Rosiński's Hexagon*

A recording technique for surround sound developed by the author based on Günther Theile's Cardioid Trapeze (fig. 168 a) and b)). The first version uses seven microphones (instead of four), including six cardioids and one omni. The other version assumes that the single omni device in the centre may be replaced by a Blumlein pair, so it requires eight microphones. As an additional departure from the Cardioid Trapeze setup, the devices point in different directions. The six triangles inscribed in the hexagon's area should be equilateral, while the distances between devices should be adjusted to achieve a sound image consistent with the sound engineer's intention.

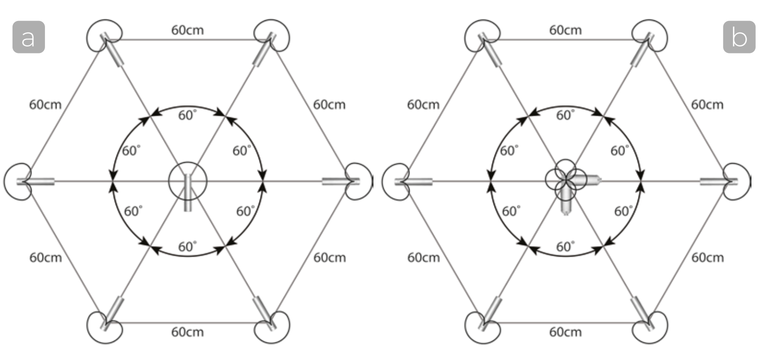


Fig. 168. a) The layout of microphones in the Rosiński's Hexagon technique and b) An alternative layout of microphones in Rosiński's Hexagon.

6.1.33. *Rosiński's Decagon*

A recording technique for surround sound developed by the author based on a Polyhymnia Pentagon conceived by Polyhymnia International. The technique involves eleven microphones (compared to five used in a Polyhymnia Pentagon), including ten cardioids and one omni in the centre of a circle (fig. 169). Rosiński's Decagon builds upon the loudspeaker layout recommended for the 5.1 system, with the additional microphones mirroring the ITU-R BS775 layout for the 5.1 system rotated by 180°. Thus, the decagon offers also a special microphone layout for the 10.2 system. The angles between devices follow the ITU-R BS775 recommendation, whereas the distances should be adjusted to achieve a sound image consistent with the sound engineer's intention.

An interesting solution – and a new variation on the method – presents itself with a layered distribution of microphones comprising two levels. In this case, the first layer would mirror the speaker layout set out in the ITU-R BS775 recommendation, while the other would consist of the ITU-R BS775 setting rotated by 180° and placed far higher, with the microphones mounted under the ceiling. The capsules of the microphones from different layers should be spaced 1 m apart at a minimum.

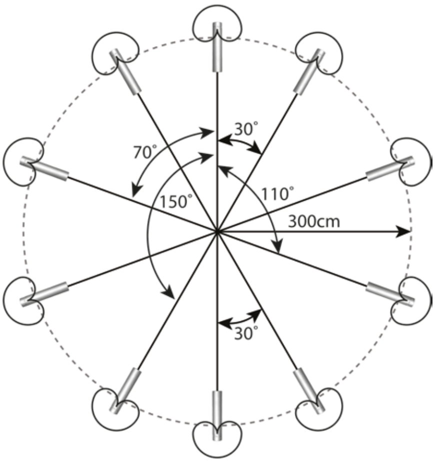


Fig. 169. The layout of microphones in the Rosiński's Decagon technique.

6.2. Dummy head recording techniques for surround sound

6.2.1. *Five channel*

A technique using an array of devices with varied characteristics. A cardioid is placed in the centre, while supercardioids or hypercardioids feature on the left and on the right. The rear is occupied by a dummy head for binaural recordings (fig. 170). The surround sound is produced by the microphones installed inside the dummy head. To achieve a full and pronounced impression of the surrounding space, especially considering the use of the dummy head, positioning all microphones in the correct manner is of crucial importance[254].

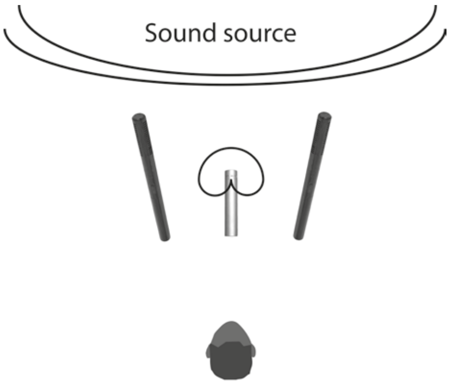


Fig. 170. The layout of microphones in the Five channel technique.

6.2.2. *Multichannel recording of a jazz band*

Uses a dummy head called Schoeps KFM360 which consists of a system of microphones (fig. 171). Two omnidirectional microphones are located on opposite sides of the sphere. Two other microphones, both having a figure-8 pickup pattern, feature next to the former

[254] Alten S. R. (2011). op. cit., p. 413; Bartlett B., Bartlett J. (2007). op. cit., pp. 147–148; Bartlett B., Bartlett J. (1999). op. cit., pp. 185–186.

pair. Mixing with the dummy head involves a skilful adjustment of signal levels coming from the front and the back of the device. The sound from omni and the "front" figure-8 devices are summed to achieve a satisfactory sound, whereas the sounds reaching the dummy head from the back serve to create surround channels[255].

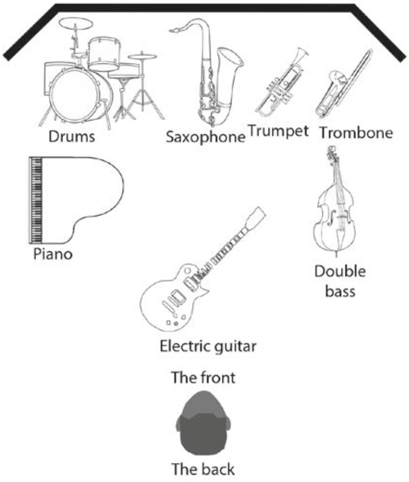


Fig. 171. The layout of microphones in a dummy head configuration during a jazz band recording (a binaural microphone).

6.2.3. *Klepko*

A recording technique for surround sound, which owes its name to its founder, John Klepko. It comprises five microphones, including two omni devices in the rear, separated with a dummy head (or alternatively placed inside the head) (fig. 172). The placement of the diaphragms mirrors the position of the eardrums. Together with the auditory canals leading outside, the entire mechanism is designed to resemble that of a human ear.

The other line of microphones, positioned at the front, includes a central cardioid which picks up the sound from the front and two lateral supercardioids or hypercardioids angled 30° or more to the left or right, accordingly. Although the technique envelops the

[255] Alten S. R. (2011). op. cit., p. 417.

listener in a consistent sound field, it is fraught with other problems, such as lacking interchannel separation, weak localization of the sound source, and a range of complications related to comb filtration produced by the use of the dummy head or the entire bust[256].

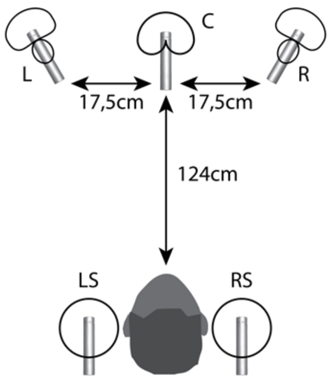


Fig. 172. The layout of microphones in the Klepko technique.

6.2.4. *Double MS + dummy head*

The Double MS technique complete with an additional specialized microphone in the form of a dummy head (fig. 173). The dummy head is pointed at the sound source, while the Double MS technique – to the sides. The reflected sound and time-arrival differences add up to create a powerful spatial effect It integrates the pros and cons of both Double MS and dummy head use. The combination reflects the astounding spatial effects of audio recorded in 3D. The listener feels enveloped in the sound coming from all directions, as if he or she were among the band members in the middle of a performance.

[256] Bartlett B., Bartlett J. (2007). op. cit., pp. 147–148; Klepko J. (1999). *5-Channel Microphone Array with Binaural-Head for Multichannel Reproduction*, McGill University, Montreal, Faculty of Music, an unpublished doctoral dissertation, https://www.collectionscanada.gc.ca/obj/s4/f2/dsk1/tape9/PQDD_0016/NQ55349.pdf [accessed: 24.06.2022], pp. 148, 150–152.

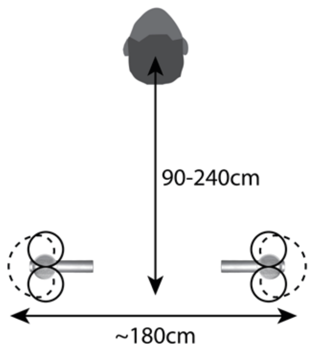


Fig. 173. The layout of microphones in the Double MS + dummy head technique.

6.2.5. *Creating your own dummy head*

Binaural recordings can be made without a specialized dummy-head microphone. To do that, separate two microphones with a baffle which will mimic the dummy head and dampen the audio reaching the left and the right device (fig. 174). The shape and size of the baffle may vary depending on the available resources. Recordings completed in this setup resemble binaural productions. The quality hinges on the selected directionality characteristics of the microphones, the size, shape, and material of the baffle and panning. In this type of production, device parameters and panning settings depend entirely on the final result desired by the sound director, so no specific recommendations are provided.

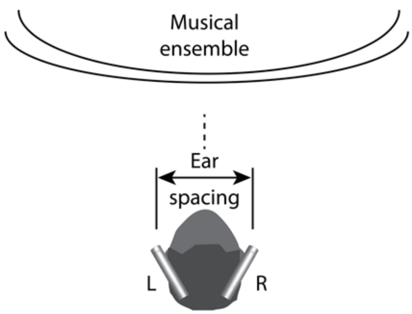


Fig. 174. A potential layout of two traditional microphones separated by a sphere mimicking a dummy head, which dampens the audio arriving at both microphones.

6.3. Multi-layer recording techniques for surround sound

6.3.1. *Immersive audio with height*

It comprises a group of recordings which are particularly tricky due to the knowledge gap regarding the identification of height (not sound) and the sound source. Spatial systems such as 5.1 or 7.1 offer sound perception on the left-right and the front-back axes. Immersive audio with height adds playback speakers (and thus also microphones for recording) elevated above the traditional surround technique (fig. 175 a) – c)). Research is underway to employ this technology in future cinemas, VR, and 3D[257].

An example variant of immersive audio with height, presented by Hyunkook Lee, consists of two recording planes:

- seven microphones placed on stands, with four cardioids and three super-/hypercardioids,
- seven microphones placed high on stands or mounted under the ceiling; all should exhibit a cardioid pickup pattern[258].

The configuration of both planes relies on the following principle: the "primary" sound is registered by the lower array, while the intensity of the signal reaching the upper plane exhibits far lesser intensity. It is recommended to reduce the sensitivity of the upper plane by 7 dB so that it picks up as little direct sound as possible, focusing on the reflections. The elevation gap between devices in the lower and upper plane should be 1 m or broader[259].

Research has proven that audio localization is based on the first arriving sound, but it fails on the vertical plane. When the same audio was played on speakers located at a higher and lower elevation, the listeners focused on the upper device. For that reason, reducing the amplitude of receivers placed under the ceiling is vital to eliminate the emerging artefacts. The listener interpreting sound direction gives "priority" to higher sounds, which makes panning the audio for proper identification and interpretation extremely difficult. Recordings of this type are the subject of multiple studies

[257] See: https://www.dpamicrophones.com/mic-university/immersive-sound-object-based-audio-and-microphones [accessed: 26.06.2022].
[258] Ibidem.
[259] Ibidem.

aiming to broaden knowledge of the spatial localization of sound with an additional layer provided by the upper microphone array[260].

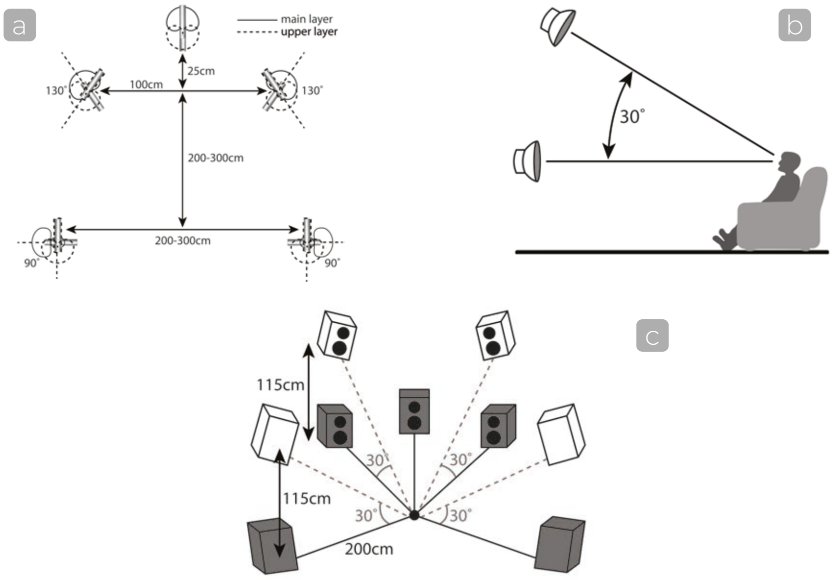


Fig. 175. a) The layout of microphones in the Immersive audio with height technique, b) The angle between first- and second-layer speakers relative to the listener and c) Multichannel speaker system positioning in immersive audio with height.

6.3.2. *Wide a / b*

A modern surround technique using nine omni devices (fig. 176). The design consists of two planes:

- five microphones positioned on stands at an elevation of 0.5–2 m,
- four microphones mounted under the ceiling or on special long stands placed at the elevation of +1 m or higher than the first-plane microphones[261].

260 Ibidem.

261 Theile G., Wittek H. (2011). *Principles in Surround Recordings with Height*, http://www.linkwitzlab.com/Links/AURO3D_Theile-Wittek_Dec_2011.pdf [accessed: 25.06.2022].

This technique creates powerful and impressive directional imaging. Though unfaithful in its recreation of the real-world auditory experience, it is often desired in games, cinemas, and VR technologies. When compared to other surround systems, Wide a / b has characteristically exaggerated directionality. Consequently, the sound is spectacular, albeit loosely related to human hearing and the actual human perception of the world. The technique finds its use in commercial recordings of film music and classical music. Wide a / b is used for the playback of channels in 9.1 systems in the Auro-3D format. The technique is currently being perfected and upgraded to ensure an even better recreation of spatial effects, which could be presented, for instance, in the cinemas of the future to immerse the viewer in sound[262].

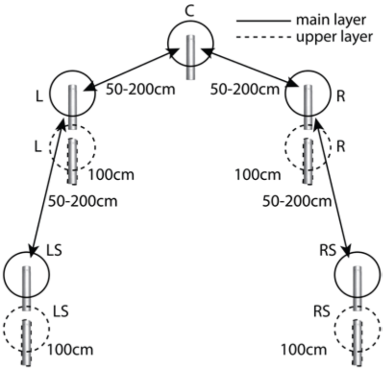


Fig. 176. The layout of microphones in the wide a / b technique.

6.3.3. *OCT 9*

A modern surround technique based on the ORTF setup and using nine microphones of varied pickup patterns. The design consists of two planes (fig. 177):

- five microphones on stands at an elevation of 0.5–2 m. The central microphone and the Left and Right Surround are cardioids, whereas the Left and Right channels are super- or hypercardioids,

262 Ibidem.

- four microphones mounted under the ceiling or on special long stands placed at the elevation of +1 m or higher than the first-plane microphone capsules. Each microphone of this plane is a supercardioid device facing upward (pointed to the ceiling)[263].

OCT 9 offers a characteristically clear directional imaging, resembling the natural sense of space. This technique is used in recordings focused on real directional imaging (such as VR technologies or the cinema). Typical applications include:

- chamber music,
- small ensembles,
- movies (such as drama, SF, horror, thriller),
- sporting events (a football match, tennis court, etc.),
- recordings made in natural surroundings (sounds of the meadow, the forest, etc.)[264].

Pros of the method include:

- low amount of crosstalk,
- balanced localization at the front,
- no direct sound in the rear and upper channels, creating an intense sense of space,
- no phase issues concerning the arriving sounds[265].

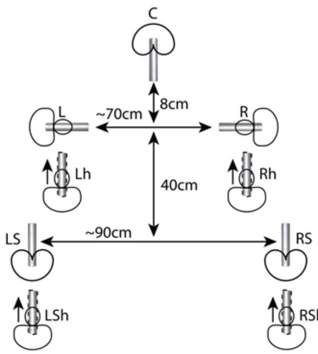


Fig. 177. The layout of microphones in the OCT 9 technique.

[263] Ibidem.
[264] Ibidem.
[265] Ibidem.

6.3.4. *The Hyunkook Lee mixed technique*

It comprises microphone arrays distributed across two layers (fig. 178–179). The first layer mounted 300 cm from the ground is the traditional Hamasaki Square technique (described on chapter 6.1.11). The other, located at least 1 m higher than the former, uses four cardioids oriented upward (to the ceiling). The setup is 10 m from the sound source, with ICA 3 located at 2.5 m[266].

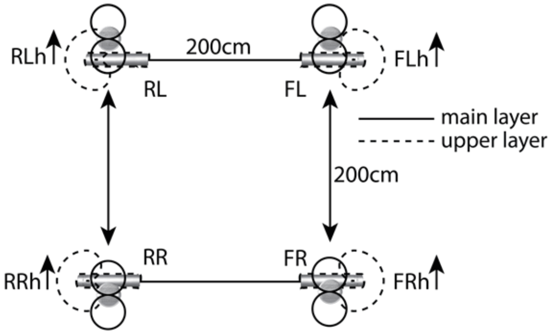


Fig. 178. Microphone positioning in the Hyunkook Lee mixed technique – top view.

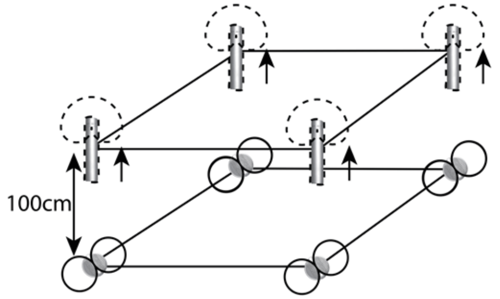


Fig. 179. Microphone positioning in the Hyunkook Lee mixed technique – side view.

266 Lee H. (2015). "2D-to-3D Ambience Upmixing Based on Perceptual Band Allocation", *Journal Audio Engineering Society*, Vol. 63, Issue 10, http://www.aes.org/tmpFiles/elib/20200316/18044.pdf [accessed: 26.06.2022], pp. 812–813.

6.3.5. *The Janet Grab mixed 22.2 technique*

A microphone setup designed to suit specific acoustic (engineering) goals. The system was conceived as a combination of several unrelated miking techniques: mono, Fukada, two Hamasaki techniques, and two examples of a variation on IRT atmo-cross. As a multilayer technique (comprising five layers), it uses 24 microphones positioned on several levels. The devices exhibit omnidirectional, cardioid, supercardioid, and figure-8 pickup patterns. All the data necessary to reproduce the technique are presented in the figure 180.[267].

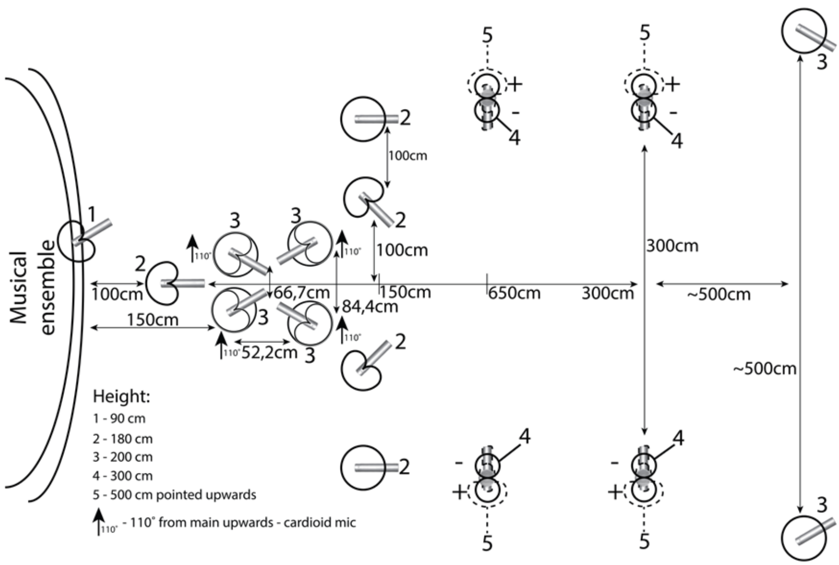


Fig. 180. The layout of microphones in the Janet Grab mixed 22.2 technique.

[267] Grab J. (2019). "Capturing 3D Audio: A pilot study on the spatial and timbral auditory perception of 3D recordings using main-array and front-rear separation in diffuse field conditions", in: 5th *International Conference on Spatial Audio*, September 26th to 28th, Ilmenau, https://www.db-thueringen.de/servlets/MCRFileNodeServlet/dbt_derivate_00045910/ilm1-2019200492_009-016.pdf [accessed: 25.06.2022], pp. 9–15.

6.4. Recording large ensembles

Large ensembles with a high number of musicians include symphony orchestras which perform in churches, philharmonic halls, and stages. A symphony orchestra has a varying lineup, as the number of instruments is subject to change. Therefore, there is no single technique or a set of techniques which would guarantee a successful orchestral production and quality sound in any circumstances. The sound engineer should accommodate the microphone setup to room conditions, ensemble size, the presence or the lack of the audience, and other aspects which have a bearing on the recording procedure. Remember to record instruments in groups – such as first violins, second violins, altos, cellos, double basses – rather than individually. If you recorded every instrument on its own and from close up, the resulting material would be useless for mixing, sounding like a few dozen separate performances instead of an orchestra. If an instrument performs a solo part, it should have an additional "spot" microphone activated and deactivated by the sound engineer at the right moments as the solo begins and comes to a close (the same applies to vocals). Microphones for particular instrumental groups should be located at a distance to ensure that the sound has the right tonal balance and the arriving signal is a sum of the naturally overlapping waves of instruments included in the group (fig. 181–184).

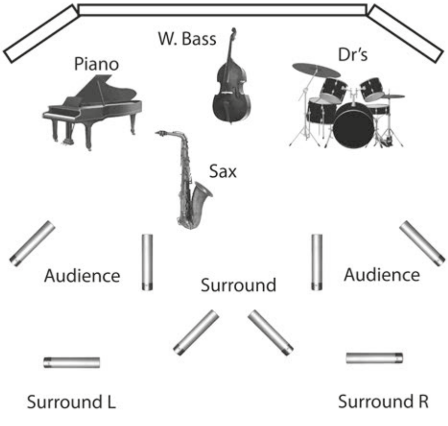


Fig. 181. Surround recording of a jazz ensemble in the NHK Japan Broadcast technique.

Some instruments, such as the piano or the timpani, may be recorded in stereo because these productions sound very well on CDs. The other instrumental groups may be captured in mono (although it is not a must). Having recorded the necessary groups, the engineer positions them in space during panning, which involves the arrangement of audio between the left and the right speaker to create a stereo recording or another production.

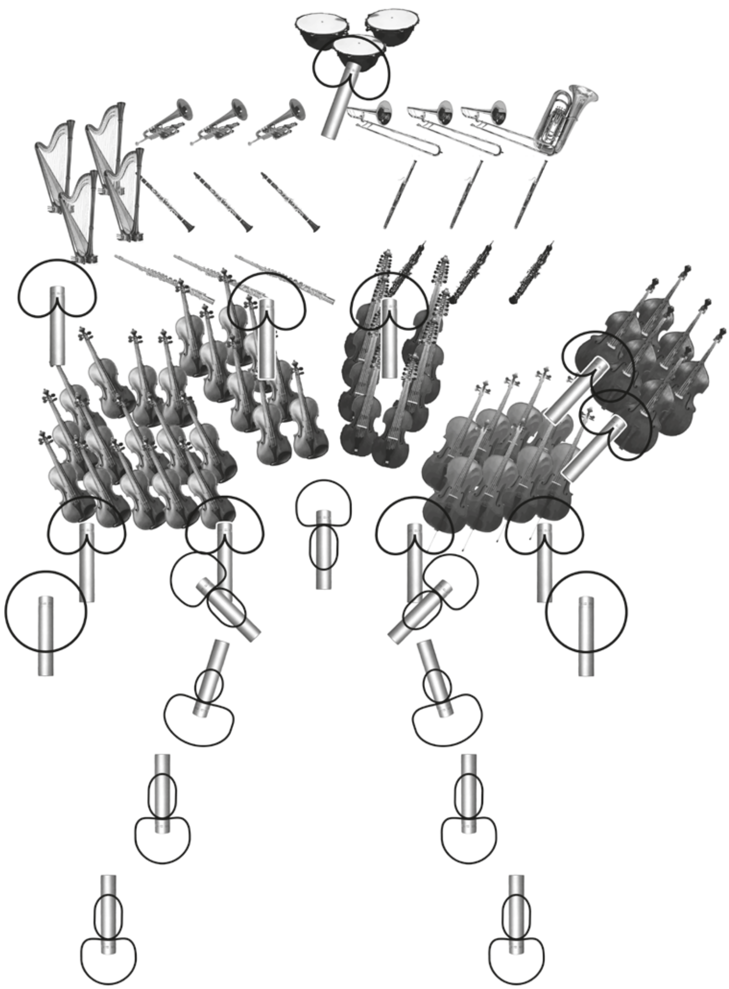


Fig. 182. Surround recording of a symphonic orchestra according in the NHK Japan Broadcast technique.

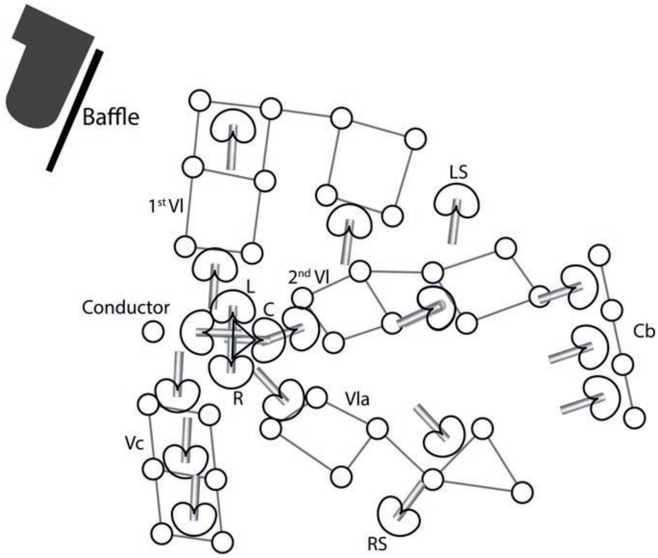


Fig. 183. A surround recording variant of a symphony orchestra in a recording studio.

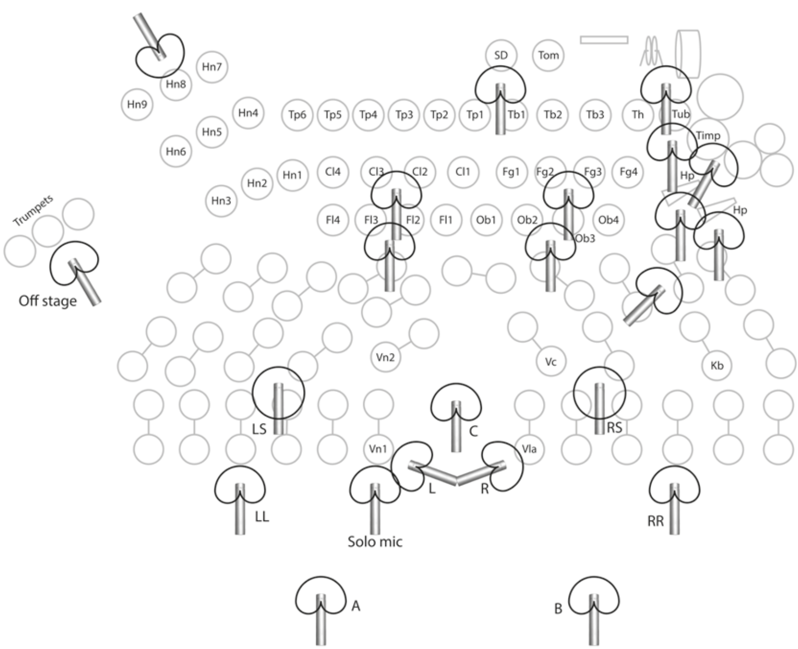


Fig. 184. A surround recording variant of a symphony orchestra in a philharmonic concert hall.

Conclusions

The author aims to present the production of audio recordings in a novel way – from the musician's perspective – which involves the search for new miking techniques and microphone applications. Original ideas give rise to new surround and ambisonic setups, shaping the overall sound of the musical work. Another goal was to present the implementation of the knowledge in practice, as it can be used by sound technicians, engineers, specialists and directors to create an illustrative and audio layer of the music tracks performed by individual instruments, soloists and large or small ensembles.

Continued effort to improve measurement methods and the ever-growing quality of electroacoustic devices may reveal that our seemingly final discoveries conceal many additional aspects which remain underused due to the current state of the art. The new demand for multichannel audio productions proves that sound engineering and the established miking techniques are an area like any other in that they can be developed, modified and polished. The subject – so vast, elaborate, and multi-sided – is still the field of experimentation and pursuit for new sound engineering methods and techniques, such as the use of innovative electroacoustic devices in the 22.2, 3D, and multilayer recording systems, etc.

The advancing technology, although it will never replace the well-tested, traditional recording techniques, will considerably complement the current body of knowledge. Fulfilment of the new needs of the 4K and 8K cinemas, virtual reality, and computer games establishes new norms whose methodical implementation fosters the enhancement of standards for surround sound.

The application of new skills in the sound director's work and the closure of knowledge gaps will significantly broaden the competence of the personnel employed in recording productions. Specialists with such comprehensive training are in high demand in the present-day job market. The growing awareness and expert knowledge encourage the use of modern tools for surround sound in a continuously broader scope, as well as the exchange of experience among professionals in the field.

The author hopes that the monograph will contribute to a better understanding of the sound engineering methods and techniques used in various types of recordings. It is intended to introduce sound engineering to people with little exposure to the topic, to increase awareness and to foster a deeper interest in the exploration of various miking techniques which could present the recorded sound in an innovative manner related to surround sound playback.

The author hopes that an insight into the multitude of recording-related problems allows transforming all musical nuances, which reflect the main idea of the composer or the conductor during a live performance, into an analog or digital phonogram which will become a high-quality **work of art**. With this approach, new productions would offer excellent, hitherto unavailable perceptual effects exploiting multichannel sound and would simultaneously open the listener to a virtual sound space.

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Sound engineering is one of the fastest-growing branches of music production. The need for a broad-based discussion on the issues constituting the art of sound engineering persists and loses none of its relevance, revealing that sound engineering should not be investigated only in the mathematical and physical context (musical acoustics) or the engineering aspect (signal processing and modification).

Publications targeted primarily at musicians are few and far between, which is why the mutual understanding for different priorities which effectively concern the same issues faced by the engineer, the acoustician and the musician, seems to be a complex problem and the main concept explored in this publication.

This book is intended for musicians or sound directors, but also acousticians and sound engineers wishing to learn how the musicians think. The monograph is also addressed to musicians who intend to record their material in the studio in the near future, but do not possess knowledge on studio construction, studio workflow or the art of recording. It seems important to familiarize the musicians with the reality that awaits them on the other side of the glass, thus fostering their responsibility for the work jointly produced by them – entering the studio – and the sound director.

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