Bachelorarbeit

im Studiengang Audiovisuelle Medien

The development of the interactive Schoeps Film Sound Application

The study of location dialogue recording

vorgelegt von
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an der Hochschule der Medien, Stuttgart
am 21. März 2014

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Bearbeitungszeitraum: 21.12.13 – 21.03.14
Sprache: Englisch

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Zusammenfassung

Die vorliegende Arbeit beschreibt die Entwicklung eines interaktiven Online-Werkzeugs, das Filmtonmeister über Mikrofone und deren geeigneten Einsatz für die Dialogaufnahme am Drehort umfangreich informiert. Dabei wird zunächst auf den Charakter der menschlichen Stimme als primäre Tonquelle eingegangen. Des Weiteren werden die technischen Eigenschaften der einzelnen Mikrofone dargestellt und ihre geeignete Anwendung in unterschiedlichen, sowohl typischen als auch speziellen Situationen, ausgeführt. Ein wichtiger Teil der Arbeit bildet die Erstellung von fünf Kurzfilmen zur praktischen Demonstration der theoretisch erklärten Aspekte. Mit modernem HTML5 wurde dazu in einem aufwändigen Verfahren ein individueller Videoplayer konstruiert, der es dem Benutzer erlaubt, während der Wiedergabe zwischen mehreren Tonspuren des Videos störungsfrei umzuschalten, um so einen gezielten Einblick über die unterschiedliche Funktion der Mikrofone und ihr Verhalten zu erlangen.

Die Arbeit wurde für die Rubrik „Anwendungen“ des Internetauftritts der Firma Schoeps erstellt und bildet das Fundament eines in Zukunft wachsenden Projekts zum Thema Filmton, an dem zahlreiche Autoren beteiligt sein werden.

Abstract

This thesis describes the development of an interactive online tool which comprehensively informs production sound mixers about microphones and their proper application for professional dialogue recording on location. The first chapter contemplates the fundamental characteristics of the human voice as the primary sound source. The technical specifications of all used microphones are going to be introduced and their suitable deployment in different both typical and rare situations are being described in detail. An important part of the thesis is the creation of five short films in order to give a practical demonstration of all theoretical aspects. For presentation purposes a complex individually designed video player was built, using up-to-date HTML5 technology. The player allows the user to smoothly switch between multiple audio tracks while playing the video to gain an in-depth view of the function of the different microphones and their behavior.

The thesis was made available for the web presence of Schoeps under the category “applications”. It is the groundwork of a new resource for film sound; an ongoing and growing project with lots of different authors in the future.

|| Contents

|| Introduction

In film business it is a common opinion that the whole topic of production sound and dialogue recording on location rarely brings radical new findings and unprecedented news these days. Although it’s a fairly new craft, people have been recording dialogue successfully since the late 1920s. Since then the challenge has always been the same: Capture clean, consistent and intelligible audio.
For this reason, science continued to help invent better equipment that is used today and provides new standards and very applicable tools so sound mixers can get the best results in difficult recording situations and noisy sets.
The microphone manufacturer “Schoeps” has sensed the necessity of research that needed to be done in the environment of recording sound for picture and pursued to develop high-end condenser microphones, both analog and digital, including all accessories that are now used on numerous film sets all around the world.
One of their prime concerns is to not only provide their customers with the best equipment, but also enhance their service by adding useful tools showing their products in action.
In 2008 four students of the University of Media in Stuttgart created the Schoeps Showroom, an interactive application that shows Schoeps microphones in use for piano, ensemble, and vocal recordings. The advantage lies in the approach of how the recordings were done and the way they are being presented: The user can now switch through a number of microphones and microphone set ups and listen to the different ways they sound while the file is playing. It also presents information about the company itself, their microphones etc.
Since the showroom has been the only application on the Schoeps website for years, it was the company’s intention to create yet another application, dedicated to the area of film sound. The original idea was to deliver information similar to the already existing showroom while keeping it a whole new and separate application.
The function of this thesis is now to serve as kick-off for this project that is way overdue, according to Schoeps themselves.
It is the development of an online tool to not only give international customers the chance to learn about Schoeps products for film sound in a very interactive and demonstrative way, but also to help them understand all the details, specifications, odds and ends that there are to film sound. This thesis starts the project by covering

the topic of microphone selection and placement for professional dialogue recording on location.

The application is designed and implemented in latest HTML5 technology which supports the interchangeability of content and also allows to be expanded whenever needed. The Schoeps film sound application is going to be a hands on and growing project with lots of authors sharing their knowledge, tips and experience with sound mixers worldwide.

|| The Human Voice

Recording production sound really is only about one thing: "Capture clean, consistent and intelligible audio"[1]. Production sound primarily consists of dialogue recordings. It has often been said that dialogue in a film should never distract the audience from the action on screen[2], so it has to come across as naturally and realistically as possible. For this reason it is very important to discover and understand the source of sound that is to be recorded primarily: the human voice.

What seems to be trivial and obvious at first really becomes vital and deep by taking a closer look at what the human voice actually is and does. It's not a question of how to record just words that are put into sentences, it's rather a question of how to record phonemes, words and phrases that are thoughtfully put together, arranged and performed by the talent in order to communicate emotion, sentiment and passion. It's the sound mixer's part to get exactly that on audiotape, or HDD nowadays:

> *„Das Verständlichste an der Sprache ist nicht das Wort selber, sondern Ton, Stärke, Modulation, Tempo, mit denen eine Reihe von Wörtern gesprochen wird, kurz, die Musik hinter den Worten, die Leidenschaft hinter dieser Musik, die Person hinter dieser Leidenschaft: Alles das also, was nicht geschrieben werden kann." – Friedrich Nietzsche*[3]

In this first chapter we are going to look at how the voice is being evolved. We will then break it down to the differences of male and female voices. The directivity behavior of the sound field around the human talker will sum it all up.

[1] Ric Viers, *The Location Sound Bible: How to Record Professional Dialogue for Film and TV* (Studio City, CA: Michael Wiese Productions, 2012), p. 4.

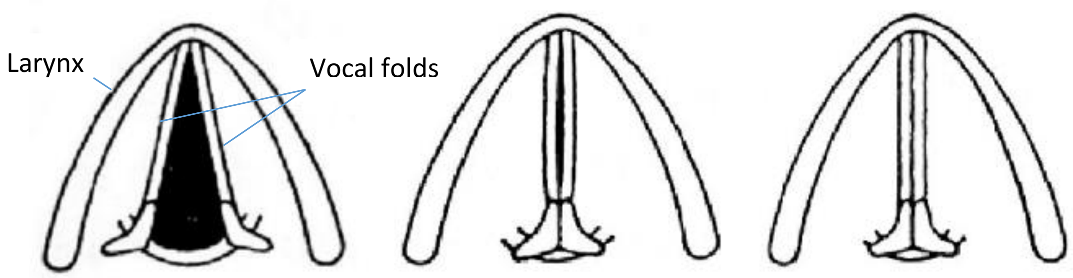
[2] see Viers, pp. 1–2.

[3] Wolf Schneider, *Deutsch für Kenner: die neue Stilkunde* (München: Piper, 1996), p. 131. ; The most understandable of speech is not the word itself, but rather tone, power, modulation, tempo with what a set of words is spoken, in short, the music behind the words, the passion behind this music, the person behind this passion: Everything that cannot be written.

1. Steps to Create Voice

The human voice originates in the larynx. It contains cartilages and ribbons, but is mostly hollow on the inside. Only the vocal folds are located there. These are elastic ribbons about 15 – 20mm in length. While the talent is breathing, the vocal folds are completely relieved. Now when the talent decides to speak, the vocal muscles force the vocal folds to close. But since air from the lungs pushes against the vocal folds, trying to open the crack (glottis) between them, they begin to oscillate.[4]

Depending on length and size, they oscillate at about 120 Hertz (men) or 220 Hertz (women).[5]



Position of rest *Position while speaking* *Position of closed vocal folds*

Figure #1[6]

This is called the neutral pitch of a voice. The amplitude of the oscillation is at maximum which means that the talent can speak very powerfully and over a long period of time at this pitch. All regular speech is nearby this frequency and definitely in the same octave, the lower two-thirds of the whole speech range.[7] Changes in pitch are achieved by straining the muscles in and around the larynx (for higher pitch), or by relieving strain to lower the pitch.

The glottis produces the *fundamental frequency* at roundabout 100 Hz (200 Hz respectively) and many more harmonics at 200Hz, 300Hz, 400Hz etc.

[4] see Heidi Puffer, *ABC des Sprechens: Grundlagen, Methoden, Übungen* (Leipzig: Henschel, 2010), pp. 52–53.
[5] see Marita Pabst-Weinschenk, *Grundlagen der Sprechwissenschaft und Sprecherziehung: mit 32 Abbildungen und 15 Tabellen* (München [u.a.]: Reinhardt, 2004), p. 22.
[6] Heinz Fiukowski, *Sprecherzieherisches Elementarbuch* (Berlin: Walter de Gruyter, 2010), p. 37 <http://site.ebrary.com/id/10498672> [accessed 13 February 2014].
[7] see Fiukowski, p. 46.

2. Timbre, Formants & Loudness

The area above the glottis, the vocal tract (all the way up to oral and nasal cavity), is now significant for building the final vocal tone. Each and every vocal tract has its own natural resonance to make for intensifying on *harmonics* or dampen them – in this way formants, “the spectral peaks of the sound spectrum of the voice”[8], are formed giving each voice its unique kind and character.

Different vowels can change the form of the vocal tract and ultimately its resonating behavior. Basically, vowels are considered emphases on formants. High frequencies (vowel “e”, emphasis on 200-400Hz & 3-3.5kHz) resonate in the head area, mid-frequencies cause the nasal cavity to vibrate and lower frequencies are perceived further down in throat and mouth cavity, all the way down into the chest area (vowel “u”, 200-400Hz).[9] Consonants are mostly unvoiced (frequency above 500Hz)[10] with a few exceptions depending on words and language, e.g. “**v**anilla”, “**th**ey”, “**z**ero”.[11]

Another component is the *loudness* of a voice. Since high pressure air is being moved through the glottis in order to produce a loud sound, molecules in adjacent resonating cavities are more likely to fall into their own oscillation process, too. The reason is because the vocal folds now oscillate at higher amplitude (not frequency). As a result the molecules inside the vocal tract have a higher sound particle velocity (thus higher sound pressure ⇒ higher sound intensity)[12] compared to low pressure air, causing more other molecules to resonate. This of course changes the sound of the voice. Additionally, when the voice is raised “the emphasis in the speech spectrum shifts one or two octaves towards higher frequencies”[13, Appendix figure #1].

The distinct sound of a voice is therefore defined by fundamental frequency, number and amplitude of harmonics and loudness.[14]

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[8] ‘Formant: What Is a Formant?’ <http://www.phys.unsw.edu.au/jw/formant.html> [accessed 13 February 2014].

[9] see ‘- Wagner_Arneton.pdf’, p. 6 <http://www.hdm-stuttgart.de/~curdt/Wagner_Arneton.pdf> [accessed 14 February 2014]; see Puffer, pp. 78–83.

[10] see ‘DPA Microphones :: Facts about Speech Intelligibility’, para. 1 <http://www.dpamicrophones.com/en/mic-university/technology-guide/facts-about-speech-intelligibility.aspx> [accessed 14 February 2014].

[11] see ‘Consonants: Voiced and Unvoiced | E Learn English Language’ <http://www.elearnenglishlanguage.com/blog/learn-english/pronunciation/consonants-voiced-unvoiced/> [accessed 14 February 2014].

[12] see ‘Schallfeldgrößen - Die Schallschnelle von Ivar Veit Nahbesprechungseffekt - Schallschnelle-Veit.pdf’ <http://www.sengpielaudio.com/Schallschnelle-Veit.pdf> [accessed 13 February 2014].

[13] ‘DPA Microphones :: Facts about Speech Intelligibility’, para. 1.

[14] see Günter Wirth, *Stimmstörungen: Lehrbuch für Ärzte, Logopäden, Sprachheilpädagogen und Sprecherzieher* (Köln: Deutscher Ärzte-Verlag, 1995), pp. 86–87.

3. Male vs. Female Voice

The frequency range can be from F_1(43Hz) up to e^4(2607Hz, sound of a newborn).[15] The following picture shows the average ranges for male (basso, baritone, tenor) and female voices (alto, mezzosoprano, soprano).

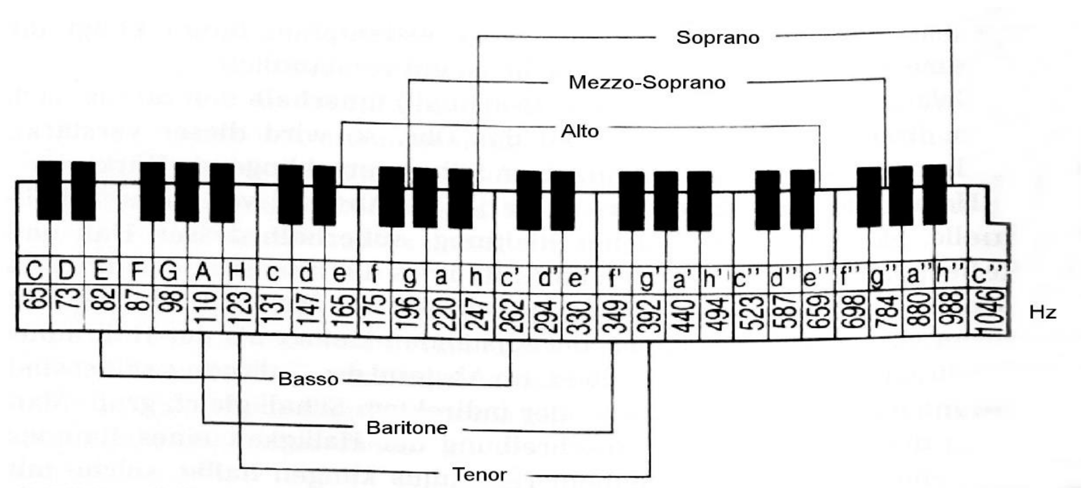


Figure #2[16]

The frequency ranges shown above do not refer to the average fundamental frequencies but rather the average timbre of a voice.[17]

As mentioned before, the usual speech range doesn't exceed the area from a quint to octave around the fundamental frequency. The whole range is between two and maximum three octaves. The mid pitch of the voice (always nearby fundamental frequency) and all other important parts determining the difference of voices can be estimated as follows:[18]

	Men	Women
Mid Pitch	98 - 131Hz	196 - 262Hz
Timbre	Basso: Dull	Soprano: Bright
Vocal Tract	Basso: Big, long	Soprano: Small, short
Larynx	Basso: Big, wide	Soprano: Small, narrow
Vocal folds	Basso: 24-25mm	Soprano: 14-17mm
Chest	Basso: Long, flat	Soprano: Squarish

[15] see Wirth, p. 95.
[16] Wirth, p. 118.
[17] see Wirth, p. 118.
[18] see Wirth, p. 123.

The whole physiological dynamic range of the human voice is 50-55dB. The loudness of regular speech ranks at 70-80dB[19], other literature states at roundabout 60-65dB[20] at a listening distance up to 1m[21]. For comparison, a clean and intelligible voice record requires a level of at least 50dB. As speech can have a dynamic range of 27dB for adults (5-12dB for children) over a longer period of time, the momentary speech level alternates too, approximately 5dB.[22]

4. Intelligibility and Directivity behavior of Sound Field around the Human Talker

On one hand, the vowels show the true sound of a human voice, on the other hand, especially in Western languages, it's the consonants that are crucial for speech intelligibility. Both are to be captured as best as possible. Therefore it's important to know what the dispersion of speech looks like. Tests show that applying a high-pass filter (e.g. at 500Hz) indeed reduces the speech energy, but the signal remains understandable. On the flipside, a low-pass filter (cutting at 1 kHz) sucks in about 60% of the intelligibility. One should avoid to cover or completely mask the frequencies 1-4 kHz by background noise in order to obtain a good ease of understanding. Figure #4 shows the formants of speech and their frequencies.

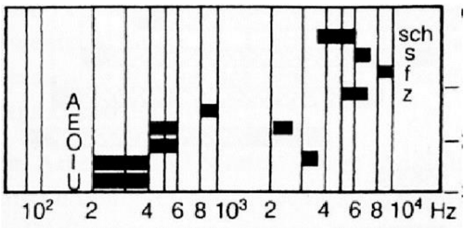
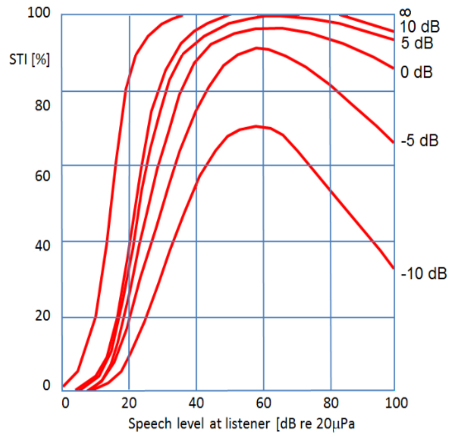


Figure #4[23]

Figure #5[24]



Also, too much reverb will be perceived as noise and worsen intelligibility. Therefore a high signal-to-noise-ratio is key. Background noise below 40dB(A) is not considered a problem if speech level is constant (referring to a listening distance of 1m). If the noise level increases to 40dB(A) and above, optimum speech level shows a s/n-ratio of 15dB

[19] see Wirth, p. 98.
[20] see 'DPA Microphones :: Facts about Speech Intelligibility', para. 1.
[21] see 'DPA Microphones :: Facts about Speech Intelligibility'.
[22] see Wirth, p. 100,121.
[23] Michael Dickreiter, *Mikrofon-Aufnahmetechnik: Aufnahmeräume, Schallquellen, Mikrofone, Räumliches Hören, Aufnahmeverfahren, Aufnahme einzelner Instrumente und Stimmen* (Stuttgart [u.a.]: Hirzel, 2003), p. 69.
[24] 'DPA Microphones :: Facts about Speech Intelligibility'.

or more. An increase of the speech level in postproduction will have a significant impact on how noise is being perceived as its level increases too. As figure #5 displays, no matter if s/n-ratio is positive or negative, the ideal speech level is identified at 60~75dB. Noise will seem louder and more interfering at higher levels.[25]

While frequency and level influence the comprehensiveness of the voice, it is very much important to also take a look at the directivity behavior of the human talker and the effect that not only the vocal tract has on it, but also the body and head. Below there are two polar patterns showing the differences of directivity in all directions. The level decreases almost exactly 7dB comparing front to back. Front to side is about 3dB less. The little boost at 330° vertical is interesting and has to be ascribed to the reflection off of the chest. The overall behavior is similar for both male and female performers.

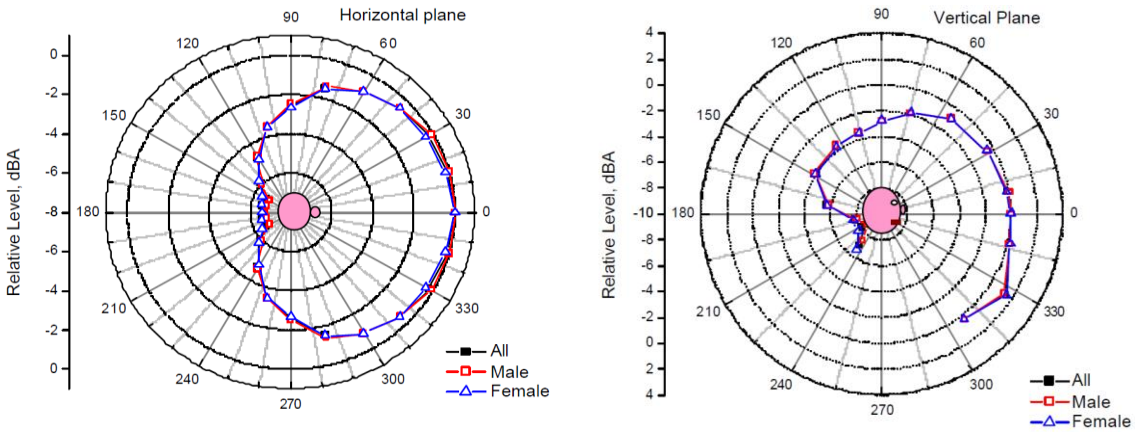


Figure #6, levels are A-weighed, measured at 1m[26]

The frequency dependent polar pattern then gives a better idea of how the directivity increases as frequency increases – for higher frequencies the polar pattern becomes more the shape of a cardioid, revealing a drop of 18dB from 160Hz to 8 KHz at 180°.[Appendix figure #2]

As proven in a recent and very accurate test by the National Center for Voice and Speech, University of Utah, the directivity indeed grows further for the highly important HFE[27], especially concerning male voices[28]. Another yet very subtle effect (starting at

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[25] see ‘DPA Microphones :: Facts about Speech Intelligibility’, para. 2; see ‘AV: An Examination Of Bandwidth, Dynamic Range And Normal Operating Levels - Pro Sound Web’, p. 2 <http://www.prosoundweb.com/article/an_examination_of_bandwidth_dynamic_range_and_normal_operating_levels/P2/> [accessed 14 February 2014].

[26] ‘Detailed Directivity of Sound Fields Around Human Talkers - rr104.pdf’, p. 14 <http://archive.nrc-cnrc.gc.ca/obj/irc/doc/pubs/rr/rr104/rr104.pdf> [accessed 14 February 2014].

[27] HFE = High Frequency Energy; the energy of 8 and 16 kHz octave bands

[28] Due to larger mouth sizes

1 kHz) is that loud speech becomes more directional (~3dB deviation) versus soft speech.[29]

1m of distance gives a good overall perspective on directivity and frequency behavior. But for film sound the microfone most likely gets placed even closer to the mouth. Eddy B. Brixen investigated the positions at 10, 20, 40 and 80cm from the mouth at 0°, 45° and 315° and presented his results at the AES convention 104. If the change of distance and direction (later referred to as *distance and angle*) had no effect on the frequency spectrum, the curves would be straight lines. However the charts teach us that the deviation at 45° is pretty narrow which means that the distance of the microfone doesn't effect the sound too much at this angle.

0° on-axis shows that the spectrum changes as the distance changes in a passable manner. The chart from a microphone position at 315° clearifies that the spectrum is effected tremendously by varying the microphone position – the influence of the body reflection gives a boost at 1400Hz.[30]

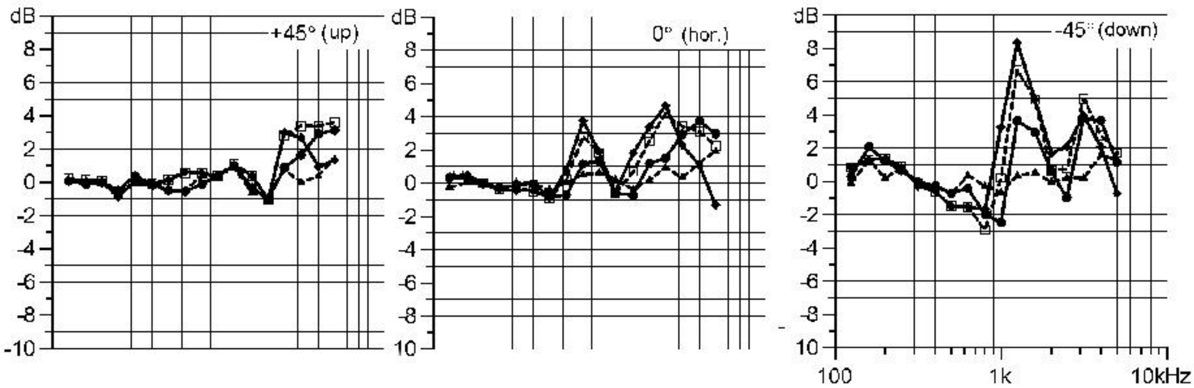

Figure #7[31]

Now that we've discovered the fundamentals, physical aspects and ways that underlie the behavior patterns of how the human voice 'works', in the following chapters we are going to come back to this information in order to see how the knowledge about the human voice as a source of sound effects the work of the production sound mixer.

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[29] see 'Horizontal Directivity of Low- and High-Frequency Energy in Speech and Singing' <file:///C:/Users/Markus/Desktop/Studium%20Film/7.%20Semester/Themengebiete/Richtcharakteristik%20Stimme/Horizontal%20directivity%20of%20low-%20and%20high-frequency%20energy%20in%20speech%20and%20singing.htm> [accessed 14 February 2014].

[30] see 'DPA Microphones :: Facts about Speech Intelligibility', para. 3.

[31] 'AES E-Library » Near-Field Registration of the Human Voice: Spectral Changes Due to Positions' <http://www.aes.org/e-lib/browse.cfm?elib=8452> [accessed 14 February 2014].

|| Microphones Used for Location Sound

The variety to choose from is almost endless. Microphones come in all kinds of different sizes, shapes and colors these days. The question of which ones to use for dialogue recording on a film set becomes most important during the process of pre-production. Acquainting oneself with the different locations and concerns they bring along as well as studying shot lists, learning about technical equipment that is going to be used (loud rain- or wind machines etc.) and, never least, talking to the wardrobe department is crucial in order to wrap one's mind around the circumstances that are going to come up regarding sound recording for a movie.

Certainly it is not so much about size, shape or color than it is about the performance of the microphone in a specific situation. As far as construction types of microphones for film sound are concerned, three main aspects are most important: The transducer principle, frequency response, and the microphone polar pattern. In this chapter we are going to look at microphones in general and which ones are used in the particular recording situation. For this thesis realistic comparative field tests have been done on inside and outside locations (see page 38). Schoeps small diaphragm microphones (CCM series) were used and examined during all tests and will serve as general examples.

1. Polar pattern, frequency response, transducer principle

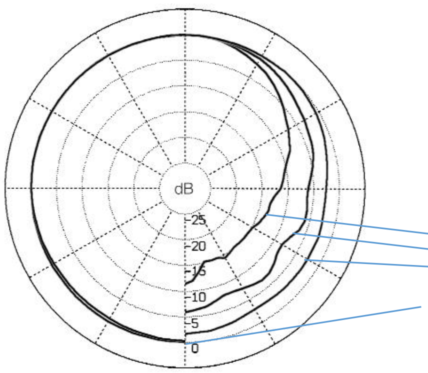
"The microphones convert acoustic energy into electric energy through a process called transduction"[1] as soon as sound waves hit the diaphragm inside the microphone capsule. This principle applies to all microphones even digital ones. The microphone only responds to either one physical quantity of a sound wave: Sound pressure, air pressure difference, or the particle velocity. This mainly determines the polar pattern of the microphone.

[1] Viers, p. 13.

Pressure transducer

Pressure fluctuation causes the diaphragm to oscillate as it is part of an airtight capsule (with a capillary, see below). Since pressure impacts the diaphragm from all directions in the same way, the pattern picks up sound in a 360° sphere around the capsule. This way a pressure transducer gets an omnidirectional polar pattern. For wavelengths smaller than the size of the capsule, a phenomenon called 'pressure doubling' comes into effect, meaning that these sound waves are being reflected by the diaphragm. The reflected wave superposes the arriving frontal wave resulting in an increase of sound pressure level up to 6dB. In addition, sound waves coming from a 180° angle can only bend around the capsule up to a certain wavelength (approx. 5 kHz). So for frequencies above, the omnidirectional pattern becomes a more directional pattern. A third effect is called 'wave interference'. High frequencies that are reflected by the microphone body to the side can cause interference with other sound waves to the point of complete cancelation. The polar pattern is therefore no perfect circle especially for very high frequencies. In short: The smaller the capsule the better the ideal omnidirectional pattern. An important feature considering film sound is that pressure microphones are not sensitive to wind noise.[2]

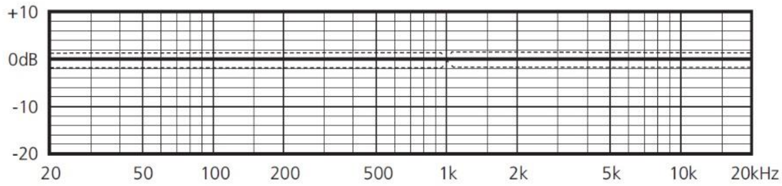
For professional dialogue recording, microphones with a flat frequency response are preferred since this means that all frequencies are reproduced without any amplification or attenuation (both in general called 'coloration').[3] The frequency response is influenced by capsule and transducer (see below). All following figures show the frequency response for sound coming from 0° (frontal). Since the diaphragm of a pressure microphone reacts differently towards high frequencies, one has to distinguish between diffuse field and free field frequency response: Diffuse sound field equalization is linear for diffuse high frequencies and shows a 6dB amplification in the free field. Free field equalization is linear for direct high frequencies and shows an attenuation of 6dB in the diffuse field. The Schoeps CCM 2 is a free field equalized omnidirectional microphone.



[2] see Thomas Görne, *Mikrofone in Theorie und Praxis: mit 23 Tabellen* (Aachen: Elektor-Verl., 2007), p. 33,34,39; see Dickreiter, p. 94.

[3] see Viers, p. 17.

Figure #8[4]



Pressure transducers are optimal for low frequencies – the diaphragm moves, no matter how slow or fast the pressure fluctuates. The cutoff-frequency is defined by a tiny hole in the capsule, the capillary. Without it, even a high pressure area would influence the diaphragm.[5]

Gradient transducer

This kind of microphone reacts to the air pressure difference or pressure gradient between front and backside of the diaphragm. Basically there are two ways of capsule construction.[6]

1. If the sound waves get to the diaphragm without any hindrance, the microphone has a figure of eight (or bidirectional polar pattern). Since sound has to travel a short way from front to back, phase differences occur. These eventually lead to a pressure gradient meaning for sound incidence from 0° and 180° the pressure gradient reaches its maximum. Sound waves from the side (45° and 90° angles) are fully cancelled (waves hit the diaphragm from both sides at the same time, no phase difference).[7]

   It is important to mention that the pressure gradient is dependent on the frequency. Pressure gradients are rather small for low frequencies and increase to high frequencies. The maximum pressure gradient is reached when the way from front to backside of the diaphragm, A-B, equals λ/2 of the soundwave (λ/2 ≙ 180° phase difference ⇒ maximum pressure gradient, as the front wave entirely “pushes”, the back wave entirely “pulls”). A-B is a fixed distance and therefore cannot stay smaller than λ/2 for all frequencies. The highest possible frequency with A-B = λ/2 is called transmission frequency. Pressure gradients for frequencies above will be decreasing. The length of A-B differs between

[4] ‘Kompaktmikrofon CCM 2 - Grafiken - SCHOEPS.de’ <http://schoeps.de/de/products/ccm2/graphics> [accessed 25 February 2014].
[5] see Dickreiter, p. 106,108,109; see Görne, p. 35,53.
[6] There are more; see further reading
[7] see Görne, p. 35.

microphones; it depends on the microphone where the transmission frequency is set at.[8]

2. By building in an acoustic delay element behind the diaphragm, sound waves coming from 180° have to take the same way to both back and front side of the diaphragm resulting in cancelation. For low frequencies this means an attenuation up to 20dB, becoming less as frequencies increase. Sound waves coming from 0° are getting delayed by the time they need to bend around the microphone plus getting through the acoustic delay element until they reach the back of the diaphragm; again, maximum pressure gradient is reached when A-B ≙ λ/2.

   Sound from the side only gets delayed by the time through the delay element to the back of the diaphragm, causing an attenuation of roughly 6dB.[9]

   For frequencies above the transmission frequency (depending on microphone size it is set at 4-10 kHz) the microphone has to work like a pressure transducer again in order to avoid a drop of the frequency response. The wave interference effect keeps the shape of the polar pattern of high frequencies similar compared to the one of low frequencies; pressure doubling (starting a little bit below the transmission frequency) even boosts the high frequencies to a certain extent and compensates a minimum.

The more microphone diameter, length of A-B and transmission frequency harmonize with each other, the merrier the transition from gradient transducer to pressure transducer.[10]

Depending on the amount of influence the acoustic delay elements have, there are different polar patterns and frequency responses from subcardioid to cardioid to supercardioid and hypercardioid. The following figure shows the Schoeps subcardioid CCM 21, Schoeps cardioid CCM 4 and supercardioid Schoeps CCM 41.

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[8] see ‘Bore_Peus_Mikrofone.pdf’, pp. 11–14 <http://www.ak.tu-berlin.de/fileadmin/a0135/Unterrichtsmaterial/Skripte/Bore_Peus_Mikrofone.pdf> [accessed 25 February 2014].

[9] see Dickreiter, p. 98,99.

[10] see ‘Bore_Peus_Mikrofone.pdf’, p. 14+21.

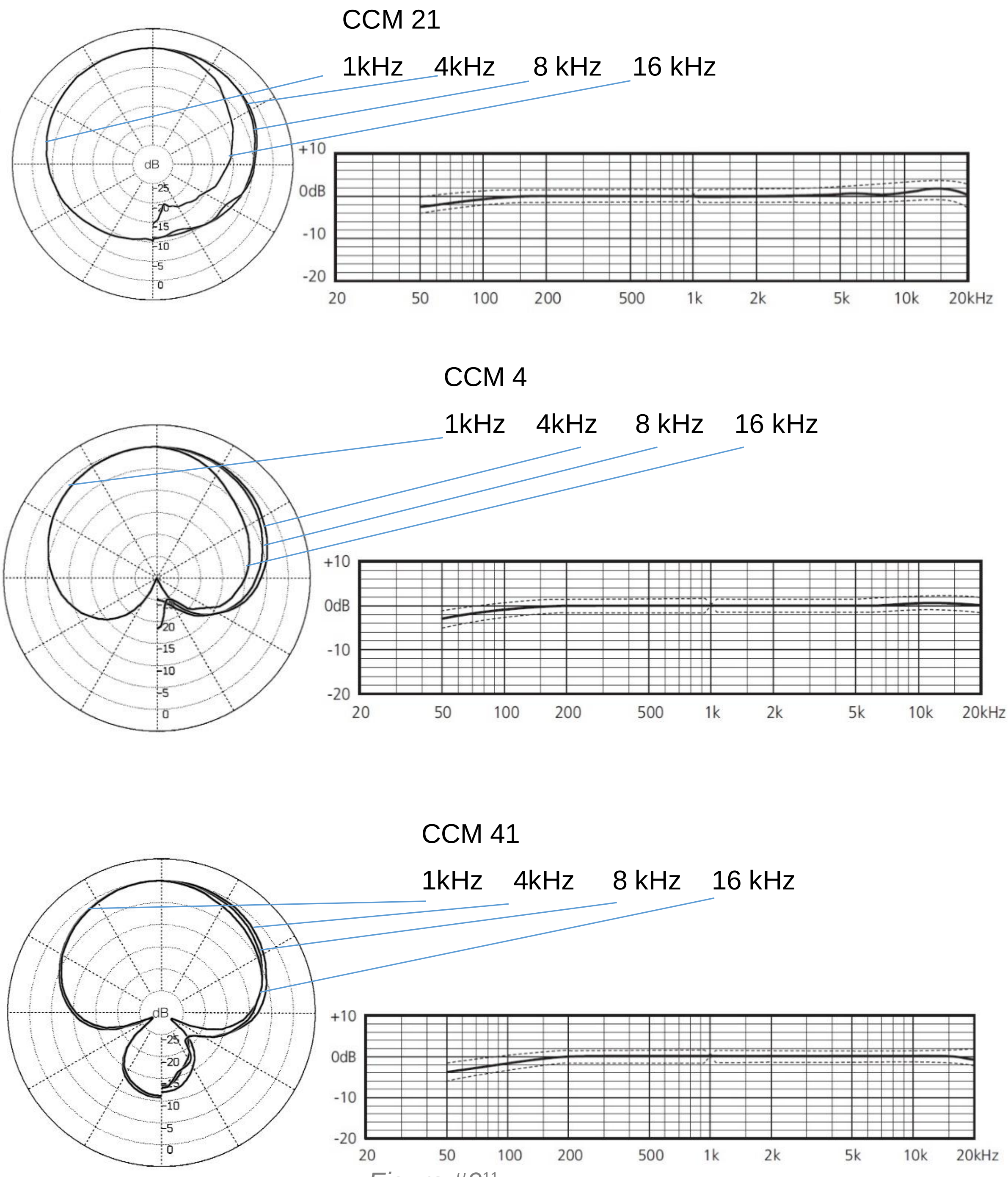

Figure #9[11]

The graphs show that gradient transducers have a very linear frequency response for most of the frequency range for both diffuse and direct sound. Due to physics (see above) low frequencies are not getting reproduced as well whereas pressure transducers are known for a good reproduction of the lows.[12]

[11] see 'Kompaktmikrofon CCM 2 - Grafiken - SCHOEPS.de', pp. 12–14.
[12] see Dickreiter, p. 106.

In the diffuse field, unidirectional microphones have a directivity index of 3,17dB (subcardioid), 4,8dB (cardioid) and 5,72dB (supercardioid); see pages 46-49.[13]
In the near field all gradient microphones experience an increase or decrease of sound pressure changing proportional to the distance of a sound source (1/d; d=distance). For this reason there is yet another, distance related, pressure gradient from A-B, next to the phase-dependent one explained earlier. As opposed to the phase-dependent pressure gradient, the distance related deviation is not dependent on frequency. Since low frequencies don't cause such big phase differences from A-B, here the distance related pressure gradient comes more into effect. The result is a strong artificial amplification of low frequencies called 'proximity effect' which is most intense for the figure of eight pattern and lessens towards non-directional patterns. For film sound recording the proximity effect is undesirable, but mostly unproblematic because the distance from microphone to mouth is almost always greater than 20cm. The following figure gives an idea of how different frequencies would affect dialogue recording within certain distances to the microphone.

Tune	Frequency	Distance
C_2	16 Hz	1.40 m
C_1	33 Hz	70 cm
C	66 Hz	35 cm
c	131 Hz	17 cm
c'	262 Hz	8 cm
c''	523 Hz	4 cm

Figure #10[14]

Gradient transducers are therefore about 20dB more sensitive to impact sound, wind noise and plosives than pressure transducers. Many gradient microphones thus have a built in low-cut filter to compensate for the proximity effect.[15, 16]

Velocity transducer

Velocity transducers behave like gradient transducers and mostly have a figure of eight polar pattern. The diaphragm is a small electroconductive ribbon which gets moved directly by the air molecules oscillating around their rest position after a sound wave

[13] 'Theoretische Mikrofondaten Zu Den Richtcharakteristiken REE DRF DSF REF REB UDI FTR Gamma - TheoretischeMikrofondaten.pdf', p. 1,2 <http://www.sengpielaudio.com/TheoretischeMikrofondaten.pdf> [accessed 3 March 2014].
[14] Görne, p. 41.
[15] see 'Bore_Peus_Mikrofone.pdf', pp. 14–16.
[16] See information about Schoeps CMIT shotgun built-in filters

has activated them. The ribbon is both diaphragm and electromechanical transducer. The light ribbons are very sensitive to wind and plosives hence not useful for dynamic dialogue recording.[17]

Transducer principles

There are two main principles of how mechanical oscillations are transduced to electric oscillations: Electrostatic and electrodynamic.[18] The transducer principle influences frequency response, (see above), pulse fidelity and the emitted voltage.

Condenser microphones use the electrostatic principle. The diaphragm is the front plate of a condenser which is powered by a constant voltage. The stored charge between the front and back plate, the capacitance, now changes as the diaphragm begins to oscillate forcing the distance between front and back plate to change too. This change then becomes the electric signal. To charge the capacitor, external power supply known as phantom power is needed. Back in the day, microphones used 12V (T-power), today phantom power is 48V ⇒ condenser microphones are active transducers reacting to the oscillation of the diaphragm.

The capsule can use either AF audio frequency (used in Schoeps small diaphragm microphones due to better linearity[19]) or RF radio frequency to create the audio-frequency signal.[20]

Considering true condenser microphones, the process to generate a linear frequency response is rather complex. Condensers are elongation transducers. Pressure transducers use light-weight high tension diaphragms and react to the oscillation of the diaphragm. At one point it reaches its maximum (resonant frequency) - the mass of the diaphragm blocks faster oscillation. The frequency response of the oscillation represents a low pass. Therefore pressure transducers are high tuned in order to get the highest resonant frequency possible. For the final frequency response this is being combined with the high pass characteristic of the capacitor.

The frequency range to the lows is only affected by the proportion of the high pass (suitable setting required for good representation of low frequencies).

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[17] see Görne, p. 43; see Dickreiter, p. 94.

[18] There are more but irrelevant for today's use in high quality microphones

[19] see 'Produktgeschichte - SCHOEPS.de' <http://www.schoeps.de/de/history_products> [accessed 27 February 2014].

[20] see Dickreiter, p. 100. for further reading

For gradient transducers the oscillation increases up to the cutoff frequency and decreases above; the frequency response of the oscillation equals a band pass. Gradient transducers are rather mid-tuned[21]; the combination of this band pass with the high pass characteristic of the capacitor delivers the final frequency response. Further electric filters and acoustical oscillators guarantee for a flattened, linear frequency response.[22]

Dynamic microphones use the electrodynamic principle. The diaphragm is connected to a moving coil which is wrapped around a magnet. When the diaphragm starts to oscillate, induced voltage occurs because of the movement of the coil inside the magnetic field. Dynamic microphones can handle high SPL, are very robust and good to use in heavy weather conditions, but at the cost of a good transient response and reproduction of high frequencies. Dynamic microphones are passive transducers and react to the velocity of the diaphragm.[23]

For professional dialogue recording, the true condenser with a small diaphragm is the most common transducer being used, for these condenser microphones have a very clear transient response, capture high frequencies over 20 KHz, have no undesired coloration like microphones with big diaphragm[24], and "faithfully reproduce subtleties in the sound wave's dynamics".[25]

Next to true condenser microphones there are passive AF transducers that don't need a power supply to operate: Electret condenser microphones. Their back plate is a permanently charged Teflon foil. With the charge difference between front plate (diaphragm) and back plate, the electret condenser capsule eventually works just like a true condenser except the power supply is already built-in. However, most electret microphones use phantom power, or the power of an external battery, to operate their impedance converter. Electret microphones actually wouldn't meet the standards for feature films as they are quite noisy compared to true condensers. Yet their capsules can be manufactured very small, so they are used in lavalier microphones.[26]

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[21] Other literature states "low-tuned"; see '() - Mikro.pdf', p. 10 <http://www.uni-koeln.de/phil-fak/muwi/ag/umdruck/mikro.pdf> [accessed 17 March 2014].
[22] see Görne, pp. 52–57.
[23] see Viers, p. 14,15; see Görne, p. 43,47,53,54.
[24] see Görne, p. 176.
[25] Viers, p. 14.
[26] see Viers, pp. 16, 28; see Görne, pp. 67, 68.

2. Fulfill requirements

Note: The following guidelines have been tested, tried and proven to be working very well. Yet they are to be considered general recommendations; different films and sets call for new paths in dialogue recording.

Each microphone has its unique specialties affecting use and application as well. For professional dialogue recording, the requirements to uphold best recording results differ from studio vs. inside vs. outside location, noisy vs. quiet situations, weather conditions, lighting, number of talents to mic, microphone positions etc. These are all factors forcing the sound mixer to choose microphones carefully. There are some general standards that always have to be met and kept in mind when selecting microphones:

The microphone must not be seen in the picture and still reach best intelligibility with low noise level, little to no background noise and a sound perspective matching the camera perspective. These are the main goals in order to achieve prime dialogue recordings.

In addition, microphones inside run the risk of producing a colored sound, whereas microphones for outside locations have to be robust, rugged, and provide maximum reduction of disturbing noise while still not missing the goal of a good transmission range.

Striving for consistency in tracks can be maintained by choosing the microphones before the recording of an entire scene using it during all shots. The sound source always has to stay on-axis, (best at 0°) even for non-directional microphones for high frequencies' sake (see polar pattern above).[27]

3. Boom Microphones

There is no such thing as a boom microphone. It is a common term that refers to the microphone at the end of a boom pole reaching out to the talent as closely as possible. As production sound is mainly about dialogue recording, directional microphones are used on a boom to minimize unpleasant sound in the picture and also background noise. For inside locations and studios, a cardioid or supercardioid is recommended because of their directionality, linear frequency response (see above) and very natural sound. Exterior shots benefit from the use of a shotgun microphone with the ability to

[27] see Viers, p. 25; see Görne, p. 245,246.

reduce noise from the sides to a much greater extent. Shotguns are a combination of line and gradient microphones. The "line" is a 10-40cm long tube with so-called phase ports (slits along the side) in front of a gradient transducer (supercardioid for CMIT 5 U). Sound waves arrive at the microphone from all directions. The phase ports force them to "arrive at the diaphragm at slightly different timings. These different timings cause a sophisticated phase cancellation"[28]. Sound waves coming from 0° are not influenced by the interference tube. The result is a polar pattern similar to a shotgun, where the name of the microphone originates. Figure #11 shows the polar pattern of the Schoeps CMIT 5 U and its frequency response.

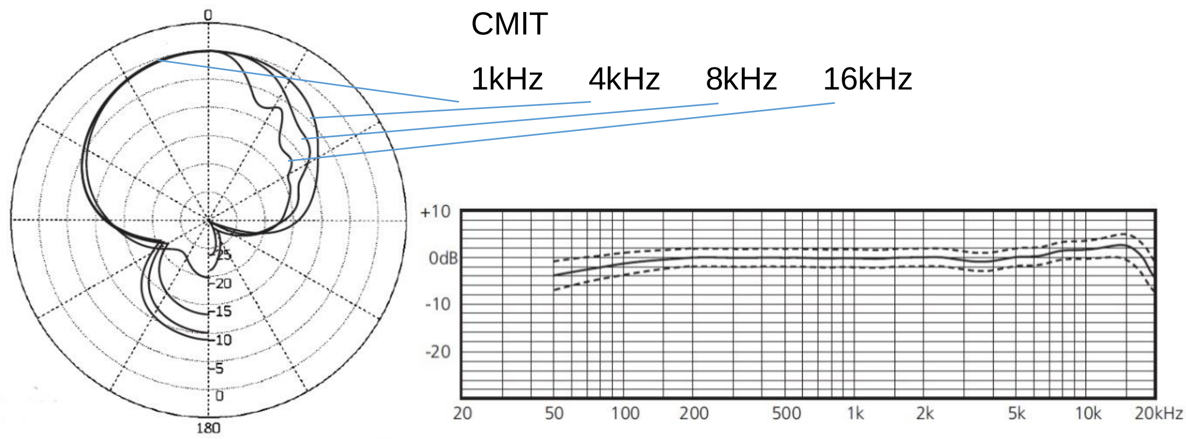

Figure #11[29]

It is self-explanatory that the frequency response curve does not look the same for different angles of incidence as the rejection by the interference tube has variant influence on each frequency.[Appendix figure #3]

Shotgun microphones actually have two polar patterns. The interference tube only comes into play for high frequencies with wavelengths shorter than tube length. The microphone behaves like a supercardioid for lower frequencies with greater wavelength. There are long shotguns (26cm of tube length) for great damping toward low frequencies. [30]

With their SuperCMIT 2 U, Schoeps offers a new kind of solution: The microphone has a second, rear-facing capsule (cardioid), set behind the forward-facing capsule and the interference tube. A DSP compares and analyzes the two signals for frequencies below 6 kHz and deduces which part of the signal is diffuse arriving sound. This part of the

[28] Viers, p. 26.
[29] 'Schoeps CMIT Manual.pdf', p. 25.
[30] see Görne, p. 84,85; see Viers, p. 26; see Dickreiter, p. 112.

sound is then suppressed 5-10dB depending on the impact the DSP gets on the microphone signal (user can switch between two presets). Frequencies above 6 kHz are not further being processed, the interference tube's effect totally sufficient.[31, Appendix figure #4]

It should be avoided to use shotgun microphones inside. Room reflections of high frequencies are not represented as much, causing diffuse sound to end up with a high-frequency rolloff. The result is an unnatural dull sound. Also, the tail in the polar pattern might be a problem enhancing the reverberation of a room. Echoes from a small room might bounce back into the microphone and not be completely cancelled out. There will be outphasing and maybe an echoey sound result.[32] To compensate for this (and to ensure speech intelligibility), a high frequency boost filter can be applied. Also, narrow lobes of the polar pattern may cause unwanted comb-filter-effects when microphone or talent are in motion.
Shotgun microphones usually come with a low-cut filter at 80Hz with an 18db/oct slope to countersteer disturbing low frequencies such as wind. The Schoeps CMIT even provides a second filter at 300Hz with a 6dB/oct slope to compensate for the proximity effect.[33]

4. Lavalier Microphones

Lavalier microphones are (most of the time omnidirectional) electret microphones with a very small diaphragm and therefore noisier than true condensers. Yet they are the go-to alternative when shots are too wide for a long shotgun microphone to get intelligible and crisp sound, or on narrow locations where booming is no option. Lavaliers are specially made to be mounted directly on the talent preferably in their chest area at about 15-25cm away from the mouth. The chest resonant frequency is at roundabout 800 Hz (for men at 700 Hz)[34] which lavaliers are not as responsive in this frequency range. The chest area on the other hand lacks high frequencies, so they are emphasized. The intention of this unusual equalization curve (standardized by the

[31] see 'Schoeps SuperCMIT Manual.pdf', p. 12.
[32] see 'Pup Tent Media 2 - Sennheiser 816 Shotgun vs Audix SCX-1 HC Shoot Out in Small Studio - YouTube', pt. 2:36min <http://www.youtube.com/watch?v=Irc7Q9ek0J4> [accessed 28 February 2014]; see 'Schoeps CMIT Manual.pdf', p. 17; see Viers, p. 26.
[33] see 'Schoeps CMIT Manual.pdf', pp. 17–19.
[34] see 'Bore_Peus_Mikrofone.pdf', p. 60.

IRT[35]) is to receive a linear frequency response curve in the end. The following figure shows the IRT curve and the polar pattern of a DPA 4071 lavalier microphone.

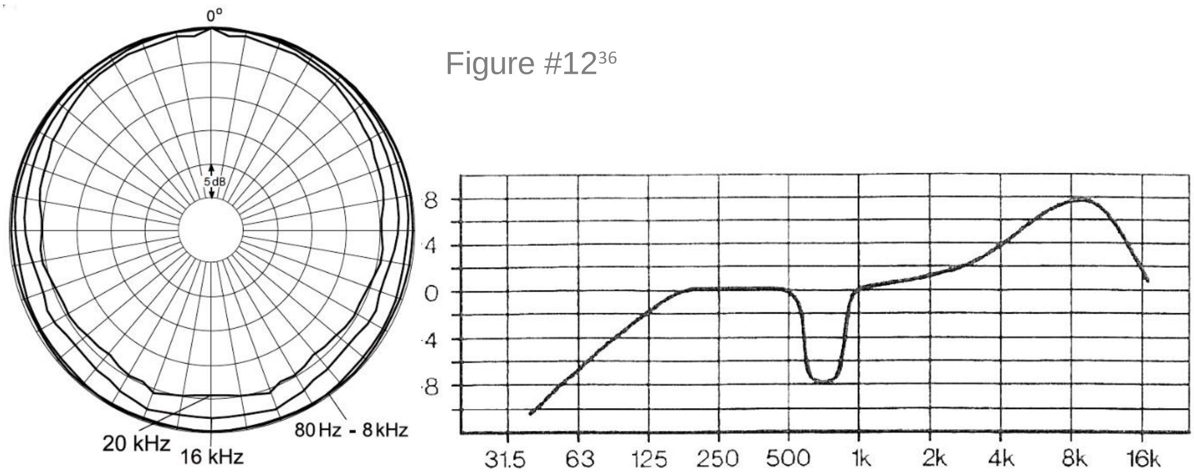

Figure #12[36]

Many lavalier microphones however do not possess an equalized curve. The necessary adjustments for proper sound have to be done by the mixer manually.
Lavalier microphones used for film sound usually are pressure transducers with an omnidirectional polar pattern. The advantage is that they are much less sensitive to clothing rustle and plosives than cardioid gradient microphones.[37] Occasional turns of the talents head and fast movements are not resulting in altered, or "fallen out" sound due to a unidirectional pickup pattern. There are side-address and top-address lavalier microphones, giving multiple and different mounting ability. The construction has also an influence on sound itself, so mixing two different models together in the same scene is to be avoided.[38] Lastly, not every lavalier sounds well for every voice; they have to be carefully selected for each actor.

5. Planted Microphones

Planted microphones can be any type of microphone that is hidden directly on the set. They are very useful for single lines of dialogue that cannot be picked up by the boom microphone. This can be an actor leaving the room, looking out the window, bending down under a table etc. Small diaphragm microphones or lavaliers are very convenient to accomplish most types of tasks - the choice of what kind and which pickup pattern

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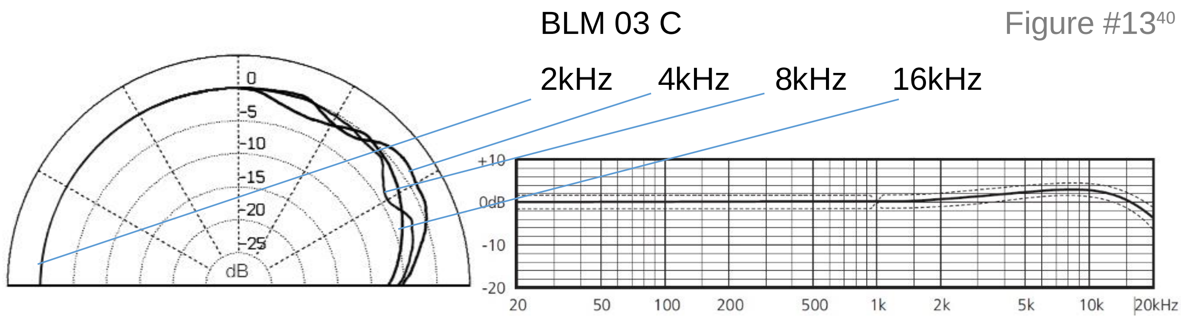
[35] Institut für Rundfunktechnik (Germany)
[36] Görne, p. 79; 'DPA-4071.pdf', p. 2
<http://www.dpamicrophones.com/en/download/~/media/PDF/Download/Users%20Manuals/DPA-4071.pdf> [accessed 3 March 2014].
[37] see Görne, p. 78,79; see Dickreiter, p. 110.
[38] see Viers, p. 30,32,67,68.

depends on each situation individually. Anyhow, a boundary layer microphone opens up considerable options on difficult, tight sets or difficult lighting situations. This type of microphone is usually an (originally) omnidirectional condenser microphone and stands for a good reproduction of low frequencies. The capsule sits inside a small base plate and is mounted flush with the boundary surface. This way the effects of reflective surfaces are leveraged: There are no resonant room modes, standing waves or disturbing comb-filter-effects , there is always a maximum pressure (buildup of 6dB; pressure doubling for all frequencies) because the arriving sound wave superposes the sound wave reflected off the surface. However, for frequencies with wavelengths greater than the diameter of the surface, the sound pressure level declines 6dB. Therefore they are placed on walls or floors most of the time.

Unlike conventional pressure transducers, moving sound sources (talents) do not cause any coloration of the hemispherical polar pattern. The frequency curves of both diffuse and direct sound are identical, which helps the consistency of the dialogue recording regardless of unpredictable action or unplanned lines. In lieu of a real boundary layer microphone, any type of small diaphragm microphone mounted on top of a surface can be used as one, even unidirectional microphones.[Appendix figure #5]

Boundary layer microphones have a directivity index of 3dB higher than omnidirectional pressure transducers (0dB). If one decides to put a unidirectional microphone close to a reflective surface (capsule has to be vertical to surface) the directivity index adds up to 7,8dB for cardioids with a directivity index of 4,8dB. Since gradient transducers are sensitive to impact sound, it is recommended to isolate them from the surface a few millimeters (use foam) to prevent unpleasant sound through surface vibrations. It depends on the setup if a directional or non-directional boundary layer microphone is advantageous.[39] Figure #13 shows the relatively small Schoeps BLM 03 C which is perfect for film sound recording.

Figure #13[40]



[39] see Dickreiter, pp. 96, 114; see Viers, p. 32,33.
[40] 'Schoeps blm3 Manual.pdf', p. 13.

Wrapping this up, it is still to mention that a lot of production sound mixers today make their choice of microphone based upon this priority list:

• Boom from above

• Boom from below

• Boom mike planted (stationary) on set

• Lavalier mike planted on set

• Lavalier mike worn by actor

• Radio mics[41]

This thesis shares and supports this point of view to the extent that it doesn't become mandatory, or narrows one down to a boxed-in way of thinking.

[41] 'Microsoft PowerPoint - Mic Techniques for Episodic TV_protected - Mic Techniques for Episodic Tv_protected.pdf', p. 9 <http://filmtvsound.com/phocadownload/presentations/audio-technica/mic%20techniques%20for%20episodic%20tv_protected.pdf> [accessed 4 March 2014].

|| Placing the Microphone

It is one thing to choose the right microphone. It is another thing to get the right microphone in the right place at the right time[1]. More often than not, microphone placement is much more essential than the directional pattern, brand, or type of microphone. There are two main factors determining proper microphone placement: *angle* and *distance*. As discussed previously, the angle is not as important for omnidirectional microphones. “With unidirectional microphones it is important to stay on-axis”[2] matching the best use of the polar pattern. The distance to the sound source should be kept as minimal as possible, so that direct sound predominates indirect sound at all times. On the one hand, the human ear is used to a certain amount of diffuse sound, so as to assess and understand from what kind of sound field (or room) sound is coming from.[3] On the other hand, it is a lot harder to remove reverb than to add reverb in postproduction. Of course there are (expensive) software plug-ins leading to good results, but always at the cost of the loss of audio quality.[4] The rule of thumb is to capture natural sounding dialogue, rather but not too dry and in no case reverberant and indirect. As discussed in the first chapter, too much reverb results in bad intelligibility.

Unfortunately it is still a common opinion not only amongst non-professionals, but even well experienced producers and directors that especially shotgun microphones are able to “zoom” in to the source of sound and reduce all background noise whatsoever. Many times it is not until postproduction that they realize how important clean, intelligible and consistent audio is for their picture. This myth of reach has been addressed in an open letter from the sound department, also including all kinds of important information about common problems and misunderstandings considering production sound.[5] Still, film sets are always noisier than one would hope for – this is why it is crucial to select microphones carefully as well as bringing them into the right position.

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[1] see Viers, p. 120.
[2] Viers, p. 39.
[3] see Viers, pp. 37–39.
[4] ‘Zynaptiq: News’ <http://www.zynaptiq.com/> [accessed 4 March 2014].
[5] ‘An Open Letter from Your Sound Department - A Production Sound Manifesto Written by Audio Professionals’ <http://filmsound.org/production-sound/openletter.htm> [accessed 4 March 2014].

In this chapter we are going to discuss microphone placement fully aware of the directivity behavior of the sound field around the human talker, sound perspective matching visual perspective, and noise reduction.

1. Boom Positions and Techniques

The boom microphone is the number one option in a production sound mixer's priority list. It always has to get as closely to the sound source as possible. The distance is determined by the framing – so the microphone goes right above frame line in order to reduce noise and hiss to the highest possible extent. As Ric Viers, author of the 'Location Sound Bible', puts it: "The boom mic will always capture the most natural audio when positioned overhead and pointing downward toward the talent"[6] in an ideal angle of 30 degrees. This way sound and visual perspective match best. Opinions differ on the question of where to aim the microphone at. Some experienced boom operators and sound mixers are convinced that aiming right at the talent's throat is best, others find the bridge of the nose as the right target. Some swear that the mouth is the best, being the spot where the voice issues. The most spread and common opinion is to focus on the sternum, or upper chest, where the voice originates.[7] This technique called *lobing* makes sense - as we've learned in the first chapter, low frequencies resonate in the chest area giving the voice a crisp but still warm sound, rather than a nasal coloration when aiming toward the facial area where mid- and high frequencies resonate. Yet there is no lack of high frequencies because the microphone is close to the mouth. Figures #6 and #7 clearly show how consistent sound is at around 45° with only subtle deviations as far as distance is concerned; this microphone position provides a very stable frequency spectrum and very good amount of sound level as well. Booming from above gets the microphone closer to the mouth than booming from below – the microphone picks up SFX while dialogue still dominates.[8]

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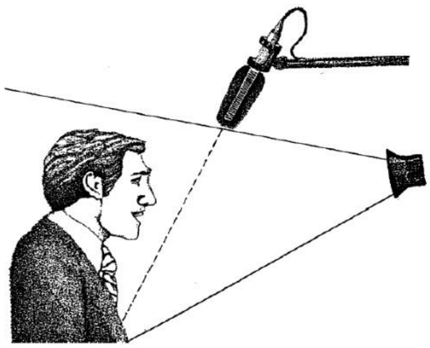
[6] Viers, p. 39.

[7] see David Lewis Yewdall, *Practical Art of Motion Picture Sound*, 4th ed (Waltham, MA: Focal Press, 2012), p. 81.

[8] see Viers, p. 39,40; see '79_Filmtonkochbuch.pdf', p. 2,3 <http://produktionslabor.mt.haw-hamburg.de/media/79_Filmtonkochbuch.pdf> [accessed 4 March 2014]; see 'Microsoft PowerPoint - Mic Techniques for Episodic TV_protected - Mic Techniques for Episodic Tv_protected.pdf', p. 15.



Sometimes actors delivering lines while bowing their head, low ceilings, lighting, rain (boom blocking intentional rain or water buildup forming a dripping stream), sun (shadows), or framing require booming from below known as *scooping*. The angle of the microphone toward the mouth stays the same. There are several drawbacks to this technique:

Figure #14[9]

- Overhead noise (planes) are likely to be picked up and disturb the dialogue; also, SFX (hand movement etc.) can now be louder than the voice.
- As seen above, the chest reflects (boosts) some frequencies more than others, especially the lower ones. Since high frequencies lack at this microphone position, the resulting sound is rather low and mid-range heavy. Figure #7 also shows big differences at 315° of the frequency spectrum depending on distance.
- If the microphone gets too far away from the sound source, audio becomes weak and thin. The proportion of direct and indirect sound doesn't match the visual perspective.[10]

Booming from horizontally 0° might not always be an option. Wind, background noise, walls etc. can determine the direction and may force one to place the microphone to the side. As figure #2 in the appendix displays, this is not a problem for low frequencies, but might become one for higher frequencies important for intelligibility. Despite horizontal movement, one has to make sure that the microphone still aims at the talent in a vertical angle of 30°. Here is the point where microphone selection and placement get intertwined. The Schoeps CCM 41 has a wider pickup pattern, gathering sound easily from +45° and -45° horizontally whereas the CMIT shouldn't go too far to the side of course. There is also a great difference between brands. For example: The Schoeps CMIT and Sennheiser MKH 60 are both high-end shotgun microphones. It is true that the CMIT has a narrower lobe than the MKH 60 in general. However the Sennheiser is much more directional for high frequencies than the CMIT which is more

[9] '79_Filmtonkochbuch.pdf', p. 3.
[10] see Viers, pp. 40–42.

balanced and optimized for sound quality rather than directivity. It depends on the sound mixer's preferences which one to choose for a placement toward the side.[11]
In general, the boom microphone must *never* fall behind the head or too far sideways. Not only is there a terrible danger of coloration across the whole frequency range, but also a decrease of sound pressure level up to more than 6dB. Especially loud speech containing HFE is more likely to 'fall out' as the directivity pattern gets narrower (see chapter one and figure #6).

Cueing

Herewith comes another reason why to aim at the talent's chest rather than pointing directly at the mouth. When actors are in motion, it is difficult to hit a target that is the size of a golf ball. It is very important to avoid coloration and hold up consistency in a dialogue track - aiming at the chest as a wider target is much more forgiving. Choosing a cardioid over a shotgun also helps fast head turns and difficult boom positions while sound remains natural.[12]
A scene with multiple actors requires a change of microphone position to gather crisp and intelligible audio from each of them by always keeping the same distance between the sound source and the microphone. This means the microphone has to go where the action is at all times. With that being said, the factor of the right *angle* yet dominates the *distance* for evenness reasons of the recording. So when changing the microphone's placement and angle, the boom operator first pulls it up a few centimeters (keeping the same angle), then turns it toward the speaking talent and lowers it again. This technique guarantees a safe and smooth change. Then again, turning the microphone is not a choice with loud noise in the background. Simply moving the microphone while keeping it at a fixed angle throughout the whole scene results in a much smoother track because background noise is not constantly switching from off- to on-axis.
In heated discussions it is recommended to use a wider pickup pattern. This way of course not all the voices are recorded perfectly from 0 degrees horizontally and 30 degrees vertically, but: one has to mediate between spoken lines by using a wider polar pattern and keeping the microphone somewhere mid-placed, so that none of the

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[11] see Yewdall, p. 76; see 'Schoeps CMIT Manual.pdf', p. 4; see 'Mikros MKH GB - 982b087fe6f041c78ed651d640dd04cc.pdf', p. 1 <http://en-de.sennheiser.com/downloads/982b087fe6f041c78ed651d640dd04cc.pdf> [accessed 4 March 2014].
[12] see Viers, p. 40.

voices sounds thin and falls off-axis with audio perspective not matching the shot. A boom operator has to memorize the lines, so the microphone can be brought into place at the right moment. It is good advice to pay attention to the non-speaking talent. Anticipating movements by watching their shoulders or eyes prepare for sudden moves of the boom microphone. Looking for an actor breathing in, or opening the mouth are almost always signs for the next line. If a shotgun microphone needs to be used in a dialogue with overlapping lines, one should favor the new line of conversation, as the human ear always pays more attention to the new dialogue. To provide full coverage, location sound mixers use multiple boom microphones (same make and model for sound consistency, possibly mounted on a C-stand) to have more options in postproduction.[13] Furthermore, production sound mixer Nicholas Allen advances the view (with only a few other mixers) that even recording off-screen dialogue is almost mandatory for a smooth postproduction process.[14]

2. Lavalier Positions and Techniques

The disadvantage of the boom microphone is that it only can do so much. It is doomed to stay outside the picture which can lead to thin and weak dialogue i.e. with a sitting next to a standing talent. For this reason especially multi camera shots are likely to not match the boom's sound perspective. Also, having more than one boom operators to cover off-camera dialogue indeed is what one would hope for, but improbable for most productions. Using lavalier microphones is an alternative solution that comes with a lot of benefits, but also opens up a world of difficulties.[15]

Nowadays lavalier microphones deliver professional crisp and intelligible audio in difficult situations (weather, tight location) and very long shots where even a Schoeps SuperCMIT loses its effect. Dependent on the director's intention (or as stylistic element), a close sounding voice is indeed contradicting a wide shot, but can still be legitimate. Sitting just a few centimeters away from the sound source, lavaliers even help isolate unwanted loud background noise, making sound even more compact, condensed and clean.[16]

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[13] see Viers, pp. 60–64.
[14] see 'SoundWorks Collection - Production Sound Mixer Nicholas Allen', sec. 1:40min <http://soundworkscollection.com/videos/production-sound-mixer-nicholas-allen> [accessed 5 March 2014].
[15] see Viers, p. 67.
[16] see Görne, p. 252,253.

They sound best when placed uncovered and on the outside of clothes. However, in dramatic productions lavaliers are not to be seen and therefore hidden from the camera underneath clothing.

This introduces mainly three problems:

1. The sound of a lavalier microphone is very unnatural compared to the boom microphone. There is almost no sound perspective (makes shot seem claustrophobic) since they are too close to the mouth, sounding dry and sterile, almost “too perfect and isolated”. Clothing acting as windscreen even enhances this effect.[17]
2. Covered by sometimes several layers of clothing, voices tend to sound muffled. Especially missing sibilance in the frequency range from 6 kHz – 10 kHz is noticeable if not corrected by an equalizer. Clothing or chest hair noise might destroy the audio and make it unusable.
3. The lavalier almost acts like a boundary layer microphone mounted on the talents chest as reflective surface. Though there can be inconsistency in volume (and maybe tone color) as the talent turns their head, causing the dialogue to fall off-mic.[18]

There are different ways to successfully treat and handle these effects:

1. An additional shotgun boom microphone can act as so-called “bleed-mic”: It is used to add the missing proportion of general ambiance, surrounding SFX, such as footsteps, body movements, interaction with props etc. Audio-technica suggests to bring about 30% of the shotgun signal into the mix (watch for comb-filter-effects, 3:1 rule[19]).[20] Another approach is to mount the microphone a little bit further down the chest, giving it a more open sound. Bassy chests or heartbeats on a lavalier audio track can be annoying. Using the microphone of another talent (when not too far apart) not only gives a more natural response, but also works as a low-cut-filter without any further equalization.[21]

[17] Viers, p. 67.
[18] see Viers, pp. 67–70.
[19] The 3:1 rule states that active microphones have to be three times as loud as inactive ones in close proximity to avoid phase cancellation.
[20] see ‘Microsoft PowerPoint - Rigging Lavs N Wireless Mics - Rigging Lavs N Wireless Mics.pdf’, p. 16 <http://filmtvsound.com/phocadownload/presentations/audio-technica/rigging%20lavs%20n%20wireless%20mics.pdf> [accessed 5 March 2014].
[21] see Viers, pp. 30,31, 69.

2. Clothing noise can either be contact noise or clothing movement. *Contact noise* means clothing rubbing against the microphone or cable. It is crucial to tape down both cable and everything that might rub against the microphone itself to reduce all mechanical noise. If garment near the microphone is immobilized, it cannot move in relation to it. A small loose cable loop right below the capsule serves as strain relief and to reduce handling noise. *Clothing noise* comes from clothing pieces or layers rubbing against each other. In particular, synthetic fabrics cause the most noise compared to quieter 100% cotton fabric. Sound mixers turn to the wardrobe department if the talent wears noisy jewelry.[22]

There literally are thousands of ways to mount a lavalier microphone on a talent. Due to clothing, camera perspective or a lot of movement, a lavalier cannot always be hidden in the 'sweet spot'; production sound mixers have to become very creative and innovative with their mounting techniques. Though the ultimate goal remains: to gather clean audio as best as possible.

In the following, the most common placements are being introduced. For the Schoeps Film Sound Application five of them were tested, examined and evaluated (see page 42).

Chest Mount

Mounting the lavalier directly on the chest is easiest with actors having large pectoral muscles. The microphone can hide right in the center about 20-30cm from the mouth. Using a microphone cage will help reduce clothing rustle. It is recommended to place a big piece of Band-Aid under the microphone to eliminate hair noise. With very hairy chests, veteran sound mixer Paul Ochsner explains how one can even place the microphone inside a little self-made pouch built of duct tape and two sticky-tape-triangles, adding some foam or wind-jammer underneath and the top side getting taped to the inside of clothing.[23]

With this technique heartbeats on the record may be an issue.

Bra Mount

A woman's cleavage is almost perfect to eliminate contact noise. The lavalier is mounted in the center where the two cups meet with either a vampire clip or a safety pin taped to the capsule. It is important to provide strain relief with a cable

[22] see 'Microsoft PowerPoint - Rigging Lavs N Wireless Mics - Rigging Lavs N Wireless Mics.pdf', p. 9,10; see Viers, p. 72,73.

[23] see 'Freelance Sound Mixers & Recordists for TV/Film' <https://www.facebook.com/groups/soundmixers/> [accessed 10 March 2014].

loop around the bra center strap. With less-gifted females, additional protection may be utile: A little bit of moleskin wrapped around the microphone body will keep the lavalier off clothing and pointed towards the chest. The cable routes underneath one of the breasts around to the backside.

Flat Mount

A very quick mount on cotton shirts is a side-address lavalier taped to the inside of the shirt with the shirt acting as windscreen. To reduce undesired noise caused by friction, the pickup element can be put in a rubber mount with a piece of moleskin facing the chest.

Placket Mount

With actors wearing a dress shirt, the spot between two buttons underneath the top layer lends itself to place a hidden lavalier microphone. The advantage is that it can be put inside a tape triangle, tilted a little to the side and stick out just a tiny bit. This way there is less contact noise – clothing noise is reduced by taping the triangle to both bottom and top layer of the placket. A small makeup sponge between the chest and shirt makes sure that chest hair noise is minimized. This placement can become obsolete when the shirt is made out of synthetics or silk.[24]

Dress Shirt Pockets or Lapels

Sometimes (only with big productions) it is possible to get microphones sewn into clothing. Production sound mixers have to work closely with the wardrobe department to find a solution that works for both parties. Lavaliers inside a dress shirt (or jacket) pocket often deliver clean sound as the microphone is not covered by clothing. It's put right under the lower edge of the pocket with a little bit of moleskin and wig tape. This strategy can be even further improved by using a hollow ink pen with the microphone hidden in the cap, pointing upward to the mouth. A similar placement is under the lapel of a jacket. Note that this placement risks dull sound for thick garments. Both applications need a little hole to be cut to get the cable from the inside out.[25]

Under collar

Another particularly good spot to conceal a microphone is under the collar. Herewith one can get rid of most contact noise, because the microphone (when

[24] see Viers, pp. 78–80.

[25] see 'Hiding a Lav Mic.. Good Info I Found..' <http://www.dvxuser.com/V6/showthread.php?203757-Hiding-a-Lav-Mic-Good-info-I-found> [accessed 10 March 2014].

wrapped with moleskin and/or tape) stays away from clothing. The only downfall is that the microphone sits behind and sideways to the mouth - the sound is dominated by low frequencies missing the higher ones which are important for speech intelligibility. Unshaven chins probably will cause a lot of friction.[26]

Neck of T-shirt

When thin or tight clothing would reveal a flat mount or a microphone mounted on the chest, most of the time the neck provides a thicker piece of fabric. The microphone is mounted with a vampire clip inside the neck of the shirt. Pulling the lavalier a little bit back inside the clip, or using a piece of blu-tack (known as Bostik in Germany), prevents the microphone from rubbing against the skin.[27]

Tie rig

If the talent wears a tie, a successful placement is on the tie itself, because noise from the tie rubbing or tapping against the shirt is unavoidable in case the microphone gets placed under the shirt. The microphone goes in a foam tie pod which keeps the capsule away from the fabric and hides inside the knot under the top layer. The two major problems with this rig is for one silk ties that will never remain calm considering friction noise, and two the talent's voice sounds "throaty" as many sound mixers report, meaning higher frequencies important for a rather natural sound are missing or attenuated.[28]

Outside, in plain sight

Sometimes picture framing, type or color of clothing, or camera perspective (very long shots) allow for a mount in a visible position. One should definitely use this opportunity because of very little chance of clothing rustle and full, natural frequency response. For some shots e.g. with almost undressed talents it is hard to hide microphone, cable and transmitter. Lavalier microphones can be taped to the brim of a hat, or on the side of eyeglasses with the cable running down the talent's neck. One has to keep in mind that due to the way sound waves behave, the sound will be containing less of the lower frequency range.

[26] see 'Videomaker - Hiding a Lav Mic - YouTube', sec. 2:02min <http://www.youtube.com/watch?v=_sNve5rNAMI> [accessed 10 March 2014].

[27] see 'Vclip Vampire Clip Demo for Sanken COS 11. - YouTube', p. 0:09min <http://www.youtube.com/watch?v=LvzXUyEcIIM> [accessed 10 March 2014].

[28] '700_dpa-4071-in-dmm0010-Tie-Pod.jpg (JPEG-Grafik, 700 × 398 Pixel)' <http://www.soundnetwork.co.uk/imagelibrary/700_dpa-4071-in-dmm0010-tie-pod.jpg> [accessed 10 March 2014].

The same is true for microphones resting in the front hair line. The benefit compared to lavaliers hidden under clothing is the more natural sound.
Extremely small lavaliers, like the Countryman B6, can be hidden in a necklace (can be bought as finished necklace)[29] , or behind a button, or are applied via the button hole. With a white response cap painted in the matching shirt color, they can’t be spotted while gathering clean dialogue.
In some occasions the microphone can even be mounted with an obvious tie-bar-clip as it is common for TV productions. The RØDE pinmic is a considerable alternative to typical lavalier microphones – although still seen, it can be camouflaged and seem hidden due to color, shape and size.[30]

All these applications have their pros and cons – every sound mixer has his or her preferences, each situation calls for a new kind of plan of attack. Generally the placements mentioned usually work fine until the talent turns their head. Even with omnidirectional lavalier microphones, sound can falls off, can become too thin and unusable immediately.

3. It is mandatory for a production sound mixer to verify the direction the talent is going to speak to in advance. This will determine the placement of the lavalier microphone. In special occasions two lavaliers prove to be useful (3:1 rule has to be kept in mind at all times).

3. Planted Microphones Positions and Techniques

Depending on the situation, the disadvantages of a lavalier microphone mounted on a talent outweigh the advantages. Plus many sound mixers are worried about giving their microphones out of hand with good reason: Not every talent handles them as carefully as necessary.
This is where planted microphones come into play. It goes without saying that any type of microphone can be planted anywhere on the set. But in order to match the sound of the primary microphone, the boom microphone, at best, many colleagues from postproduction suggest to use the same model of microphone for planted microphones

[29] see ‘Home’ <http://www.neckmics.com/Neckmics/Home.html> [accessed 11 March 2014].
[30] see ‘RØDE Microphones - PINMIC’ <http://www.rodemic.com/microphones/pinmic> [accessed 10 March 2014]; see ‘Videomaker - Hiding a Lav Mic - YouTube’, p. 2:35; see Viers, p. 81; see ‘Freelance Sound Mixers & Recordists for TV/Film’ <https://www.facebook.com/groups/soundmixers/> [accessed 10 March 2014].

as well. However, to cover a larger area (also known as ‘zones’ amongst production sound mixers), there is no way around a boundary layer microphone or a standard lavalier e.g. the Schoeps BLM 03 C as a boundary layer microphone, or the Tram TR-50 as a lavalier which both have a wider pickup pattern and are well known to be easily combined with the sound of boom microphones.[31] There are three main factors influencing the placement of planted microphones:

1. The inverse square law; [32] far distances result in unusable sound.
2. The microphones should always be placed in the direction the talent is facing (in front of talent, see first chapter).
3. If the microphone is in the picture, it has to be hidden (exceptions see above).

In the following, typical places for planted microphones are going to be introduced:

Tables & Desktop

Tables are a great place to deploy boundary layer microphones, especially to cover dialogue from several people as long as disturbing sounds like from hands on the table, silverware and dish rattles etc. do not become an issue. Lavaliers can be put directly on top of the surface or (especially unidirectional ones) on a shock mount built of a small sponge, blu-tack, or a double-sided loop of tape to absorb surface vibrations. A lavalier or boundary layer microphone can be hidden on a desk to cover the dialogue of a sitting next to a standing talent. Movements, clacking and loose rattling office items can be a problem and have to be factored in.

Doorways & Windows

In some cases talents have to deliver just one line of dialogue outside a window or from the outside through a doorway. Booming from overhead becomes a problem with all the nearby walls and ceilings. Another boom microphone hanging from the ceiling or placed at the wall above a window is the best way to cover the lines or words that would be difficult to reach with a boom.

Flat surfaces

A similar case to windows and doorways is an actor facing a flat surface such as whiteboards, televisions, cupboards etc. There might not be enough room to

[31] see ‘TramMicrophones.com - TRAM Lavalier Microphones and Tram Mic Clips’ <https://www.trammicrophones.com/> [accessed 11 March 2014].

[32] When distance between sound source and microphone is halved, the volume gets doubled (+6dB)

place a shotgun microphone overhead or sideways. A lavalier or boundary layer microphone that blends in well with the boom microphone can be hidden nearby the talent.

Car Interiors

Each car interior sounds quite different from other car interiors – it is up to age of the car, motor, speed, number of speaking talents, seats they are sitting in, overall situation (normal speech level or action scene with raised speech level) etc. when it comes to microphone selection and placement. Talents wearing lavalier microphones in a car works, but is usually not a good idea because the seatbelt might produce a constant rubbing noise and maybe even causes the microphone to come loose. It is a much better choice to plant the microphones on the dash board or the sun visor. A unidirectional microphone retains perspective, however a lavalier with an omnidirectional polar pattern, a BLM 03 C or a Sanken CUB -01 might just sound cleaner, clearer and more consistent when talents turn their heads. Production sound mixer Stéphane Bucher likes using a suspension system with two CMC 41 rigged to the rearview mirror. Other professionals tape an omnidirectional microphone directly overhead to the ceiling – it's a matter of taste and situation. A lavalier microphone taped to the back of the front seat head-rest suits well for backseat passengers. In a car scene with smaller talents, or worm's eye view, a lavalier microphone can be put on the gearshift, or a more directional microphone right next to it facing up. Whichever solution is chosen, it is always important that the microphones are in a fixed position and, to reduce disturbing rattling and jingling noises, one should take all unneeded loose items out of the car and anchor visually seen items.

Miscellaneous

Placements are only limited by innovative ideas and imagination: Under the hood of a car, in a vase, in a fruit basket, behind doorposts, inside boxes or strollers, suitcases, cellphones, etc. Most of the time the road to success is called trial and error.[33]

Backup

On one hand, location sound is mostly about dialogue. Ambiances, SFX etc. during shooting are also very important but only come second especially with

[33] see Viers, pp. 103–114; see 'Stéphane Bucher Production Sound Mixer – 2011 – October' <http://www.stephanebucher.com/2011/10/> [accessed 11 March 2014].

high-budget productions. On the other hand, a simultaneous set up and planted backup microphone for ambience can be very useful when it comes to background sound: Suppose a dialogue scene with an unwanted siren in the background. In postproduction the director finds it to fit in quite well. But on location it was only there for the master shot. Now what if the editor cuts to the close-ups? A microphone that is planted in the distance would pick up a clean copy of the siren without the dialogue. This track can later be mixed in with the 'dry' close-up shots if necessary.[34]

One should not rely on planted microphones in scenarios with travelling dialogue. It is key to keep the sound perspective equal to visual perspective, meaning if the talent turns or walks away from the camera, they should not turn or walk toward or past a microphone or vice versa. It not only sounds improper, but also makes the audience feel confused. The safest way to record travelling dialogue is using the boom microphone or a lavalier.[35]

To sum up this topic: As far as *angle* and *distance* are concerned, there are some average values to help create a crisp and intelligible sound:[36]

Microphone	Angle	Distance
Shotgun	≈ 30°	Fine up to 45-90cm
Long Shotgun	≈ 30°	Fine up to 120-240cm
Lavalier	Center of sternum	≈ 15-30cm
Planted	Depends	Depends; roughly 30-60cm

It takes a lot of experience, practice and time to test and play with different microphone types and placements in order to find out which ones work best. A very inventive solution to enhance the decision making process is a new audio software tool using augmented reality to display the directivity of a microphone in a live setup.[37]

[34] see 'Multi-Track Production Recording - Using Digital Disk Recorders to Improve Quality and Simplify Post Production', para. 3 <http://www.editorsguild.com/v2/magazine/newsletter/janfeb04/janfeb04_multitrack_prod_rec.html> [accessed 11 March 2014].

[35] see Viers, p. 119.

[36] see Viers, p. 42.

[37] see 'Arapolarmic Description Audio Tool | Aratechlabs Augmented Reality Audio Technologies' <http://www.aratechlabs.com/arapolarmic-description/> [accessed 10 March 2014].

4. Sound improvement

Warily selecting and placing microphones guarantees good sound recording results on soundstages. In the field on the other hand, recording clean and undisturbed dialogue is an entire different world. 'Treating' sound is difficult because sound waves are uncontrollable. They spread as spherical and plane waves, bounce, resonate and reverberate. The best way to find out about the acoustics and background sound problems of a location is to participate in location scouts – which is almost always beyond question since most of the time production sound mixers get involved pretty late into the preproduction process. There are four different types of noise on a film set:

Acoustics

'What you see is what you hear'. A gymnasium, a garage, a church etc. are all locations that have to sound the way they do; they will color the sound. This is good and will help give the impression that the talent seen in the picture is actually on-site. Yet reverb or coloration should never be the reason for unintelligible sound (reverb is considered noise, see first chapter). Both studios and big locations, such as large hotel rooms, hallways, foyers etc. can be made less reverberant by putting up acoustic sound blankets for sound absorption. Molleton, molitan, and blankets hanging from the ceiling or put on a C-stand, or mattresses put on the wall will help get a room a lot quieter. Often it's not so much the reverberation time but the time and intensity of the first reflection and the ratio of diffuse vs. direct sound (ping pong delay) causing noticeable problems. This occurs more frequently with smaller reverberant locations compared to larger ones. Then again it is a lot easier to tame smaller rooms. If reverb is on the dialogue track because the take happened in a large, untreated room, there is almost no way around ADR[38].

Another approach is to use a lavalier rather than the boom microphone, or to record wild tracks[39] after shooting.

While most the time the location itself is problematic, it is yet even more complicated the other way round. Artificially built sets and production designs are of thin and cheap materials. A set surrounded by plywood colored like brick walls, Styrofoam or papier -

[38] ADR = Automated Dialogue Recording

[39] Wild tracks are lines that could not be picked up properly during the take. As the talent is still 'in the scene' the line can be recorded afterwards and be synced in postproduction with software plug-ins like vocalign

mâché cut and shaped like rocks and a concrete floor painted as carpet, is not going to help a realistic acoustic. In this case hollow floors, walls, stairs can be dampened with foam, felt, fleece or other insulation material. Sound mixers prefer to lay down carpets or floor mats if not seen in the picture, or put sticky felt pads under a talent's shoe to make for a clean and consistent dialogue track – Foley footsteps are added in postproduction. Also, soundproof soundstages should not be taken for granted when they are next to railways or an airport, like it was case with the trilogy of *Lord of the Rings*.[40]

Ambiences

Traffic is the major problem causing undesirable background noise. Using a long shotgun or the SuperCMIT and making sure it is not facing the traffic sometimes really is the only way to control noise on exterior locations. As explained in the first chapter, an s/n-ratio of 15dB has to be kept at all times to ensure speech intelligibility. Working inside, all doors and windows need to be closed.

Air-conditioning, generators, heating, refrigerators, ventilators, LCD monitors, computer fans produce all kinds of noise almost across the whole frequency spectrum. Using an equalizer will therefore not solve the problem without affecting important frequencies of the dialogue. They have to be turned off if they are not part of the picture. The same is true for noisy camera fans during recording, like the RED EPIC being extremely loud. Trying to put over a blanket or other kind of cover is going to reduce the noise of devices in the picture.

Rain on the roof can be treated with either hay or pine needles put on top and molleton used on the inside.

Other annoying ambiances can be: Music (at shopping malls or train stations), light ballasts, clocks, neighbors, bystanders, crewmembers, pets, planes etc.[41]

Incidental Noise

Incidental noise cannot be predicted, but it certainly must be expected in the first place. Cellphones can interfere with hardwired and wireless audio signals and have to be

[40] see 'LOTR2 TT Sound Design Part 1 - YouTube', chap. 0:40min <http://www.youtube.com/watch?v=t5cOgj4RsWg> [accessed 12 March 2014]; see 'O-Tonaufnahmen in Riesigen Räumen' <http://www.bvft.de/forum/showthread.php/18-O-Tonaufnahmen-in-riesigen-R%C3%A4umen> [accessed 12 March 2014]; see Viers, pp. 263–267, 279.

[41] see Viers, pp. 267–276.

turned off. Squeaky chairs are handled with sticky felt pad. If jewelry is not particularly needed and likely to rub against a lavalier microphone, one should ask to get it removed. City hall and church bells have to be reckoned on making a sound every fifteen minutes. All kinds of noisy props like plastic bags may be replaced with paper bags (sprayed with misting spray). Dishware is damped by another layer of molleton underneath the tablecloth and small pieces of felt under cups and dishes. Some sound mixers share the opinion that hot glue under dishware is working just as fine and comes off fairly easily. Sometimes it is one's last resort in order to salvage undestroyed dialogue to ask the talent to separate the line of dialogue from the noisy action.[42]

Working Crew

Obviously all noises the crew makes are to be deadened. Creaky dollies immediately have to be addressed. If camera operator, dolly grip, director or boom operator (!) destroy the tracks with unwanted sound from movement or footsteps, they should try working without shoes or wear hush heels. In addition, boom operators can put on cotton gloves to reduce handling noise. Removing finger rings or bracelets is a must.[43]

All these disturbing noises can affect microphone selection, placement, angle and distance. It's an art form to find a solution without accepting too many compromises.

[42] see 'Profane O-Ton Frage' <http://www.bvft.de/forum/showthread.php/19-profane-O-Ton-Frage> [accessed 12 March 2014]; see Viers, pp. 276–279.
[43] see Viers, pp. 45, 49, 277.

|| Field Tests

For the Schoeps Film Sound Application (from now being referred to as SFSA), five field tests were done in order to compare the different performances of microphones and placements relevant in location dialogue recording.

To see and listen to them start the DVD, open the file "index_1.html" in the folder "App" only in your latest version of Microsoft Internet Explorer (IE11, released in November 2013). Make sure to click on GEBLOCKTE ELEMENTE ZULASSEN if the message "Das Ausführen von Active-X-Steuerelementen wurde für diese Website eingeschränkt" comes up. Otherwise it won't work! Click through the menu items on the left in order to get to all topics. The fifth item will lead you to the tests.

1. Test #1

The video was shot on a rather quiet interior location with a medium size. It is neither a reverberant nor super dry room and represents an average type of interior that production sound mixers run into on a daily basis. There was a carpeted floor to reduce impact sound and dialogue disruptive footsteps. The Schoeps CCM 2, CCM 21, CCM 4, CCM 41, CMIT 5 U, and SuperCMIT 2 U were set in an overhead booming position right above frame line. A second CMIT 5 U was applied from underneath at about 50cm from the floor. A Schoeps BLM 03 C was used on the table covered by a light sheet of paper. All microphones were set as closely as picture framing would allow and were always aimed at the speaking talent.

For all five tests this meant to find a compromising trade-off between bringing all microphones into the best booming position without having them interfere with each other by possibly blocking or shadowing sound waves reaching the diaphragms. The applied solution was to keep the space between microphones always bigger than the smallest wavelengths (>2cm).

In the first field test there are wide shots, medium shots and close-ups to show the different sound of each microphone considering polar pattern, frequency response, and placement in different angles and distances. Male and female voice provide for a balanced representation of frequencies; dialogue of intentional quiet, normal and raised speech level gives a helpful feel of how the microphones pick up the sound at different volume levels.

Use the dropdown list to switch between microphones. Checking the information box will turn on/off further description. In the right column you can open a gallery to see pictures from production. These will give you a better visual understanding of the microphone application. The video is looped so the user doesn't get distracted by a sudden stop while they are listening closely.

Results:

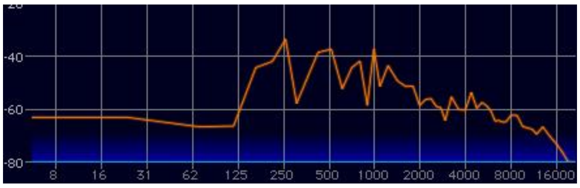
- There are noticeable on- and off-axis differences with unidirectional polar patterns when talent turns her head (0:10min).
- CCM 2 reproduces diffuse sound a lot more than the directional microphones. There already is a big difference to the subcardioid.
- Speech sounds a little bit colored with shotgun microphones when talent stands up and speaks loudly (1:15min). The microphones were too close to the ceiling where they run the risk of recording reflections that are not cancelled out, especially because of the tail in the polar pattern. Cardioid or supercardioid come out ahead - the little amount of reverb on their tracks make the recorded voices sound crisp and very natural.
- The BLM reads a distinguishable loss of high frequencies due to the paper sheet covering it, which is unfortunate. It's probably better to not cover it and accepting the downfall of a greater distance. Then again, when the talent is standing up, it definitely is too far away (>90cm) from the sound source. A way to fix this would be to place it in an overhead position on the ceiling.
- This table shows the average noise floor level of each microphone while speech was kept at the same level for all microphones. It is the averaged sound pressure level from several measurements at different points and positions of the microphone. It is remarkable to see how low noise is for the SuperCMIT and CCM 41. The difference of 3dB between CMIT and SuperCMIT approves the quality and power of the SuperCMIT.

Microphone	Noise Floor
SuperCMIT	-56,4dB
CMIT lobe	-53,4dB
CMIT scoop	-55,5dB
CCM 41	-57,2dB
CCM 4	-54,6dB
CCM 21	-47,5dB
CCM 2	-41,4dB
BLM 03 C	-40,8dB

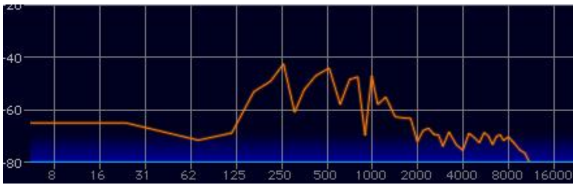
- The CMIT from underneath is an unfavorable placement. It picks up too much reverb (comparable with CCM 2) while the male talent is standing. The voice of the female actress sounds almost completely off-axis since the microphone had to be placed sideways and a little bit too far behind. As the male talent leans back, there was more space to aim the microphone toward him. However, the sound is inconsistent and rather dull altogether. Figure #15 shows very clearly the loss of frequencies from 2 kHz and greater compared to the overhead CMIT and the overall extremely natural sounding CCM 4. All frequency spectrums show the peak curve of a measurement over a period of time in RMS[1] detection mode.

Figure #15

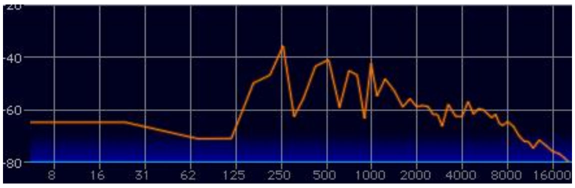
CMIT from above with speech



CMIT from below with speech



CCM 4 from above with speech



2. Test #2

The video is to demonstrate the behavior of the microphones in different angles toward a big source of noise in the background. It was shot right next to the autobahn as a very loud source of noise. Three shots total were made to show the directivity behavior of the microphones in different angles to the source of noise: First shot toward the noise (0°), second shot sideways to noise (90°) and third shot away from noise (180°). All microphones were positioned as closely as possible right above frame line in a typical booming position. A lavalier microphone, the DPA 4071, was mounted on the talent inside the scarf (see cable as evidence). A Neumann KMR 82 i serves as a high-end and well-established long-shotgun-microphone for film sound. None of the signals was further being equalized or processed in postproduction. The SuperCMIT was set to Preset 1. Each microphone had a foam windscreen to eliminate wind distortion

[1] RMS = Root Mean Square displays the energy as being averaged over time

without changing the original microphone signal too much; the DPA had a typical lavalier windjammer.

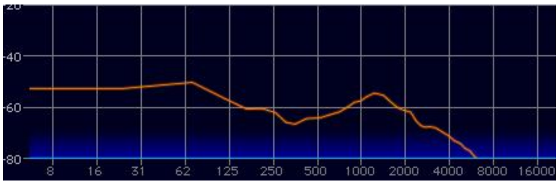
Again the talent speaks at different speech levels to get a better view of each microphone behavior at different volumes. Unfortunately it started to snow during the 90° and 180° angle shot. To fix this, the crew would have had to redo the whole test, equipment would have had to be rented again which both was way beyond cost, time and effort for this thesis.

Results:

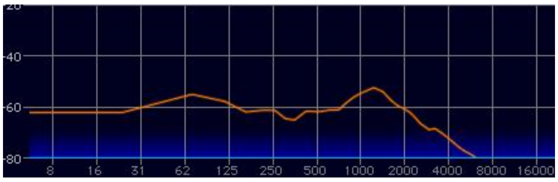
Microphone	**0° to noise (noise level in dB)**	**90° to noise (noise level in dB)**	**180° to noise (noise level in dB)**
SuperCMIT	-29dB	-34,4dB	-39,5dB
CMIT	-30dB	-32,8dB	-36dB
KMR 82 i	-30,3dB	-30,4dB	-37,2dB
CCM 41	-30,5dB	-31,3dB	-33,5dB
CCM 4	30,3dB	-33dB	-34,6dB
CCM 2	-27,2dB	-26,6dB	-26,1dB
DPA 4071	-53dB	-47,3dB	-44,9dB

- In the shot with the talent's back facing the autobahn, the lavalier delivers the best results considering noise level. This is because the body and the scarf work as perfect elements to block all unwanted sound from behind. Noise increases in the other shots tremendously as the omnidirectional polar pattern faces the noise directly.
- The SuperCMIT seems to have a surprisingly high noise level in the first shot (0°). This is in its nature as it reduces the noise from the sides, but 'sucks in' the noise coming from the front to a higher extent than less directional microphones.
- Both SuperCMIT and KMR 82i indicate an enormous decrease of noise floor level comparing all three angles 0°, 90° and 180° to the source of noise. CCM 41 and CCM 4 show only slight differences of picked up noise from 90° compared to 180°. Their pickup pattern is still too wide for a noisy environment to say nothing of the CCM 2 which only was included to have an unweighted and non-directional reference signal.
- The SuperCMIT can handle undesirable low frequencies much better than the conventional Neumann long shotgun (see figure #16 and level at 90° angle). This supports intelligibility. Figure #16

Neumann KMR 82 i only noise



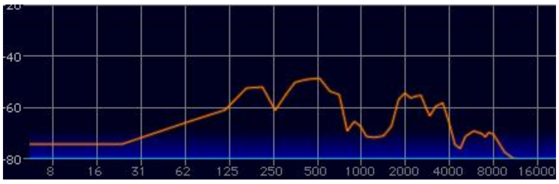
Schoeps SuperCMIT only noise



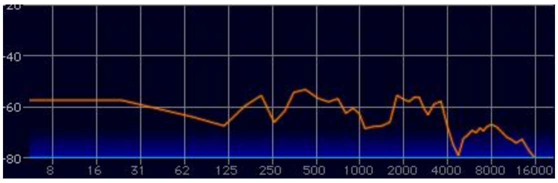
- Speech intelligibility is not ensured at all times. In the 0° shot all microphones (besides lavalier) are suboptimal e.g. at 01:24min. At 01:11min there is a semi-truck in the background. The CCM 2 and CCM 4 still have problems to reduce the noise level to a point at least 15dB lower than speech level although pointed straightly away from the source of noise (180° angle).
- The angle of 90° proves not to be ideal as the unidirectional microphones still not only pick up too much of the noise from the front left, but also have the tail of the polar pattern maybe picking up unwanted noise from the rear left. The angle of 180° proves to be best for all microphones except the DPA 4071. A higher-angled microphone prevents the tail from picking up sound from behind. The lavalier definitely still has an outstanding s/n-ratio and attenuation of low frequencies at 180°, but at the cost of losing high frequencies. With that the signal is a less natural reproduction of the voice.

Figure #17

DPA 4071 with speech



Schoeps SuperCMIT with speech



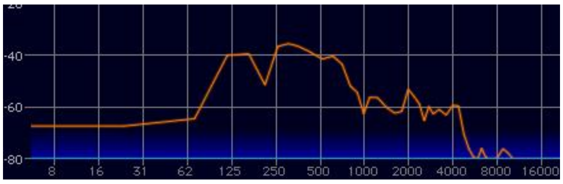
3. Test #3

The third comparative field test shows the sound of different lavalier microphone placements mounted on the talent. They were all planted hidden and invisible to camera. In order to tell the difference between boom and lavalier microphone in an even better and improved way, only the male talent wears a lavalier microphone. The actress remains always on the CCM 4 signal. This may create small ‘jumps’ in room tone because the CCM 4 delivers a rich natural sound as opposed to the lavalier microphones remaining more sterile. In a regular postproduction process, the dialogue editor would either keep the CCM 4 track throughout the whole scene and reduce it about 3dB in volume whenever the lavalier track blends in (only done if there is no off-mic dialogue on the boom channel), or she would fade it directly with the lavalier track but add a little bit of boom room tone to the lavalier track.[2] For the SFSA none of this was done, because it was the intention to keep the lavalier tracks clean and untreated in order to get a full demonstration of how they sound nakedly. The microphone used for the video is a DPA 4071 which does not represent the standardized IRT frequency response[Appendix figure #6] – however it brings a 6dB boost to the upper frequency range. The original signal was not further equalized or processed.

Results:

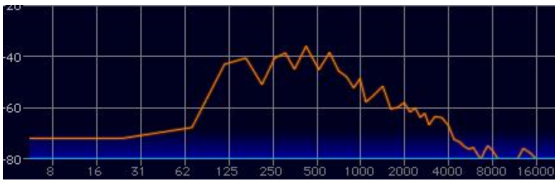
Figure #18

Under collar



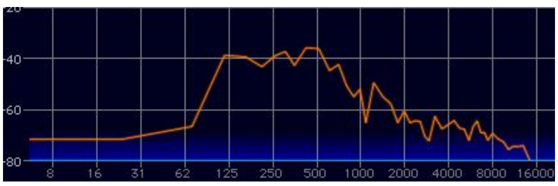
- The placement under the collar indeed sounds muffled – the frequency spectrum shows a first drop at 800 Hz and a second immense drop at about 5 kHz. The advantage with this placement is a minimum of clothing rustle for most of the time.

Front hairline



- The placement in the front hairline works great and benefits from almost no rustle or crackles. The drawback is that, depending on camera position, this placement cannot be used with a visible cable running down the talent's neck. The overall sound tends to be a little bit nasal and misses the lower frequencies, which would give the signal a warmer, natural sound.

Front pocket



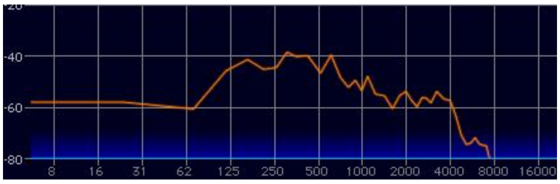
- A lavalier placed inside the front pocket of a dress shirt sounds rich, round and very direct. The frequency spectrum is balanced since no frequency bands get lost due to position or clothing covering the capsule. Note that the jacket over the shirt may cause rustling problems when moved.

---

[2] see John Purcell, *Dialogue Editing for Motion Pictures: A Guide to the Invisible Art* (Amsterdam ; Boston: Focal Press, 2007), pp. 162–166.

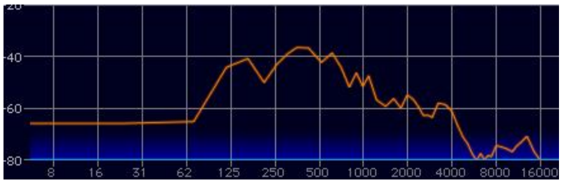
- Mounting the microphone directly onto the chest contains two problems: The placement is risky for contact noise (especially with the talent’s synthetic dress shirt) and the signal gets the so-called ‘telephone-effect’ (weak incline of frequency response curve between 70-200 Hz, missing out the power of the low frequencies of the voice; typically cutting off at roundabout 4 kHz)[3] because the shirt acts as high-cut-filter. Moreover (as shown in the graphic), low frequencies below usual speech frequency range (<60 Hz) are somehow boosted by the chest as a resonator for low frequencies (see first chapter). This can be some rumbling noise irrelevant for speech and is therefore undesirable.

Chest mount



- The placket mount is a common placement and proves to be a very good mount as far as risk of clothing rustle and frequency response are considered. Compared to the position inside the pocket, it’s missing a little bit of the high frequencies around 6 kHz which would make for a rich sound.

Placket mount



4. Test #4

This test demonstrates the directionality or directivity behavior of both the voice and the microphones. The BLM 03 C was placed underneath a light sheet of paper on the table in front of the talent. All other microphones were set in a typical booming placement and aimed toward the center sitting position of the talent in the same exact angle. In all other tests, the upper frame line of the picture determines the distance of the microphones, with the logical consequence that the diaphragms of CMIT and SuperCMIT are always about 10cm further away from the sound source than the other microphones with no tube. Keeping the same angle as well as the same distance for all microphones was crucial for this test in order to have a solid and realistic demonstration of their directivity behaviors. The picture from production shows how

[3] Also see default telephone filter in “Audacity”

the diaphragms were lined up at roughly the same height. The distance of the BLM 03 C is in approximate accordance with the distance of all other microphones.
The test gives an accurate overview of the different on- and off-axis responses of the polar patterns. Therefore the talents don't only speak in the sweet spot, 100% on axis, but to all sides 360°, up and down in different distances to the microphone setup. Their voices intentionally change in volume to get a better idea and feel for on- and off-axis response of different speech energies. "On-axis sounds are bright and crisp, while off-axis sounds are more flat with less high end"[4]. The test conveniently shows how much inaccuracy each and every polar pattern 'forgives' when cueing with the boom. Furthermore: Sooner or later every production sound mixer will run into a situation recording dialogue slightly off-axis, but with the benefit of fully rejecting the majority of excessive noise in the background. Therefore: knowing the off-axis sound of a microphone in advance tunes one's ear.
The user can toggle between microphones and listen to their performance. All Schoeps microphones are appreciated for their off-axis sound: Sound coming from the so-called rejection zone appears only to be quieter, but not colored (duller with less treble) as it is the case with cheaper manufactured microphones. Especially the CMIT remains unbeatable among other shotgun microphones since "the pick-up angles at low and high frequencies are kept reasonably similar to one another"[5]: The directivity is comparatively high for medium frequencies and the polar pattern does not get too narrow for high frequencies (see polar pattern).[6]
The table shows the attenuation of some positions and directions the talents are speaking to.

Microphone	On axis toward the microphone	Away from microphone (to the windows behind)	Downward to the side (trashcan)	To the side, about 1m of distance to the mic	Positioned to the side but speaking toward the mic
SuperCMIT	- 13dB	-29dB	-28,3dB	-24,9dB	-36,3dB
CMIT	- 13dB	-26,8dB	-28,2dB	-24,5dB	-29,3dB
CCM 41	- 13dB	-23,8dB	-27,2dB	-24,9dB	-30,5dB
CCM 4	- 13dB	-23,5dB	-25,6dB	-24,6dB	-33,7dB
CCM 21	- 13dB	-25,1dB	-25,7dB	-23,3dB	-27,1dB
CCM 2	- 13dB	-20,2dB	-22,8dB	-20,9dB	-26,4dB
BLM 03 C	- 13dB	-23,2dB	-19,2dB	-22dB	-33,5dB

[4] Viers, p. 19.
[5] Yewdall, p. 76.
[6] see Yewdall, p. 76; see Viers, p. 19.

Results:

- There is a lot of attenuation when the talents speak completely away from the microphone. In general, the more directional a pick-up pattern, the merrier the attenuation.
- If the talent delivers a line speaking downwards to the floor, a planted microphone is a must (third column, male talent at 00:40min).
- Booming sideways (fourth column, female talent at 00:47min) is still passable for most unidirectional microphone polar patterns, but of course there is a decrease of ~3-4dB in level.
- The fifth column displays the directive efficiency of each microphone very well – inattentive booming with directional microphones is fatal.
- There is an overall noticeable decrease in volume when the talent gets further away (⇒ common challenge for all boom operators ‘not to be too short’ colloquially, meaning not close enough to the talent.
- At 00:46min the male talent leans too far forward – the microphones are now pointed on top of his head, or even behind him. He immediately sounds off-axis on the unidirectional microphones – they have to stay in front of the mouth at all times.

5. Test #5

The last comparative field test was done to show the directivity factor γ and the distance factor of each of the microphones used in dialogue recording on location. The REE = Random Energy Efficiency is a ratio of a microphone’s response to diffuse energy relative to a response from on-axis. In other words: One can find out about the amount of noise reduction of a directional microphone when a directional microphone and a non-directional microphone are exposed to the same sound power while microphone sensitivities are equal. For omnidirectional microphones this factor equals 1, because there simply is no existent directivity. The table shows the REE factors for other polar patterns.

Polar Pattern	**REE (1/γ)**	**Directivity Index in dB $10 \times \log \gamma =$**	**Distance Factor $\sqrt{\gamma}$**
Omnidirectional	1	0dB	1
Subcardioid	0,481	3,17dB	1,44
Cardioid	0,333	4,77dB	1,73
Supercardioid	0,268	5,72dB	1,93

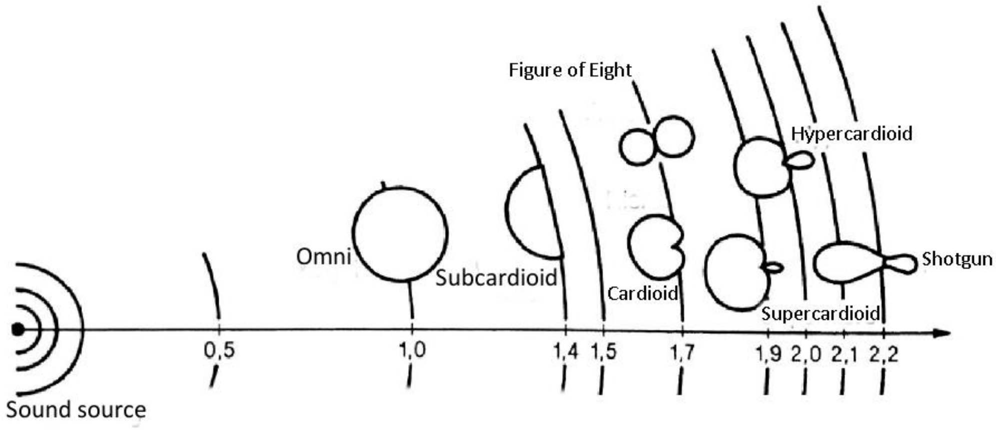
So for example a cardioid receives only 1/3 of the sound power compared to the omnidirectional microphone. The effect of this is that in a diffuse sound field the sound pressure level of a cardioid is 4,8dB below the SPL of the omnidirectional microphone in the same diffuse field. With the increase of the directivity factor γ (decrease of REE) the microphone becomes less responsive to diffuse sound. This in turn means that the unidirectional microphone can be placed further away from the sound source while the ratio of diffuse and direct sound stays the same at a certain distance. This distance is calculated by the distance factor $\sqrt{\gamma}$. If the distance between sound source and omnidirectional microphone is 1m, the distance between sound source and cardioid can be 1,44m. For shorter distances, the portion of direct sound to diffuse sound will increase, for greater distances it will decrease.[7]

In film sound this is called the 'reach' of a microphone. In this test the reach of different microphones was examined. The location was a small church with a wooden ceiling and wooden pews. This is why the reverberation time of 1,3s is fairly short. It was measured experimentally with the impulse of a clapper board. The omnidirectional microphone was always placed right above frame line. The determined distance between the sound source and the omnidirectional microphone was then to be multiplied by the different distance factors in order to set up all directional microphones. It is not easy to figure out the correct distance factors for shotgun microphones, as they differ in tube length and original polar pattern (supercardioid vs. hypercardioid). For most shotgun microphones the distance factor of 2 or 2.1 is very accurate according to technical literature.[8] For this test, the number 2.1 was chosen.

[7] see 'Buendelungsgrad Und Buendelungsmass Der Richtmikrofone - BuendelungsgradBuendelungsmassMikro.pdf' <http://www.sengpielaudio.com/BuendelungsgradBuendelungsmassMikro.pdf> [accessed 14 March 2014].

[8] see Dickreiter, pp. 31–33.

Figure #19[9]



The beamforming algorithm of the Schoeps SuperCMIT increases the directivity index to 11dB (in Preset 2 to 15dB). As a result, the REE of 0.08 equals a distance factor of 3.5 in preset 1[10]. The way from diaphragm to 15cm in front of the mouth (this is where the microphone is aimed at) was taken as distance in all shots. All shots happened inside the critical distance which is calculated as follows:

$$d_c \approx 0{,}057 \sqrt{\frac{V}{RT}} \quad [\,m\,]$$

RT, or also called RT_{60}, is the reverberation time, often using Sabine's reverberation formula[11]. The church has an approximate volume of 450m^3 which makes $d_c \approx 1.06$m. The following table shows the distances, distance factors and delay times referring to the omnidirectional microphone of each shot.

Establishing & Ending Shot; 67cm to sound source

Microphone	Distance (to Omni)	Delay
CCM 2	0cm	0ms
CCM 4	47cm	1,37ms
CCM 41	60cm	1,75ms
CMIT	73cm	2,13ms
SuperCMIT	167cm	4,84ms

[9] Dickreiter, p. 32.
[10] see 'Paper_SuperCMIT_Development.pdf', p. 7.
[11] See further reading

Medium shot with both actors; 76cm to sound source

Microphone	Distance (to Omni)	Delay
CCM 2	0cm	0ms
CCM 4	53cm	1,55ms
CCM 41	68cm	1,98ms
CMIT	84cm	2,45ms
SuperCMIT	190cm	5,66ms

Over (his) shoulder; 23cm to sound source

Microphone	Distance (to Omni)	Delay
CCM 2	0cm	0ms
CCM 4	16cm	0,47ms
CCM 41	21cm	0,60ms
CMIT	25cm	0,74ms
SuperCMIT	57cm	1,66ms

Over (her) shoulder; 36cm to sound source

Microphone	Distance (to Omni)	Delay
CCM 2	0cm	0ms
CCM 4	25cm	0,74ms
CCM 41	32cm	0,94ms
CMIT	40cm	1,15ms
SuperCMIT	90cm	2,62ms

Results:

- The demonstrations works not as well, because as soon as the talent moves or turns (which is quite typical for actors) the fine calculation fails since the microphones were in a fixed position and could not be moved by real boom operators.
- There is a very disturbing hiss on the audio track of the SuperCMIT making every cut, fade and transition noticeable in the mix. For a real film the changes could typically be covered with ambiance or Foley sounds. But the hiss itself would still be there. The problem has already been noticed during shooting.
- The test gives a good view of how much more reach the SuperCMIT has compared to the CMIT or the CCM 41.

The two topics ‘Wind protection’ and ‘Recording for the ideal post-pro workflow’ are on the website as future ideas.

Creating the Film Sound || Application

“No one will be using Flash […]. The world is moving to HTML5”[1] said Steve Jobs in 2010. Today, four years later, the development status of the new web standard HTML5 shows: Jobs was probably right.

Hyper-**T**ext-**M**arkup-**L**anguage (HTML) has always been the markup of the World Wide Web. In the past, HTML described documents of a number of types semantically; and to a certain extent even design could be added and determine the presentation of content.[2] Now a big mess started to happen, mixing up style and structure into one document. The solution was to separate the two – the idea of CSS (Cascading Style Sheets) was born in 1996.[3] CSS3 has become the standard and is still under development in 2014.

However, back then the area referred to as Web Applications had not been addressed for many years. Until the invention of the external plug-in called Flash by Macromedia, there was a real jungle of web players allowing browsers to show animated content. In the year of 2005 Adobe bought Macromedia and made the Adobe Flashplayer the internet standard for all animations, video and audio content on the web.[4]

In the meantime, HTML went through several different stations and changes; from HTML to HTML 4, XHTML, and HTML DOM Level 2. After a W3C workshop in 2004, “Apple, Mozilla and Opera were becoming […] concerned about the W3C[5]’s direction with XHTML”[6] – so some individuals of the three companies founded the WHATWG[7]. Their mission was to refocus on the needs of real world authors and they developed a first version of HTML5 specification. In 2006 the W3C showed interest again and was cooperating with the WHATWG until 2011. Now they are going separate ways again

[1] ‘Jobs: “No One Will Be Using Flash. The World Moves to HTML5”’ <http://www.rossul.com/2010/blog/jobs-%E2%80%9Cno-one-will-be-using-flash-the-world-moves-to-html5/> [accessed 16 February 2014].
[2] see ‘1 Introduction — HTML Standard’, para. 1.3 <http://www.whatwg.org/specs/web-apps/current-work/multipage/introduction.html#is-this-html5?> [accessed 16 February 2014].
[3] see Eric A. Meyer, *Cascading Style Sheets: The Definitive Guide*, 1st ed (Beijing ; Sebastopol, CA: O’Reilly, 2000), p. 9.
[4] see ‘HTML5: Das Web von Morgen - CHIP’, p. 1 <http://www.chip.de/artikel/HTML5-Das-Web-von-morgen_41539437.html> [accessed 16 February 2014].
[5] World Wide Web Consortium
[6] ‘FAQ - WHATWG Wiki’ <http://wiki.whatwg.org/wiki/FAQ#What_is_the_WHATWG.3F> [accessed 16 February 2014].
[7] Web Hypertext Application Technology Working Group

as they are heading towards different goals of what HTML5 should be. The W3C wanted the HTML5 specification to be a finished version despite some known problems; yet, they have been copying bug fixes from the WHATWG who is more focused on keeping the HTML5 specification a living standard “adding new features as needed to evolve the platform”.[8]

There are many, but the three main reasons to the development of HTML5 are as follows:

1) It combines what had been specified in HTML4, XHTML1 and DOM2 HTML in one document by being completely backwards compatible. So now there is only one doctype declaration `<!DOCTYPE html>`.

   This declaration will forever stay the same for all upcoming versions of HTML.
2) It’s designed to run cross-platform: PC, Tablet, Smartphones, and Smart TV.
3) It does no longer require additional plug-ins like Flash to deliver media and will reduce the need of plug-ins in general; in fact, with HTML5 authors can build complicated web applications using markup, not scripting.[9]

HTML5 is already here, but still under construction. According to a plan by the W3C it’s not going to be fully established and working until the end of 2014. The following version HTML5.1 is going to be released a stable recommendation in 2016.[10]

Furthermore not all browsers are ready to display the rich amount of features coming with HTML5 just yet. Tests and research that were done for this thesis proved Google Chrome as the most supportive browser for HTML5 in general. Mozilla Firefox, Microsoft Internet Explorer, Apple Safari, and Opera Software’s Opera browser (in that order) are still behind. But with their continuous updates, they more and more add HTML5 APIs and elements in order to keep up with the time and to be ready come December 2014.

It was the request from Schoeps to focus on and use this latest future-oriented technology with its many advantages for the Schoeps Film Sound application. As mentioned earlier, they want it to be a work in progress with changing and developing content over time. Unlike the Schoeps Showroom (which is a Flash-Plug-In and therefore difficult to change as it is a finished piece of work) this new application should

[8] see ‘1 Introduction — HTML Standard’, para. 1.6.

[9] ‘HTML5 Introduction’ <http://www.w3schools.com/html/html5_intro.asp> [accessed 17 February 2014].

[10] see ‘Plan 2014’ <http://dev.w3.org/html5/decision-policy/html5-2014-plan.html> [accessed 17 February 2014].

allow authors to interchange and redesign existing sections and add new material very easily in order to stay up to date at all times.
In this chapter we are going to look at the HTML5 elements used for the Schoeps Film Sound Application and their implementation in specific. Lastly we will learn how they combine their potential with the written content to make the application a useful online resource as a whole.

1. Elements

For the SFSA three HTML5 elements were needed: The video, audio and track element. All three elements are very powerful and offer a huge number of attributes, properties and methods. Covering all of them would go beyond the scope of this thesis, so we are only going to discuss the ones that were needed for the SFSA.

Video

Until Opera introduced the first version of the `<video>` element in 2007, videos could only be used with the `<embed>` or `<object>` tag which then accessed the Adobe Flashplayer.[11] With HTML5 and its `<video>` tag it is really intuitive to embed a video on a website. The following code displays a 320 x 240 pixel video player:

```
<video width="320" height="240" controls>
      <source src="movie.mp4" type="video/mp4">
      <source src="movie.ogg" type="video/ogg">
Your browser does not support the video tag.
</video>
```

With the @controls attribute HTML5 now detects the video controls needed and provides a standard controls section automatically. In this case there would be a play/pause-button, duration-slider, volume control and fullscreen-button.
In this example there are actually two videos as `<source>` elements with different formats. This is important because depending on the browser not all formats are supported. The table below shows the five big browsers and the formats and codecs they support:

[11] see ‘HTML5 Video’ <http://www.w3schools.com/html/html5_video.asp> [accessed 18 February 2014].

Figure #20[12]

Browser	Nightly	Release	Formats
Safari	November 2007	March 2008 (Safari 3.1)	MP4 H.264/AAC
Firefox	July 2008	June 2009 (Firefox 3.5)	Ogg Theora, WebM
Chrome	September 2008	May 2009 (Chrome 3)	Ogg Theora, MP4 H.264/AAC, WebM
Opera	February 2007 / July 2008	January 2010 (Opera 10.50)	Ogg Theora, WebM
IE	March 2010 (IE9 dev build)	September 2010 (IE9 beta)	MP4 H.264/AAC

In case the chosen browser can load neither .mp4 nor .ogg files there will be a short note as fallback content. If one of the videos can load of course no message will be displayed. Fallback content is very important to tell the user why they don't see what they had expected. This way they know that it's the browser's fault and not theirs. Authors can also refer to a flash video etc. as fallback content

<u>Audio</u>

Basically the audio element works the same as its sibling the video element which makes it very practical and handy. Of course we now don't have any visual display.

```
<audio controls>
     <source src="horse.ogg" type="audio/ogg">
     <source src="horse.mp3" type="audio/mpeg">
Your browser does not support the audio element.
</audio>
```

Like the video element a basic audio player is displayed bringing along a play/pause-button, duration-slider and volume controls slider. Audio codecs supported by major browsers are shown in the appendix.[Appendix figure #7]

[12] Silvia Pfeiffer, *The Definitive Guide to HTML5 Video*, The Expert's Voice in Web Development ([Berkeley, Calif.] : New York: Apress ; Distributed to the book trade worldwide by Springer Science+Business Media, 2010), p. 6.

Track

Audio or video descriptions can be very useful to explain content to deaf people, showing as subtitles that can be activated and deactivated as needed. Extra information provides a look behind the scenes and can be written as captions. In-band captions are provided as separate tracks in the media resource itself. The problem though is that browsers need to support multitrack media files which still is an issue to be fixed in some major browsers.

Out-of-band captions on the other hand are external text descriptions that are linked to the media resource through markup. The browser is now only required to load the content and synchronize it with the main media element during playback. The track element in fact allows for synced additional information within a media element such as video.[13] It is an external timed track which cannot represent anything on its own. Therefore it sits inside the `<video></video>` tag just like the `<source>` element.

Various attributes like @kind, @src etc. define the track.[14] There is a default setting to styling and animation which can be easily varied if necessary (e.g. color code for different speakers etc.). In general there are two ways, or formats respectively to link subtitles and captions to a media file: WebVTT or TTML.[15] WebVTT-files are UTF-8-formatted text files. An example shows a subtitle from position 00:00:01:000 (1s) to position 00:00:6:00 (6s into the video):

```
WEBVTT
00:00:01.000 --> 00:00:06.000
Hello World
```

TTML is an XML-based language and more complicated and circuitous to include. Compared to WebVTT it is directly written as markup (simplified demo) [16]:

```
<p begin="00:00:01.000" end="00:00:06.000" >Hello World</p>
```

[13] see Pfeiffer, pp. 248–251.

[14] 'HTML/Elements/track - W3C Wiki' <http://www.w3.org/wiki/HTML/Elements/track> [accessed 18 February 2014].

[15] Web Video Text Track or Timed Text Markup Language

[16] 'Erstellen von WebVTT- Oder TTML-Dateien Mit Caption Maker (Windows)' <http://msdn.microsoft.com/de-de/library/ie/jj152136(v=vs.85).aspx> [accessed 18 February 2014].

2. Using the Elements

As the Schoeps Showroom had proven in the past, “listening to” microphones by activating and deactivating them while an audio file is playing can be a very neat and helpful way to show their performance in each setup.
So why not do the same for film sound except combined with video in order to make it an application as realistic as possible. There were three main criteria to be met:

1) The video must not be a download option that opens in an extra pop-up window. Instead it has to be embedded in the main webpage. Nevertheless, technical video problems such as preload, slow buffering, or even judder would distract the user from listening to the recordings and would call for another solution.
2) As it is one of the major strengths of the Schoeps Showroom, the possibility to switch between microphone signals has to happen as smoothly as possible. Gaps, out-of-sync-problems, any type of glitches etc. were absolutely no option.
3) The video needs to provide extra information without having to leave the video to go to another page e.g. technical specification, steps and approach to the recordings themselves (peculiarities and noticeable problems), usage and application of the microphones during the session.

The first and foremost question that had to be dealt with was how to synchronize a video track (preferably in HD) with multiple audio tracks. HTML5 offers several different ways to do that, however not all of them are effective and most of them don’t get the browser support needed to meet all three criteria mentioned above. To understand why the solution that is now implemented in the SFSA one should know some initially considered coding. Syncing media in HTML5 seems to be quite easy, but as we will see later: It’s always the browsers that have more pull.

Solution #1

The easiest way is to implement a video element and several audio elements, one for each audio track needed. With an attribute called @mediagroup it is possible to sync all players together, making the audio players slaves to the video player. So when the user hits the play button of the video element, all other players start to play along. Moreover, the user is able to have full control over volume and the duration slider position by just changing the one of the video element. To make this solution work in a neat way, the Boolean attribute called @controls muted would have to be set to all but one audio player to ensure that only one track is activated at the beginning. All others would be deactivated though playing. By setting the @hidden attribute on all audio players, they continue playing (both activated and deactivated) but cannot be seen. The desired audio track is chosen by clicking a separate button of a button list. This button works like an activation of the selected track and automatic deactivation of all other tracks. As far as markup goes, the button needs to be equipped with an onclick event attribute referring to e.g. function button1 in the included javascript code within the `<script></script>` tag. The function handling the activation process would look like this (simplified demo):

```
function button1(){
        audio1 = document.getElementById("audioplayer1");
        audio2 = document.getElementById("audioplayer2");
                audio1.muted = false;
                audio2.muted = true;
                 }
```

This solution has been tested for this thesis with the following results:

Pros	Cons
The video doesn't contain audio ⇒ downsize of data; no danger of failure	Runs only smoothly on Firefox and Chrome
Audio tracks are relatively in sync with the video	Starts to easily come out of sync when switching too often and too fast ⇒ danger to fail
Easy implementation	Both browsers don't support (multiple) WebVTT tracks (Firefox only from version 28, current is 27)[17].

Solution #2

The next approach that suggests itself is to extend function button1 by a sync function. This function handles synchronization between video and audio players. So when the activation/deactivation button is clicked @onclick links to function button1() as

[17] see 'Track - HTML | MDN' <https://developer.mozilla.org/en-US/docs/Web/HTML/Element/track> [accessed 18 February 2014].

described above which is now pointing to another function sync() that is in charge of updating the current time of the audio player to the one of the video player. The javascript function would then look like this:

```
function sync(){
     var audio = document.getElementById("audioplayer1");
     var video = document.getElementById("video");
     audio.currentTime = video.currentTime; }
```

Tests show that in IE, Chrome and Firefox the currentTime property only updates to a certain number of times, so depending on how the user is interacting with the activation/deactivation buttons, video and audio come out of sync by about a second or more. The way to circumvent this would be a timer within function sync() that checks for sync every 10 or 100 milliseconds:

```
setTimeout(function(){sync()},100);
```
[18]

The only downside that comes with this is since setTimeout() constantly finds the players being out of sync, the audio is getting chopped up and interrupted by little breaks not allowing for a smooth monitoring.

Solution #3

As described earlier, both video and audio element come with a standard set of controls needed for playback. However, these controls can be hidden (see 'Solution #1') and replaced by a custom made controls bar. Keeping this in mind, it is possible to create only one play/pause button, progress- and volume slider etc. controlling video and all audio elements simultaneously via javascript. HTML5 contains a bunch of applicable properties and methods to get very satisfying results.[19, 20] This solution is the successor of solution #2 and combines it with solution #1 since firm and uninterrupted

[18] The setTimeout() method calls a function (in this case sync()) after the specified number of milliseconds (100).
[19] see 'HTML/Elements/video - W3C Wiki', para. Media Elements <http://www.w3.org/wiki/HTML/Elements/video> [accessed 19 February 2014].
[20] A list of attributes and methods used for this thesis is found in the appendix.[appendix figure#8]

fully in-sync audio is guaranteed. So no matter at which point the progress bar of the video is, the progress bar of the audio tracks automatically follow.

But: There is still a slight offset noticeable when switching quickly between tracks. Also there is still no support for the track element for Firefox and Chrome not to mention Safari.

Note from February 18, 2014: As of today, this method runs mostly smoothly and in-sync in Firefox and Chrome. However, it will not be considered a complete solution since the browsers still have no full WebVTT support, but will be offered to Schoeps as a working solution for Chrome and Firefox.

Open the file Analyzer/index_chrome.html in Chrome to see an unfinished, but functioning version of it (it would work in Firefox, too, but only without the frequency analyzer, see page 62).

Of course there are lots of javascripted fallback solutions that could be downloaded from the web, such as VideoSub for WebVTT e.g. taking effect whenever WebVTT is not supported.[21]

Solution #4

Note: In order to follow on, check and evaluate what had to be done and accomplished, please open the files player.pdf, player_js.pdf, player_css.pdf on the DVD.

To see the application in action, please open the file App/index_1.html in Internet Explorer 11 and click on GEBLOCKTE ELEMENTE ZULASSEN if the message "Das Ausführen von Active-X-Steuerelementen wurde für diese Website eingeschränkt" comes up. Otherwise it won't work! Click through the menu items on the left in order to get to all topics. If you have an editor (Microsoft Expression Web, PSPad etc.) available, you can also look at all html, javascript and css documents in the folder "App".

Building the SFSA it was important to keep in mind fulfilling all three main expectations: The video needs to run trouble-free online, switching between audio tracks must be smooth and thirdly there has to be additional information without leaving the page. Since straightforward solutions don't seem to work properly and enjoyably enough, a clever idea to work one's way around is now presented as solution #4:

[21] see 'Stories In Flight | HTML5 Video with SRT Subtitles' <http://www.storiesinflight.com/js_videosub/#example> [accessed 19 February 2014].; only supports one track and ignores further tracks

The video is merged with all audio tracks needed as one file in an mp4 container. Unfortunately there is no support for in-band multitrack webm files for Google Chrome browser yet. Ogg (Firefox) is supported to some extent, but a software to merge multitrack ogg-files could not be found yet.[22]

Each audio track is assigned to a different language. So when the video is opened in a standalone video player, the user would be able to choose the audio track by choosing a language as it is common on many movie DVD's. The standard HTML5 video player also imports the multiple audio tracks as languages and shows an icon where the desired language can be chosen. Switching between tracks happens with no interruption of neither video nor audio. Since the audio is in-band it is always 100% in-sync of course. For the implementation it was now key to make use of these two advantages.

Naturally the user wants to choose between 'microphones' rather than languages. So just as in the prior presented solutions, the video controls are now overwritten by a complete new custom made video controls bar containing all usual controls plus an activation/deactivation button for each track. The javascript for this button then looks like this (simplified demo):

```
function button (){
    //  Get list of audio tracks
    var oAudioTracks =
    document.getElementById("video").audioTracks;
    for (var i = 0; i < oAudioTracks.length; i++) {
    // Step through audio track list
        var audioTrack = oAudioTracks[i];
        if (audioTrack.language =="de"){
            audioTrack.enabled=true;
          } else {audioTrack.enabled=false;}}}
```

First off, the function loads the video with all audio tracks. By using the audioTrackList object (representing all audio tracks available; officially not yet expected to be working

[22] see 'HTML5 Multi-Track Audio or Video | Ginger's Thoughts', sec. Comments <http://gingertech.net/2011/05/01/html5-multi-track-audio-or-video/> [accessed 19 February 2014].

in any browser[23]) audioTracks.length can get the number of all audio tracks - so the for loop now checks through all of them. The if statement within the for loop searches for the language needed by parsing the language codes, enabling the correct language track and deactivating all others.

Each activation/deactivation button is coded separately, so nothing can get mixed up between tracks.

In addition, the SFSA comes with information tracks selectable for each audio track. Therefore the controls bar features an information box. If the checkmark is set, the information will show in the video and also change as audio tracks are getting toggled on and off.

All description tracks are placed as track elements inside the video containing a webvtt file and are given the same source language as the corresponding audio track:

```
<track id="detrack" label="Deutsch" kind="captions"
src="../vtts/blm.vtt" srclang="de">
```

When the info box is checked, it triggers the function handling the text tracks. The function is similar to the function handling audio. The textTracks property returns a textTracksList object. A for loop goes through all tracks and the if statement determines which language is being turned on. There are three states to text tracks: Mode 0 = DISABLED, Mode 1 = HIDDEN, Mode 2 = SHOWING.[24]

The function button() is now extended by the following code ('deutsch' being the name of the function that turns on the German text track):

```
var checkbox = document.getElementById("myCheckbox");
     if(checkbox.checked){deutsch(checkbox);}
```

In a nutshell, what seems to be a player with different microphone signals and description tracks in effect is a player that activates and deactivates language audio tracks as well as language text tracks. Ultimately there are advantages and some disadvantages to be considered:

[23] see 'HTML Audio/Video DOM audioTracks Property' <http://www.w3schools.com/tags/av_prop_audiotracks.asp> [accessed 19 February 2014].; working in IE 11
[24] see 'HTML Audio/Video DOM textTracks Property' <http://www.w3schools.com/tags/av_prop_texttracks.asp> [accessed 19 February 2014].

Pros	Cons
Only one mp4 file	Multitrack mp4 videos are only supported by IE; webM does not support multitrack yet. A software to merge multitrack ogg-files could not be found yet.
Multitrack in-band audio therefore 100% in-sync playback	If one wants to get rid of an audio track in the future, the video must be rendered again
Text tracks give background information while the video is playing	
Once Chrome, Safari and Firefox support multitrack video files the application is going to be working immediately in these browsers too, no extra coding needed	
The mp4 video can be downloaded, stored and reviewed offline if desired	

Even though the cons are valid, the pros of this solution definitely outstand all other solutions mostly because it is running perfectly.
Some things had to be coded very carefully though e.g. the fullscreen button. The API contains a method called requestFullScreen(). By the end of this year this and many other methods, CSS pseudo class selectors etc. are supposed to be working just fine. So far, many APIs (such as the fullscreen specification etc.) are still in an early experimental technology status. The functionality is likely to change as the spec changes under construction. By adding prefixes authors can use these functions, but at the same time have to agree to the not yet stabilized status and potential to fail. Prefixes look as follows:

Chrome and Safari: -webkit
Firefox: -moz
Internet explorer: -ms[25]

[25] see 'Using Fullscreen Mode - Web Developer Guide | MDN' <https://developer.mozilla.org/en-US/docs/Web/Guide/API/DOM/Using_full_screen_mode> [accessed 19 February 2014].

A feature which exceeds the expectations of the player functions is a frequency spectrum analyzer.

<u>Note</u>: See its complete javascript code open the file player_frequency_analyser.pdf. See a demo by opening Analizer/index_chrome.html in your latest Google Chrome browser. For now it only works in Chrome.

The analyzer can help notice differences in the frequency spectrum as the user listens to the different signals. The AudioContext interface represents connections between a source and a designated output as well as multiple AudioNode objects and attributes that can be used to create online analyzers, stereo panners, gain knobs, equalizer, delays, compressors, signal modulation etc. For now audioContext objects are only supported in Chrome, but once all other browsers move up, this tool will be fully working as a neat feature and nice add-on.[26]

In order to create a spectrum analyzer, an audioContext() object and an analyzer node (using the createAnalyser method) had to be created. When the window is loaded, a function loads all audio elements and sets them as sources connecting with the analyzer node. This node is "able to provide a real-time frequency and time-domain analysis information"[27] without changing the signal. Then the node connects the source (input) to an output – in this case simply the default audio hardware.

A `<canvas>` element is created for on the fly visual rendering of the analyzer output as 2D graphics.

A second function frameLooper() makes sure that the analysis is constantly happening. This is done the following way:

```
window.webkitRequestAnimationFrame(frameLooper);
```

Using the requestAnimationFrame() method, the frameLooper() function will now loop at the default frame rate that the browser provides – approximately 50fps - causing it to run through all code within this function for every frame. A new so called Uint8Array() instance is an array that eventually will hold all data of the sound frequency. A for loop then creates a 2D-bar for each frequency that needs to be shown in the canvas element representing the current value of each frequency. Since this process happens

[26] see 'Web Audio API', para. 4.1 <http://www.w3.org/TR/webaudio/#AudioContext-section> [accessed 24 February 2014].

[27] 'AnalyserNode - Web API Interfaces | MDN' <https://developer.mozilla.org/en-US/docs/Web/API/AnalyserNode> [accessed 24 February 2014].

many times per second, it is a very accurate display of the sound frequency spectrum. However, the display is a linear scale which is very unfortunate for the range between 20 – 1000Hz. Finally, this kind of analysis is FFT (Fast Fourier transform) based causing data loss to the displayed result.[28]

Both coding a logarithmic scale for better view of the low frequencies and the approach to a loss-free visualization are very complex and have to be done outside of this thesis.

3. The Application as a whole

The customized video player surely is the core tool of the SFSA. But there are two more main parts utilizing and expanding opportunities, making even more information available for customers.

All topics considering dialogue recording for picture that are discussed and reviewed in this thesis appear as selectable menu items in the menu bar on the left. The code is structured in a way that new topics can be easily added in the future.

By hovering an item with the mouse cursor, all sub menus become visible. For arrangement and clarity purposes, the content of a main topic is displayed in the center area of the website after clicking on the appropriate item. All sub menus are on this same site. Clicking on the sub menu items just leads the user straight to the requested paragraph. The video player is now embedded within text, functioning as interactive demonstration gadget. The column to the right contains further information on the demo videos like a commented picture gallery. This is very valuable for users who want to take a look at how the videos were shot and recorded considering were the microphones were placed.

Also, all Schoeps microphones used are shown again in this section. When clicking on the microphone link, a separate so-called modal window opens up, allowing the user to stay on the site, but receiving special facts and background knowledge about the microphone in an extra window in the foreground. Schoeps had emphasized that they didn't fancy pop-up windows at all. The modal window solution complies with this requirement, but still uses the benefits of a pop-up window.

Users who are interested in more specifics about the microphones themselves reach the correspondent page in a new tab by clicking on the icon of each microphone.

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[28] see Boris Smus, *Web Audio API* (Beijing: O'Reilly Media, 2013), chap. 5 <http://search.ebscohost.com/login.aspx?direct=true&scope=site&db=nlebk&db=nlabk&AN=548863> [accessed 24 February 2014].

This way the SFSA interleaves its content with content on the Schoeps website allowing the user to discover Schoeps microphones in a whole new way.

|| Conclusion

Recording professional dialogue for film can be summed up in one goal with three key points: Capture clean, consistent and intelligible audio.

This thesis makes a valuable introduction into the character of the human voice as the number one sound source considering production sound. This acts well as vital fundament for all following content in order to refer back to and explain how microphone selection and placement make best sense.

The thesis also presents a widespread but at the same time dead on target information about technical specifications of microphones used for location dialogue recording and connects it to a variety of tips of how to utilize the behavior of each microphone to the full.

In order to fulfill the main goal mentioned above, the presented amount of numerous different microphone positioning techniques and ideas interlocked with the right microphones completes a good guide for location dialogue recording. However, up to this point this thesis imparts only a few new aspects and advice next to well-known and over the years successfully applied knowledge of experienced production sound mixers and would therefore be just another online information service, or piece of literature in a vast pool of teaching.

For the Schoeps Film Sound Application something new and innovative had to be created to present the existing knowledge in a whole new way on a shared platform where thoughts, views and suggestions could also be contributed by various authors.

Unlike many resources the Schoeps Film Sound Application is to provide technical and creative information and tips rather than to define right or wrong. Although the thesis states many do's and don'ts in film sound, it is not the intention to judge microphone selection or placements because so often a compromise has to be made for the sake of picture. Production sound requires a good set of principles that can be customized and applied to multiple situations. Sound mixer Thom Shafer puts it this way:

> *"Being a great mixer is knowing what sounds good and what doesn't as it applies to the situation you find yourself in..."*[1]

[1] 'Thom Shafer Location Sound Mixer Film and High Definition Video' <http://www.televisionsound.com/> [accessed 15 March 2014].

Even though the application is going to become part of the official Schoeps website, it is not to be considered an advertising rostrum for Schoeps, but a presentation tool discovering the fascinating world of film sound.

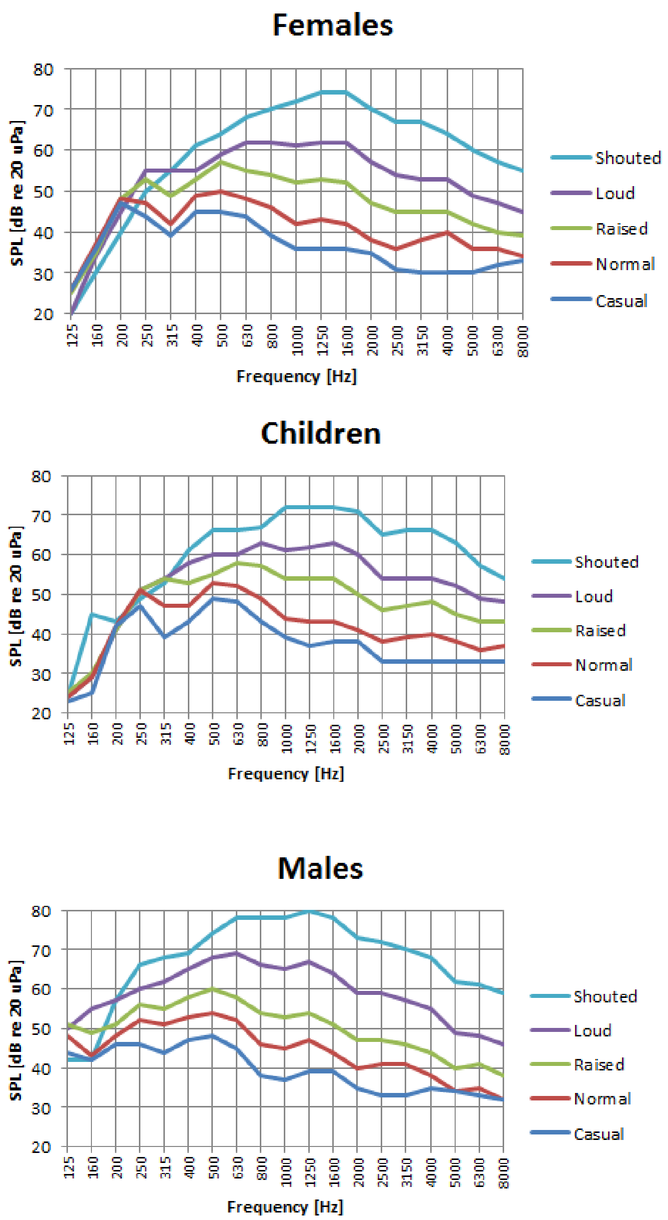
The thesis helps start this growing project especially with a capable and never before seen video player implemented in HTML5 allowing not only the content of the website to grow, but also its technological opportunities. It is to be expected that upcoming HTML versions in the future are always going to be built upon HTML5 – a promising start for an ongoing online tool. The short films show in a phenomenal way how all explained topics were put into practice and give every user a realistic audio and video experience. The Schoeps Film Sound Application provides and shares information in an appealing way for both interested newcomers and troupers who really care about sound in order to make their work easier and even better. This way the Schoeps Film Sound Application connects people with a passion for the great art form of film sound. Because in the field, theory is one thing - ultimately it's the person behind that makes the difference. Technology can only do so much for clean, consistent and intelligible audio and sometimes even limits the really passionate mixer.

> *"Production sound can never be allowed to be a track on some machine that's just gonna get it…so…you'll have to have people that care about sound. I find that the best sound doesn't come from the best technicians. It comes from the best technicians that really care about sound."*[2]
> *- Nicholas Allen*

[2] 'SoundWorks Collection - Production Sound Mixer Nicholas Allen', chap. 06:23min.

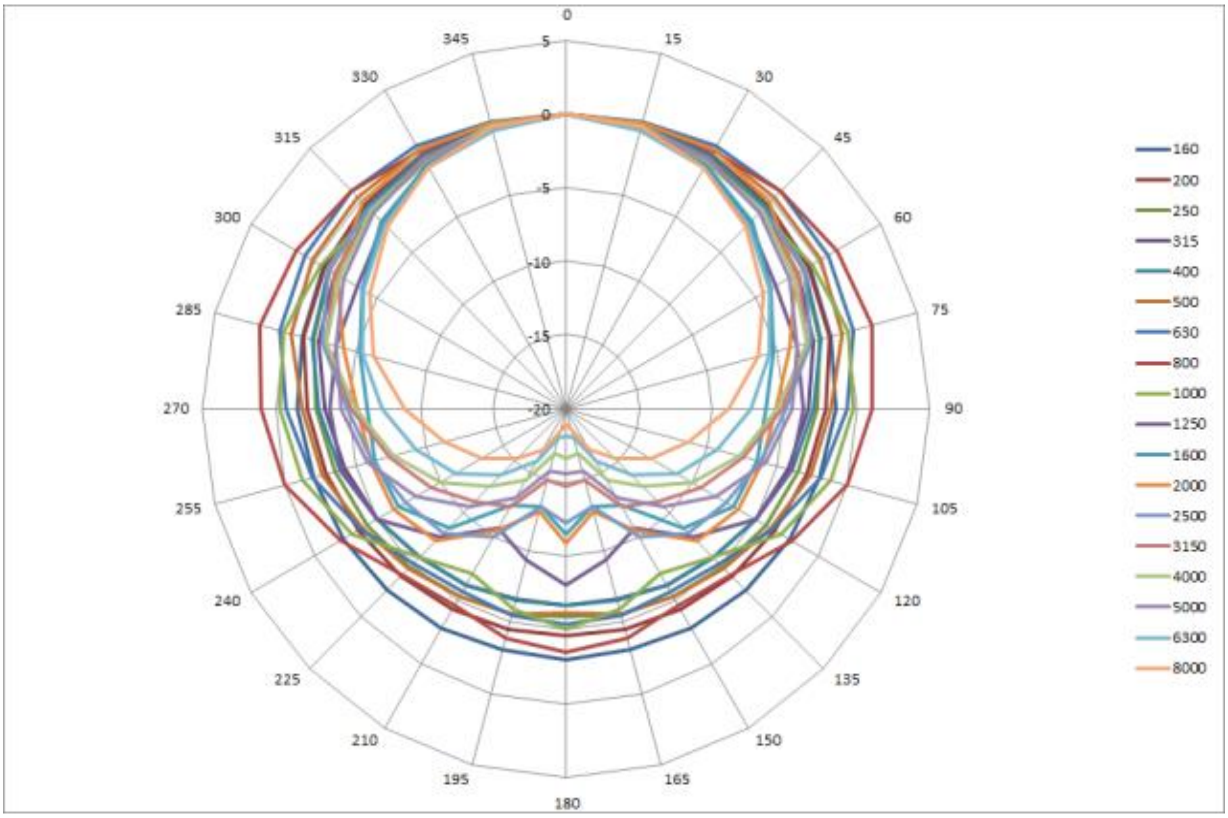
|| Appendix

Appendix Figure #1: Voice spectra (1/3 octave) depending on efforts[1]

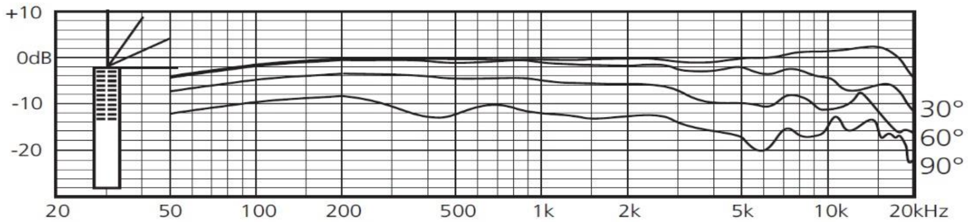


[1] 'DPA Microphones :: Facts about Speech Intelligibility', para. 1.

Appendix Figure #2: Frequency dependent polar plots from 160Hz – 8 kHz[2]



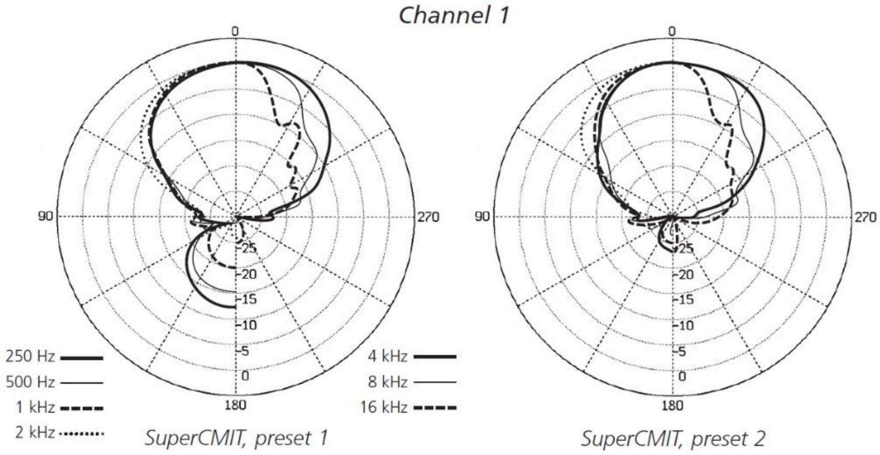
Appendix figure #3: Frequency response of the Schoeps CMIT from different angles[3]



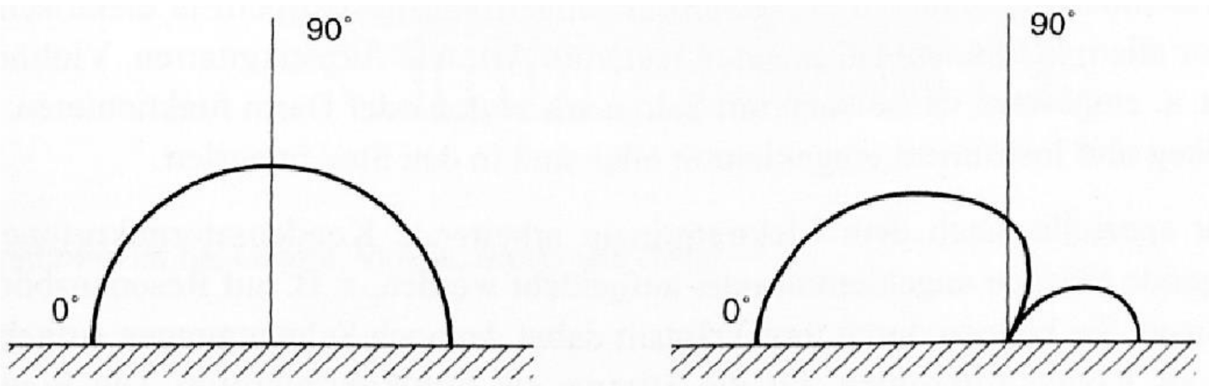
Frequency response curve without filters at 0°, 30°, 60°, 90° angle of incidence

[2] 'DPA Microphones :: Facts about Speech Intelligibility', para. 3.
[3] 'Schoeps CMIT Manual.pdf', p. 25.

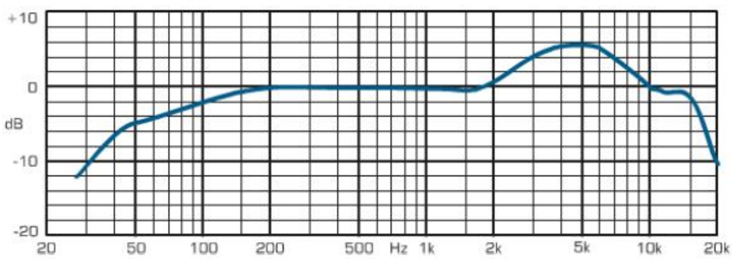
Appendix figure #4: Polar patterns of SuperCMIT Preset 1 and 2[4]



Appendix figure #5: Polar patterns of omnidirectional and cardioid boundary layer microphone[5]



Appendix figure #6: Frequency response curve of the DPA 4071[6]



[4] 'Schoeps SuperCMIT Manual.pdf', p. 6.
[5] Dickreiter, p. 115.
[6] 'Microphone Data -' <http://www.microphone-data.com/microphones/4071/> [accessed 14 March 2014].

Appendix Figure #7: Table of audio codec support in HTML5[7]

Browser	WAV	Ogg Vorbis	MP3
Firefox	✔	✔	--
Safari	✔	--	✔
Opera	✔	✔	--
Google Chrome	✔	✔	✔
IE	--	--	✔

Appendix Figure #8: Attributes & methods of the video element[8]

HTML Attributes

- `autoplay` = "autoplay" or "" (empty string) or empty
  Instructs the UA to automatically begin playback of the video as soon as it can do so without stopping.
- `preload` = "none" or "metadata" or "auto" or "" (empty string) or empty
  Represents a hint to the UA about whether optimistic downloading of the video itself or its metadata is considered worthwhile.
  - "none": Hints to the UA that the user is not expected to need the video, or that minimizing unnecessary traffic is desirable.
  - "metadata": Hints to the UA that the user is not expected to need the video, but that fetching its metadata (dimensions, first frame, track list, duration, and so on) is desirable.
  - "auto": Hints to the UA that optimistically downloading the entire video is considered desirable.
    Specifying the empty string is equivalent to specifying the value "auto".
- `controls` = "controls" or "" (empty string) or empty
  Instructs the UA to expose a user interface for controlling playback of the video.
- `loop` = "loop" or "" (empty string) or empty
  Instructs the UA to seek back to the start of the video upon reaching the end.
- `poster` = URL potentially surrounded by spaces
  The address of an image file for the UA to show while no video data is available.
- `height` = non-negative integer
  The height of the video, in CSS pixels.
- `width` = non-negative integer
  The width of the video, in CSS pixels.
- `muted` = "muted" or "" (empty string) or empty
  Represents the default state of the audio channel of the video, potentially overriding user preferences.
- `mediagroup` = string
  Instructs the UA to link multiple videos and/or audio streams together.
- `src` = URL potentially surrounded by spaces
  The URL for the video.

[7] Pfeiffer, p. 20.
[8] 'HTML/Elements/video - W3C Wiki'.

Methods Event Name

loadstart	waiting
progress	playing
suspend	canplay
abort	canplaythrough
error	seeking
emptied	seeked
stalled	timeupdate
play	ended
pause	ratechange
loadedmetadata	durationchange
loadeddata	volumechange

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