http://www.josephson.com/c700a.html

B-Format Natif

Micro Ambisonique Natif

la solution ?

JOSEPHSON

C700S

Bernard Lagnel

Février 2019

https://www.lesonbinaural.fr

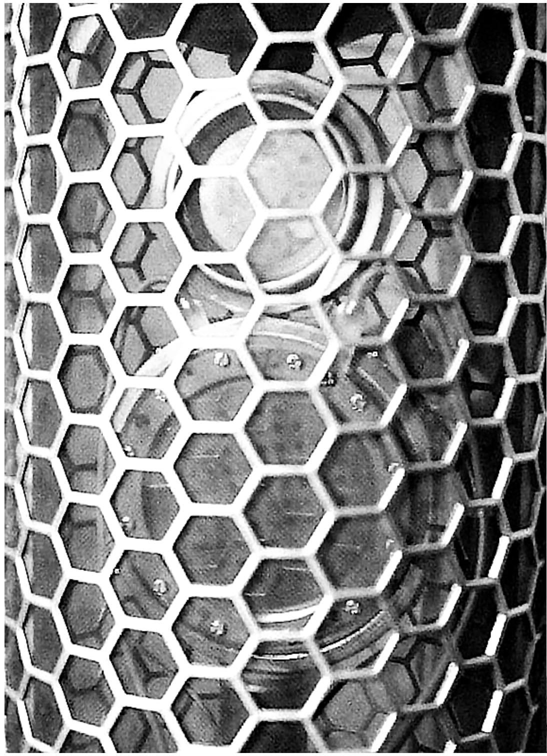
C700S

Importateur :

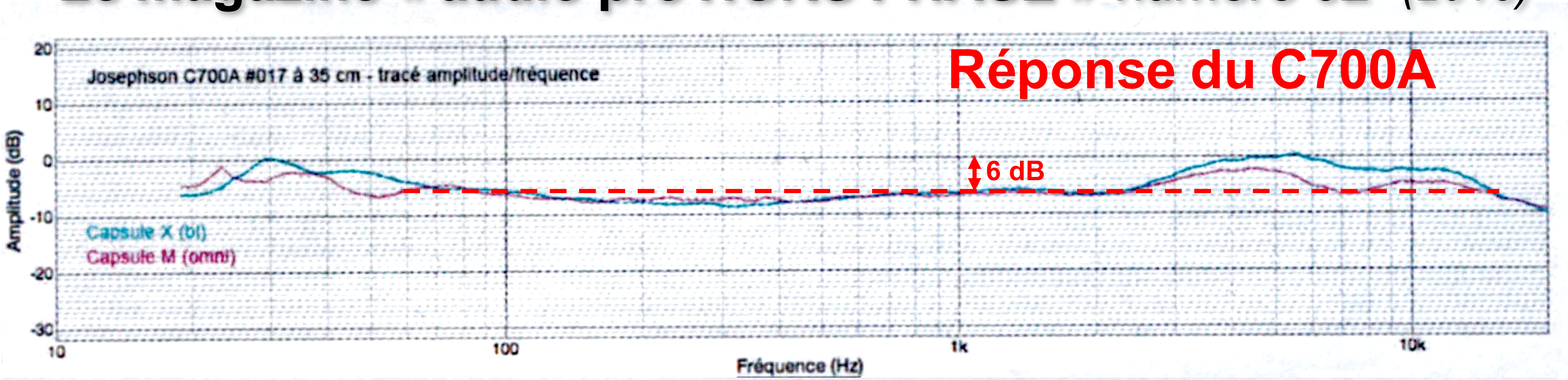
http://www.joystick.be/contact.html

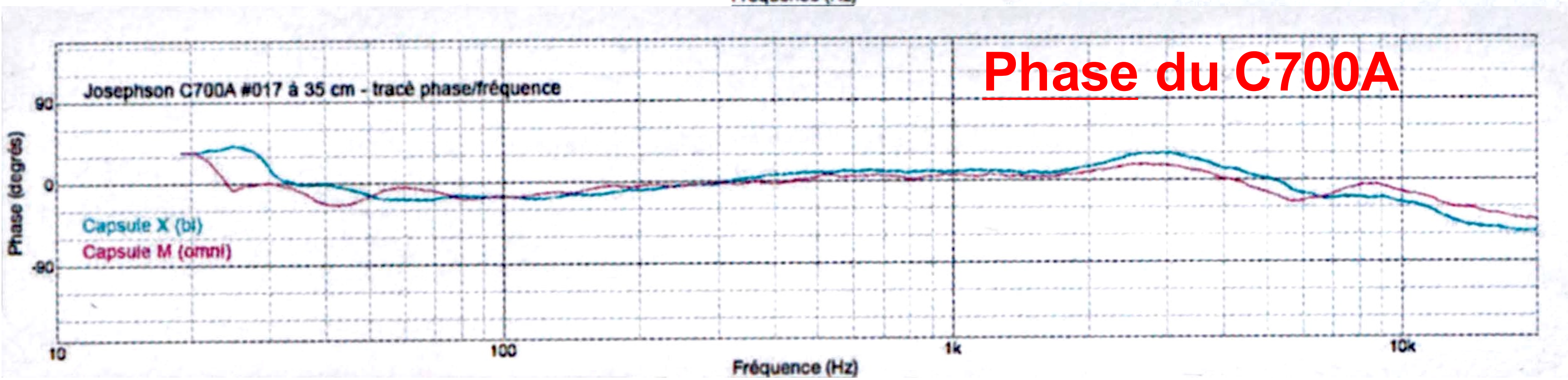
PROFESSIONAL CONDENSER MICROPHONES
SINCE 1988 • MADE IN SANTA CRUZ, CALIFORNIA

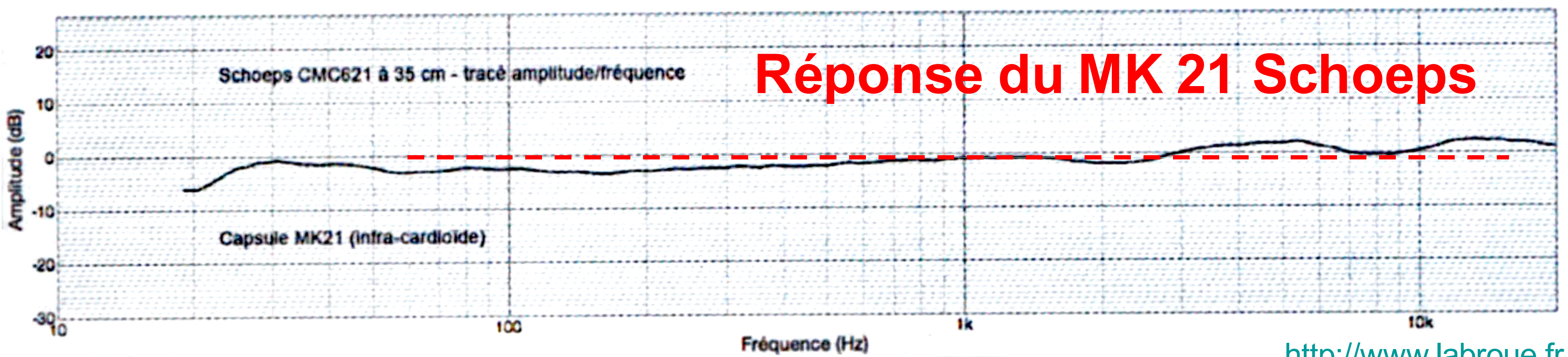
JOSEPHSON



Le magazine « audio pro HORS PHASE » numéro 02 *(2013)*





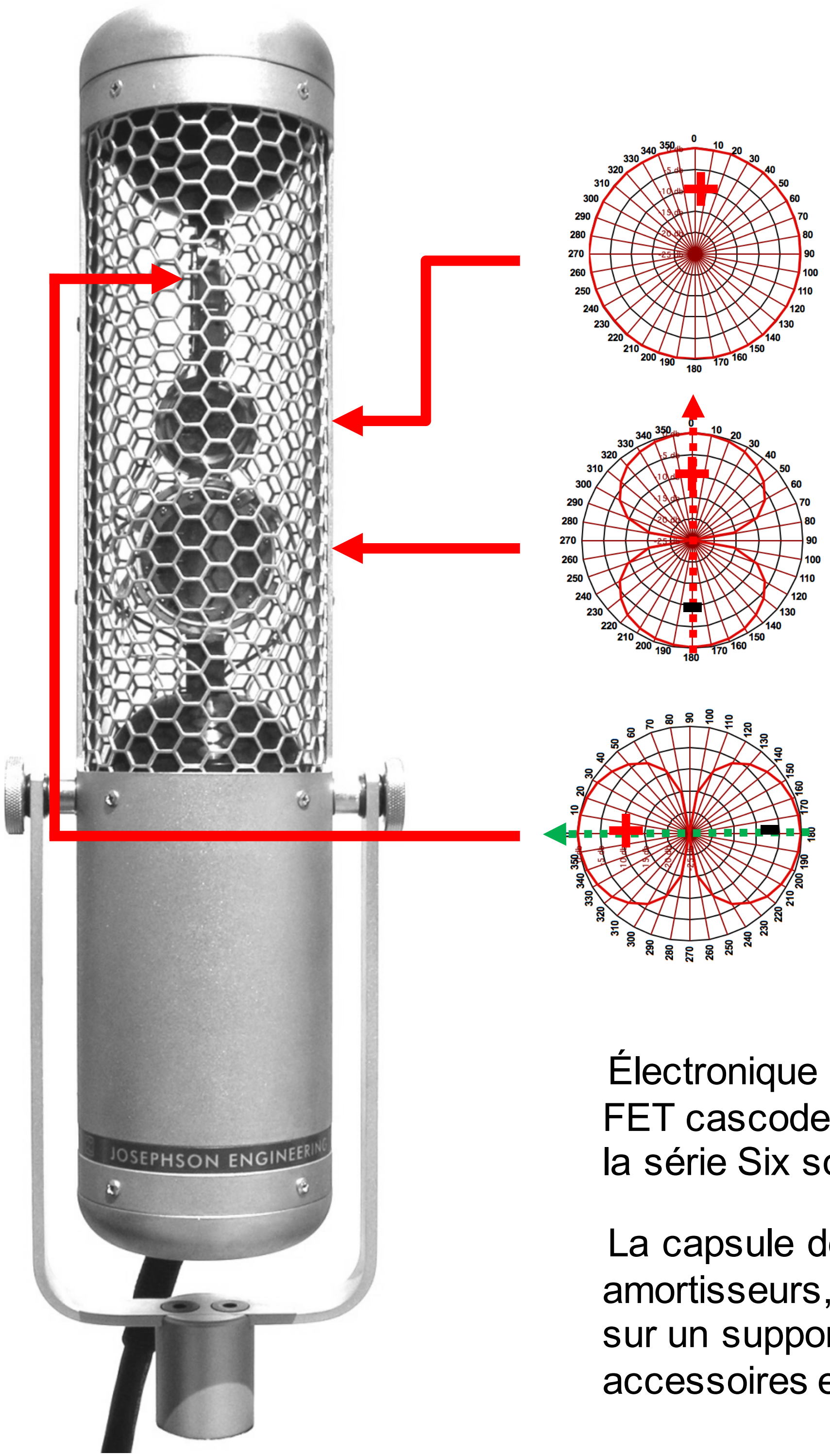


http://www.labroue.fr


XLR 7 Broches


AMBISONIQUE NATIF
FORMAT B (W X Y)
X
W
XLR 7 Broches

C700S



OMNI ⌀ 16 mm W

(semblable à la capsule KA11)

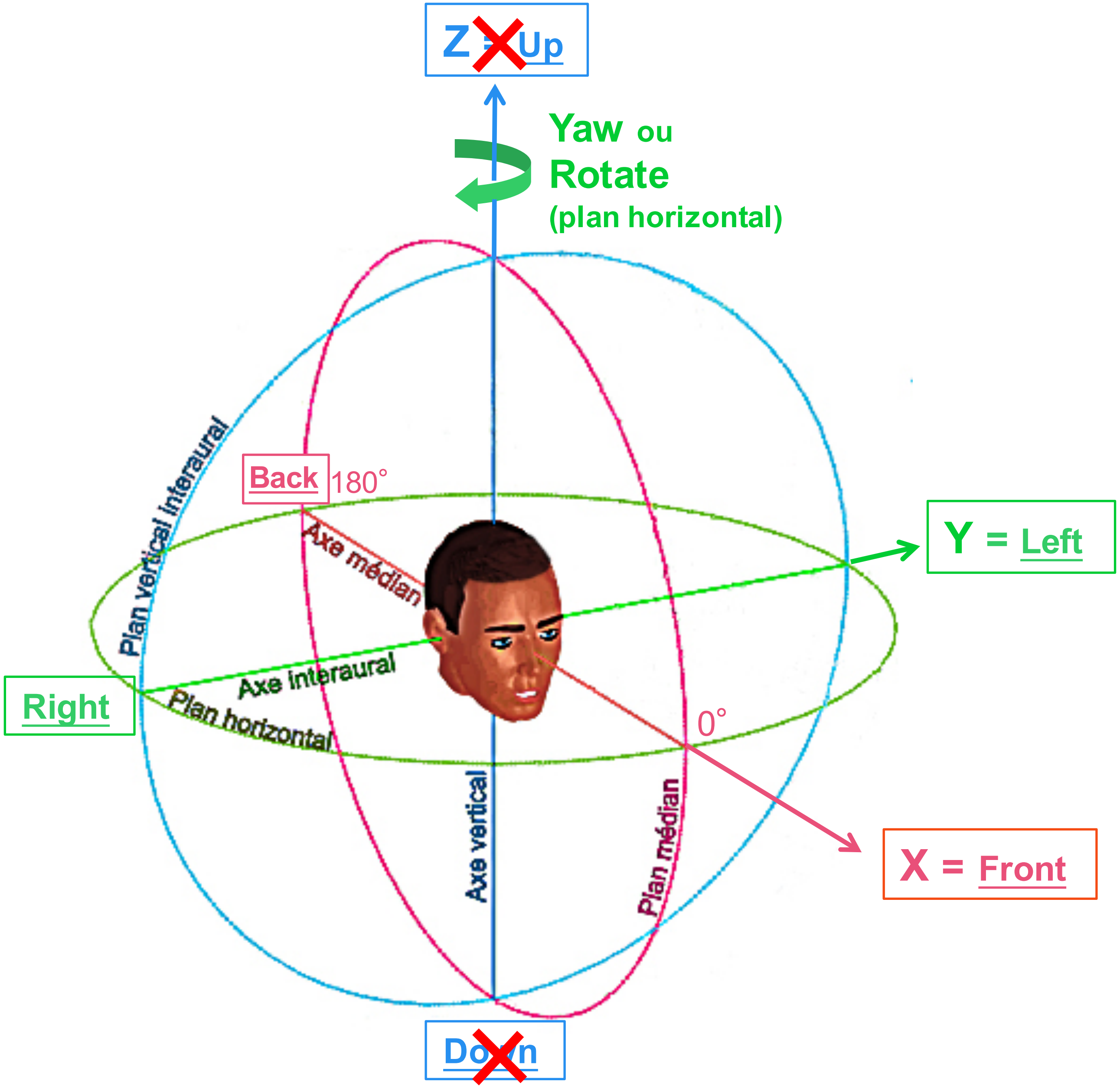
Bi ⌀ 26 mm X

Bi ⌀ 26 mm Y

Électronique complètement neutre, utilisant la même entrée FET cascode, classe A active circuit de sortie symétrique de la série Six sont fournis pour chaque signal.

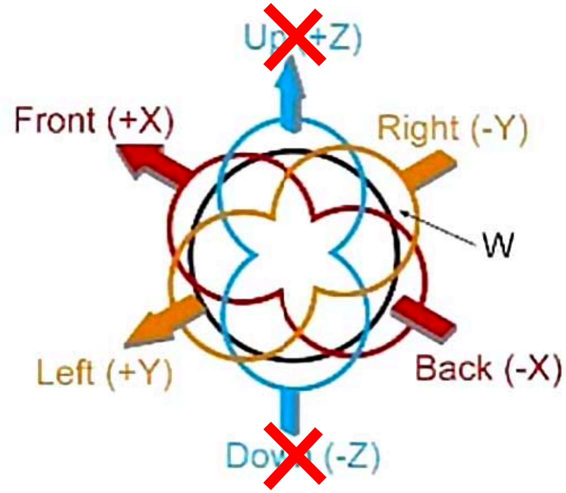
La capsule de la série Seven est montée à l'intérieur des amortisseurs, de sorte que le micro peut être fixé directement sur un support par le biais de sa monture sans aucun accessoires externes.

Rotation Ambisonic 2D



Les 3 Plans :

1. ~~Plan médian : Pitch ou Tumble~~
2. Plan horizontal ou azimutal : Yaw ou Rotate
3. ~~Plan vertical ou interaural : Roll ou Tilt~~

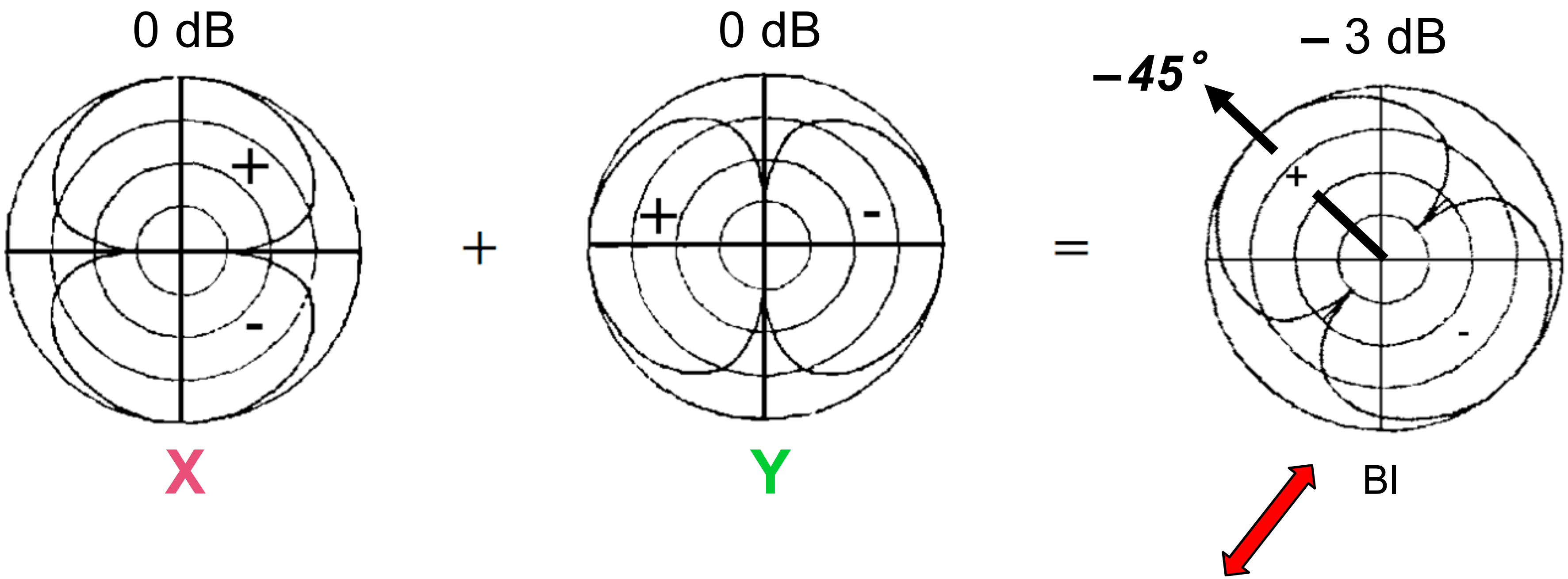


2.0 issu du B-Format <u>*Création d'un XY*</u>

- **– 45°**
- **+45°**

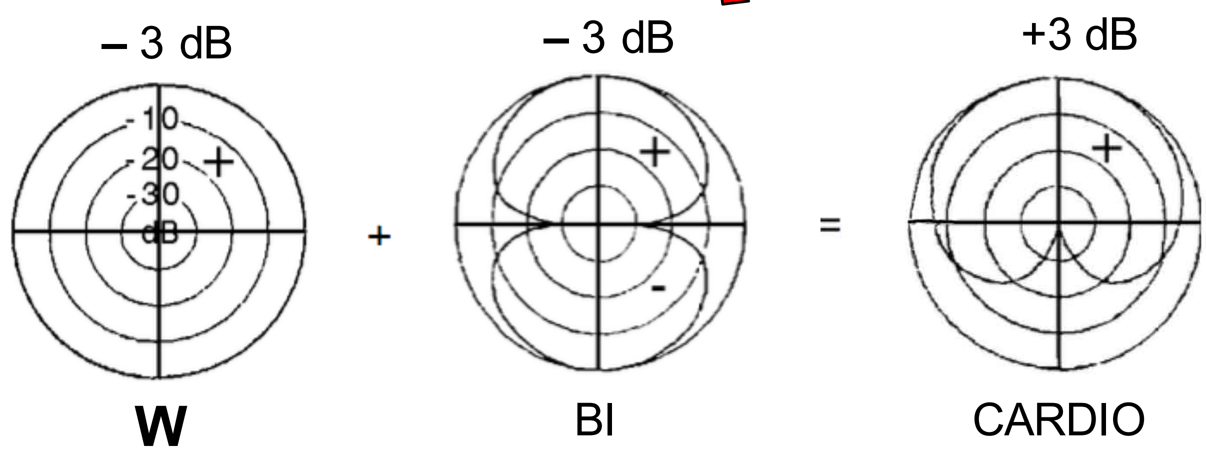
- **L = (– 3 dB) W + (0 dB) X + (0 dB) Y**
- **R = (– 3 dB) W + (0 dB) X – (0dB) Y**

- **Somme = + 3,0 dB**
- **Somme = + 3,0 dB**



Antilog	**Niveau**
1	**0,0 dB**
1/√2	**– 3,0 dB**

20 **x** log(1/√2) = – 3,0 dB



Création d’un XY à partir des axes X et Y :

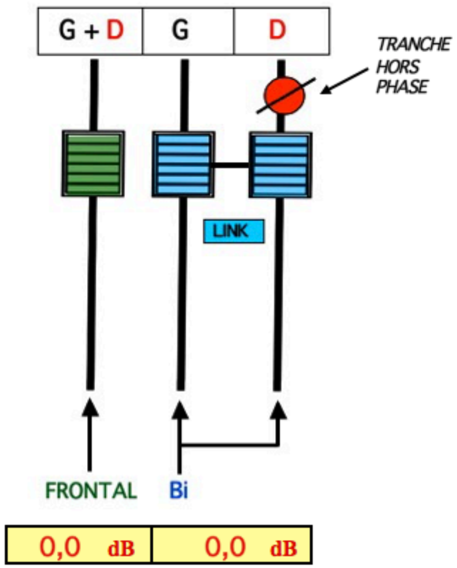
DONNÉES
* Caractéristique du micro FRONTAL
0,000
Différence de sensibilité entre le micro FRONTAL et le micro Bi
0,0 dB

RÉSULTATS à L' AVANT	
1/2 Angle de prise de son utile	1/2 Angle mécanique fictif du couple
32 °	45 °
1/2 Angle maximum de prise de son en PHASE	Perte de gain du à la sommation
60 °	-3,0 dB
Atténuation à l'avant 0° du couple	Atténuation à l'arrière 180° du couple
-3,0 dB	-3 dB

* NOTE .
Caractéristiques des micros :

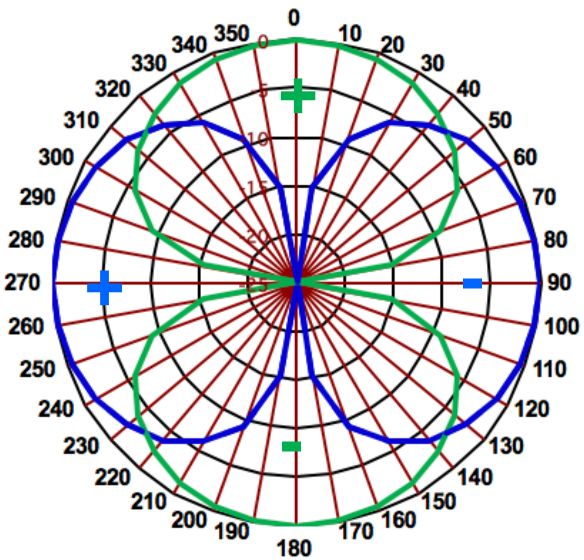
Micro OMNI = 1
Micro INFRA ≈ 0,660 (-10 dB arrière)
Micro CARDIO = 0,5
Micro SUPER ≈ 0,375 (-12 dB arrière)
Micro BI = 0

MATRICAGE

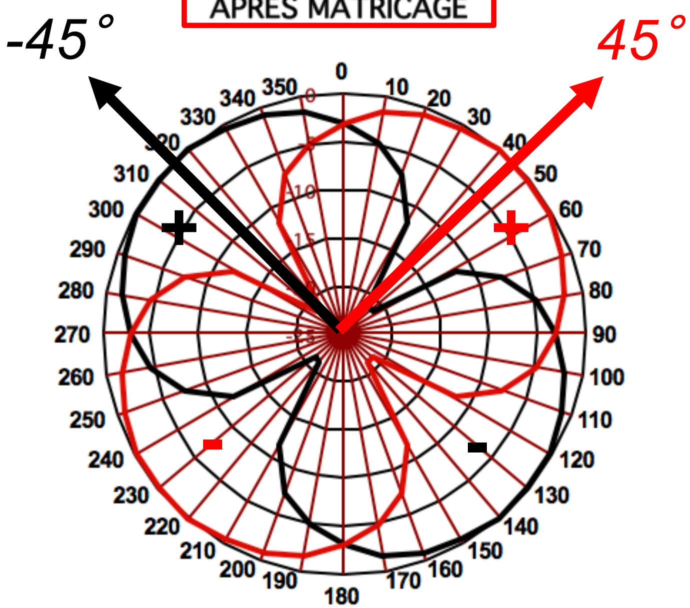


VALEURS AUX POTS

AVANT MATRICAGE



APRÈS MATRICAGE



https://www.lesonbinaural.fr/EDIT/EXCEL/double_ms_simple.xls

Création d'un XY à partir de l'axes W :

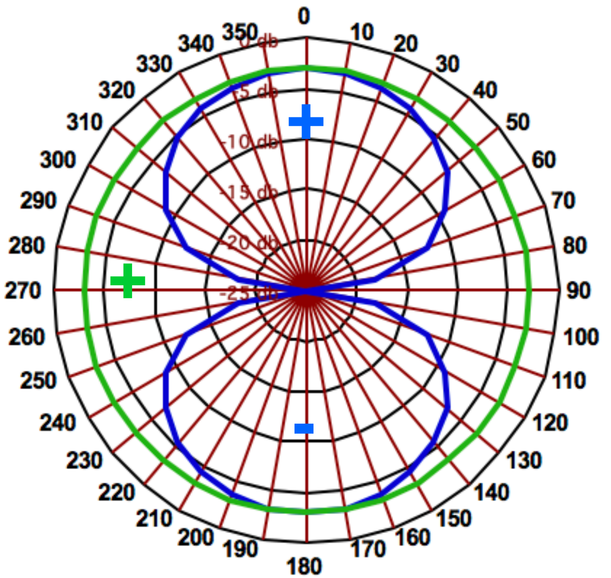
* Caractéristique du micro FRONTAL 1
1,000
*** Caractéristique du micro FRONTAL 2**
0,000

RÉSULTATS DE LA SOMMATION DES 2 MICROS VISANT LA MÊME DIRECTION	
* Caractéristique de directivité du micro après sommation	Niveau maximum du micro après sommation
0,500	3,0 dB

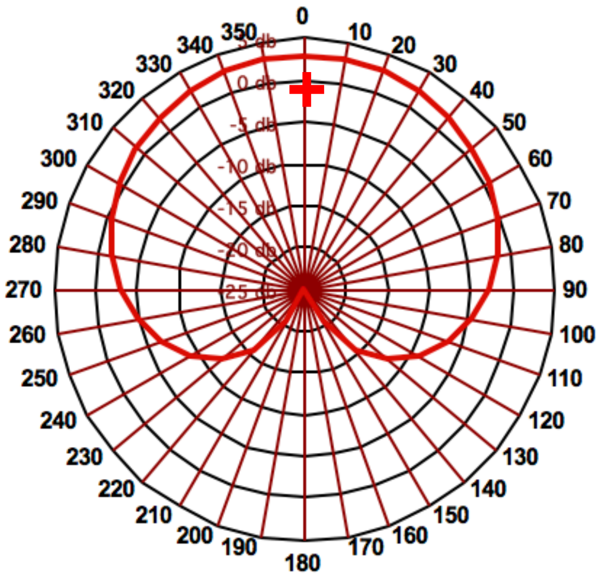
Différence de sensibilité entre le micro FRONTAL 1 et le micro FRONTAL 2
3,0 dB

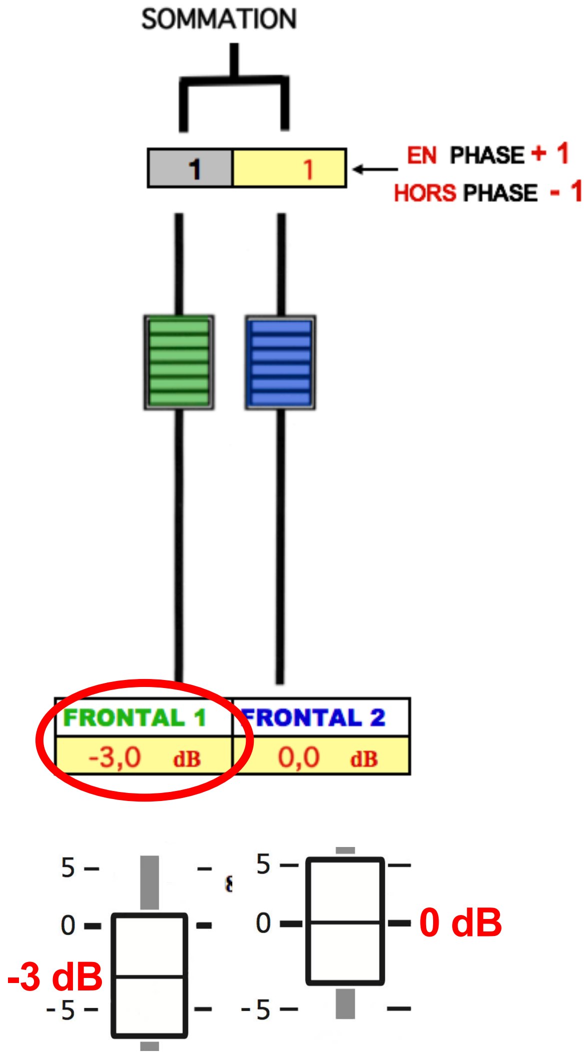
* NOTE :
Micro OMNI = 1
Micro INFRA ≈ 0,660 (-10 dB arrière)
Micro CARDIO = 0,5
Micro SUPER ≈ 0,375 (-12 dB arrière)
Micro BI = 0

AVANT SOMMATION



APRÈS SOMMATION





https://www.lesonbinaural.fr/EDIT/EXCEL/somme_2micros.xls

Caractéristiques du couple stéréophonique :

* Directivité des micros L et R	Angle entre les micros L et R	Distance entre les micros L et R
0,500	90 °	0 cm

* Directivité après la SOMMATION de L et R (signaux en phases)
0,586

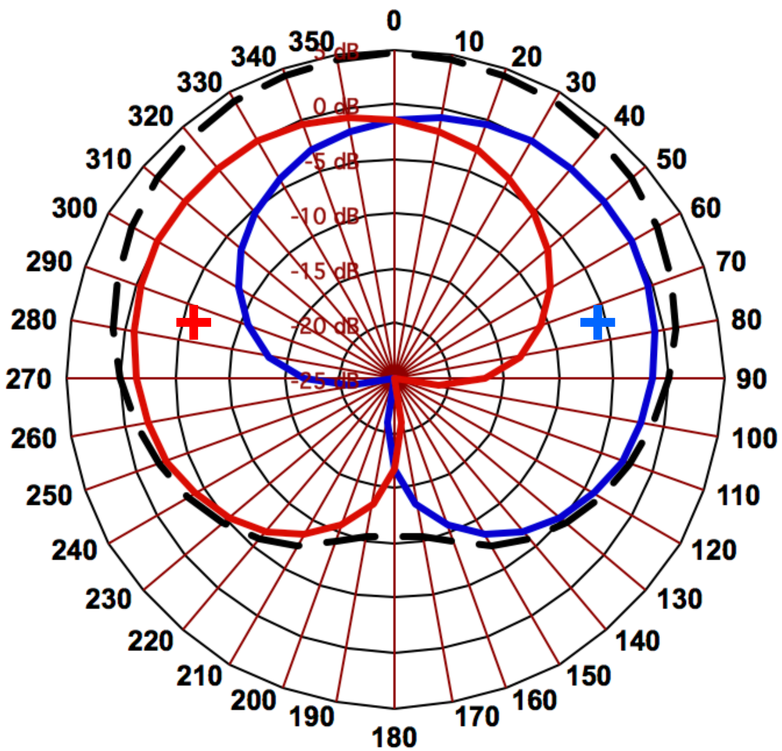
Distance de la source sonore
1,0 m

Angle total de prise de son utile du couple
150 °

Pourcentage en niveau ΔL et en temps ΔT (entre les micros L et R)

ΔL dB	ΔT ms
100 %	0 %
Affaiblissements à l' avant 0° du couple	Affaiblissements à l' arrière 180° du couple
-1,4 dB	-16,7 dB

Après SOMMATION : coefficient de directivité du couple Q (réf du Cardio : Q = 3)
2,5
Rapport de capture ou Facteur de Distance = √Q
1,6



* NOTE :

Micro OMNI = 1

Micro INFRA ≈ 0,66 (-10 dB arrière)

Micro CARDIO = 0,5

Micro SUPER ≈ 0,375 (-12 dB arrière)

Micro BI = 0

LES LIENS :

SYSTÈME MMAD (WILLIAMS -LE DÛ)

hauptmikrofon Image Assistant 2.0

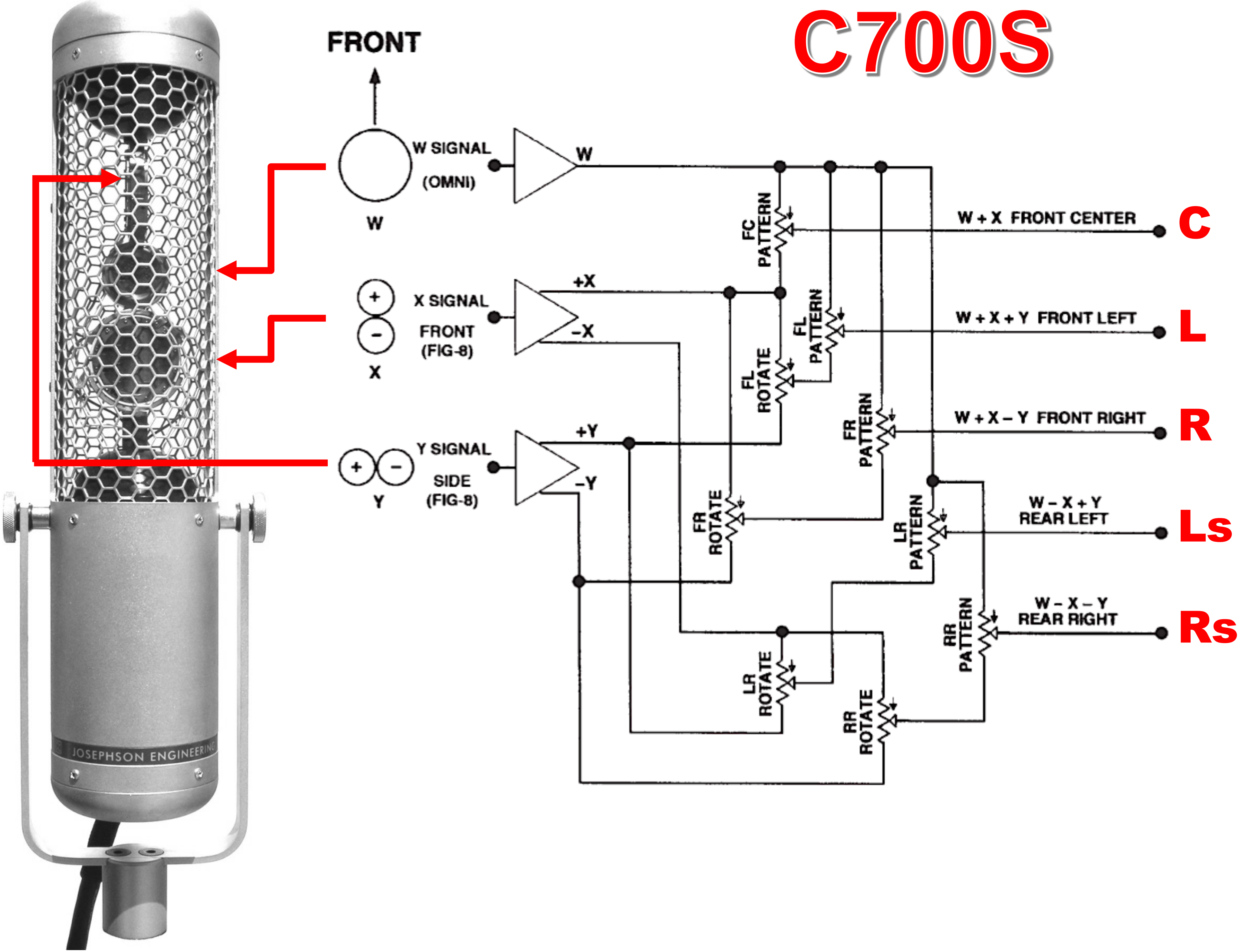
Sensibilité des microphones (re. 1 V / Pa = 0 dB)

mV / Pa	dB	Cliquez sur les micros = Voir l'aperçu.		
100,0	-20,0	Sphère SCHOEPS KFM 6		
89,1	-21,0			
79,4	-22,0			
70,8	-23,0	DPA 4041 SP		
63,1	-24,0			
56,2	-25,0			
50,1	-26,0	SENNHEISER ME 66		
44,7	-27,0	M149 Cardio		
39,8	-28,0	Primo EM172-Z1	DPA 4006A	
35,5	-29,0			
31,6	-30,0	SP-TFB-2		
28,2	-31,0	U87 cardio	Audix SCX25A	
25,1	-32,0	Luhd PM-01Binaural	MS-TFB-2	MKH 40
22,4	-33,0	Gefell M 930	AKG 414	
20,0	-34,0	DPA 4060	KU 100	M147
17,8	-35,0	Panasonic 61A	SCHOEPS CMIT 5 U	
15,8	-36,0	SCHOEPS MK2	KM 140	KM184
14,1	-37,0	SCHOEPS MK41	KU 80 i	
12,6	-38,0	SCHOEPS MK4 et MK4v	KM 130	C700 S en XY
11,2	-39,0			
10,0	-40,0	SCHOEPS MK8	Roland CS-10EM	DPA 4006
8,9	-41,0	DPA 4007A	KU 81 i	C700 S
7,9	-42,0	TLM 170		
7,1	-43,0	AKG C535 EB		
6,3	-44,0	DPA 4061		
5,6	-45,0	Shure SM 81		
5,0	-46,0			

+3 dB

Gamme de fréquences 20-20 000 Hz ± 2 dB depuis la courbe de référence
Sensibilité –41 dB réf 1 V / Pa (9 mV / Pa) pour chaque signal
Surpression sonore 135 dB SPL à 1 kΩ pour <1% THD
Niveau de bruit équivalent 15 dB SPL, pondéré A
Alimentation fantôme P48, 4 mA par sortie
Diamètre 63 mm (largeur 100 mm à la culasse), longueur 328 mm (C700A), 365 mm (C700S)
Connecteur de sortie type XLR 5 broches (C700A), type XLR 7 broches (C700S)

C700S



FRONT
W SIGNAL
(OMNI)
W
W
X SIGNAL
FRONT
(FIG-8)
X
+X
–X
Y SIGNAL
SIDE
(FIG-8)
Y
+Y
–Y
FC PATTERN
FL PATTERN
FL ROTATE
FR PATTERN
FR ROTATE
LR PATTERN
LR ROTATE
RR PATTERN
RR ROTATE
W + X FRONT CENTER
W + X + Y FRONT LEFT
W + X – Y FRONT RIGHT
W – X + Y REAR LEFT
W – X – Y REAR RIGHT
C
L
R
Ls
Rs

https://fr.rode.com/soundfieldplugin#footer_download

5.1 issu du B-Format
(Lobes Cardio)

• 0°	• C = (– 3 dB) *W* + (0 dB) *X*	• Somme = + 4,6 dB
• –35°	• L = (– 3 dB) *W* + (0 dB) *X* + (– 3 dB) *Y*	• Somme = + 3,1 dB
• +35°	• R = (– 3 dB) *W* + (0 dB) *X* – (– 3 dB) *Y*	• Somme = + 3,1 dB
• –110°	• Ls = (– 3 dB) *W* – (– 9 dB) *X* + (0 dB) *Y*	• Somme = + 3,5 dB
• +110°	• Rs = (– 3 dB) *W* – (– 9 dB) *X* – (0 dB) *Y*	• Somme = + 3,5 dB
	• LFE = Pass-Bas (– 3 dB) *W*	• Somme = – 3 dB

Antilog	Niveau
1	0,0 dB
$1/\sqrt{2}$	–3,0 dB
1/2	–6,0 dB
$1/2\sqrt{2}$	–9,0 dB

$$20 \times \log(1/\sqrt{2}) = -3{,}0 \text{ dB}$$

X et Y donnent la direction (pour 35°) :

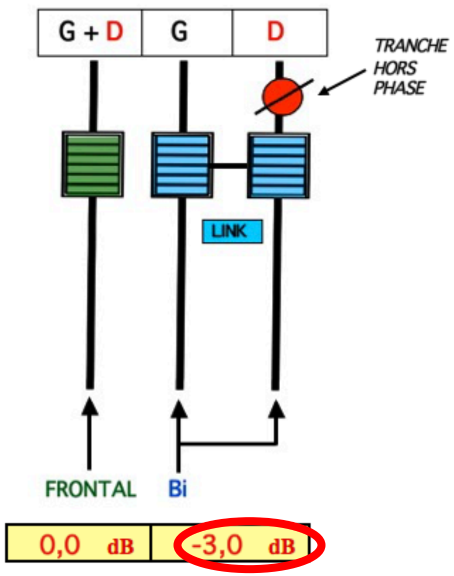
DONNÉES
* Caractéristique du micro FRONTAL
0,000
Différence de sensibilité entre le micro FRONTAL et le micro Bi
0,0 dB

RÉSULTATS à L' AVANT	
1/2 Angle de prise de son utile	1/2 Angle mécanique fictif du couple
41 °	35 °
1/2 Angle maximum de prise de son en PHASE	Perte de gain du à la sommation
70 °	-2,9 dB
Atténuation à l'avant 0° du couple	Atténuation à l'arrière 180° du couple
-1,8 dB	-2 dB

* NOTE .
Caractéristiques des micros :

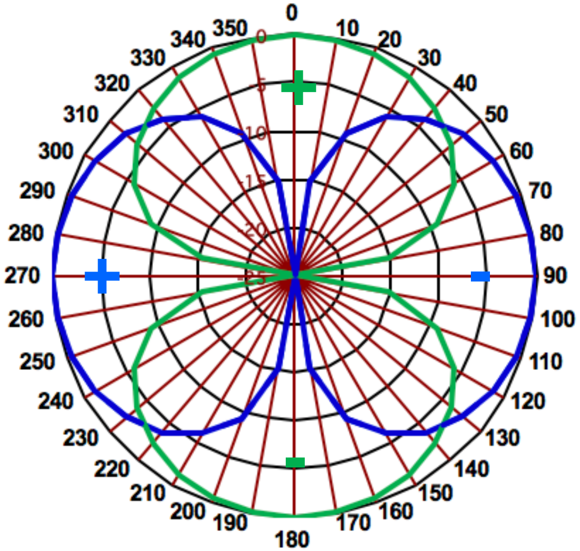
Micro OMNI = 1
Micro INFRA ≈ 0,660 (-10 dB arrière)
Micro CARDIO = 0,5
Micro SUPER ≈ 0,375 (-12 dB arrière)
Micro BI = 0

MATRICAGE

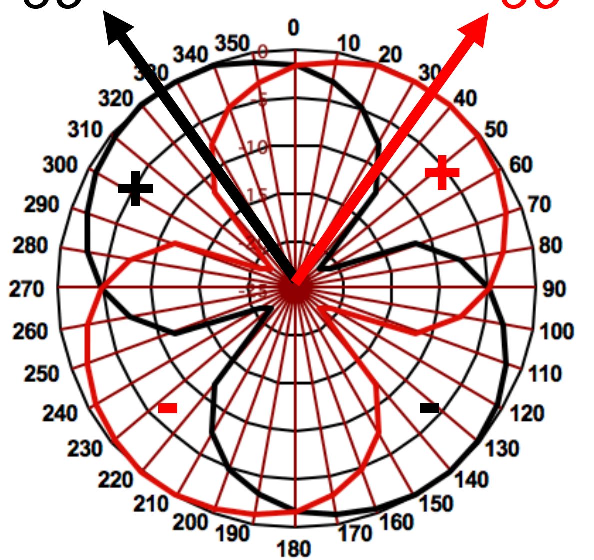


VALEURS AUX POTS

AVANT MATRICAGE



APRÈS MATRICAGE



https://www.lesonbinaural.fr/EDIT/EXCEL/double_ms_simple.xls

W donne la directivité (pour 35°) :

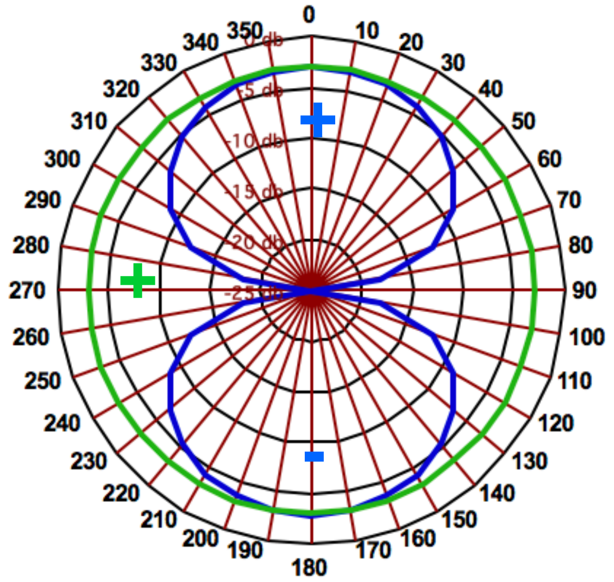
* Caractéristique du micro FRONTAL 1
1,000
* Caractéristique du micro FRONTAL 2
0,000

RÉSULTATS DE LA SOMMATION DES 2 MICROS VISANT LA MÊME DIRECTION	
* Caractéristique de directivité du micro après sommation	Niveau maximum du micro après sommation
0,497	3,1 dB

Différence de sensibilité entre le micro FRONTAL 1 et le micro FRONTAL 2
2,9 dB

* NOTE :
Micro OMNI = 1
Micro INFRA ≈ 0,660 (-10 dB arrière)
Micro CARDIO = 0,5
Micro SUPER ≈ 0,375 (-12 dB arrière)
Micro BI = 0

AVANT SOMMATION



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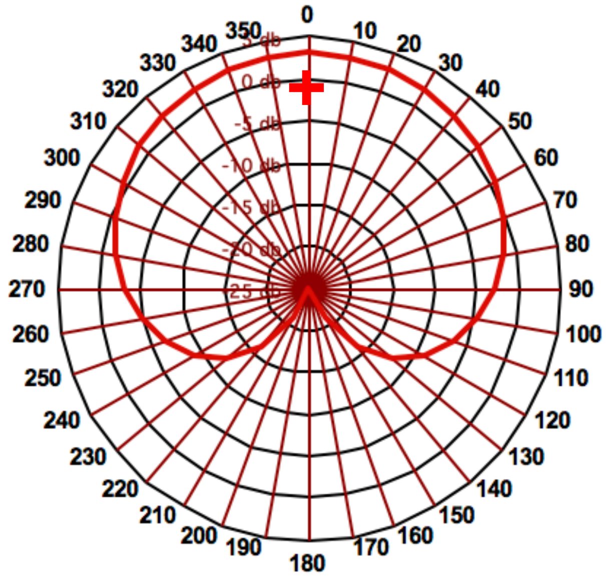
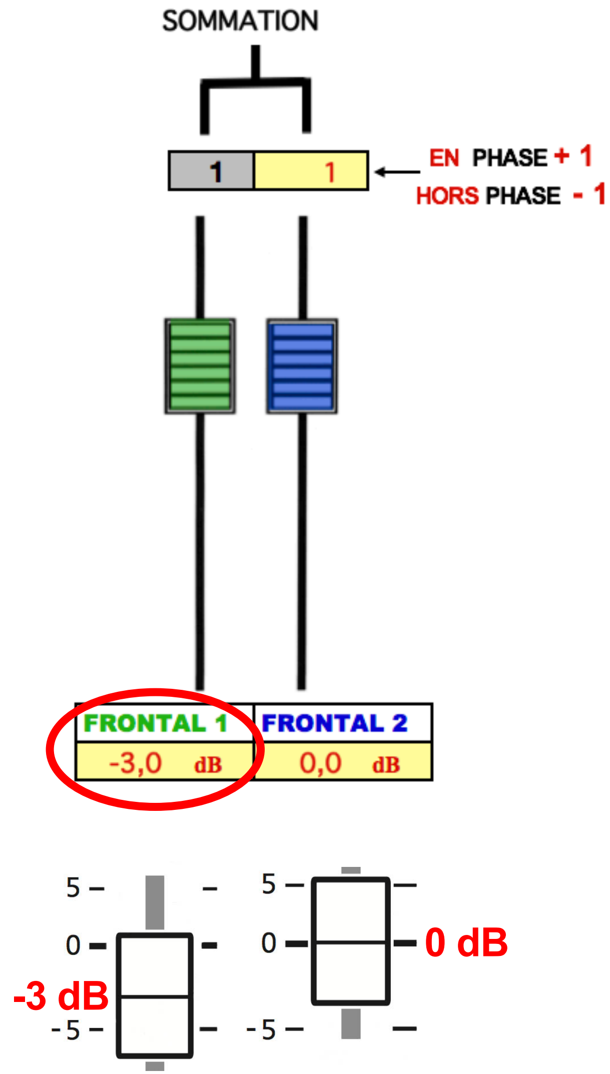




https://www.lesonbinaural.fr/EDIT/EXCEL/somme_2micros.xls

X et Y donnent la direction (pour 110°) :

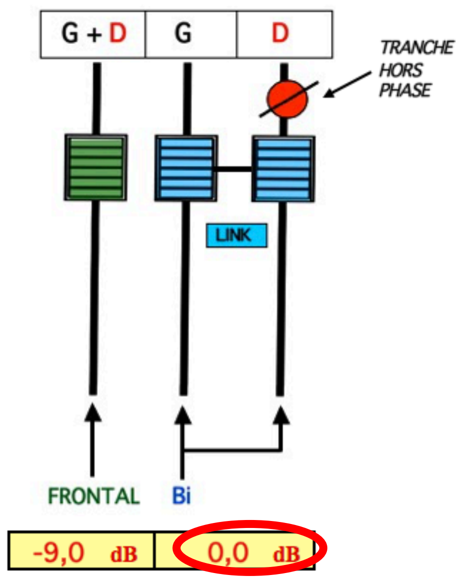
DONNÉES
* Caractéristique du micro FRONTAL
0,000
Différence de sensibilité entre le micro FRONTAL et le micro Bi
0,0 dB

RÉSULTATS à L' ARRIÈRE	
1/2 Angle de prise de son utile	1/2 Angle mécanique fictif du couple
90 °	70 °
1/2 Angle maximum de prise de son en PHASE	Perte de gain du à la sommation
30 °	-2,1 dB
Atténuation à l'avant 0° du couple	Atténuation à l'arrière 180° du couple
-9,5 dB	-10 dB

* NOTE .
Caractéristiques des micros :

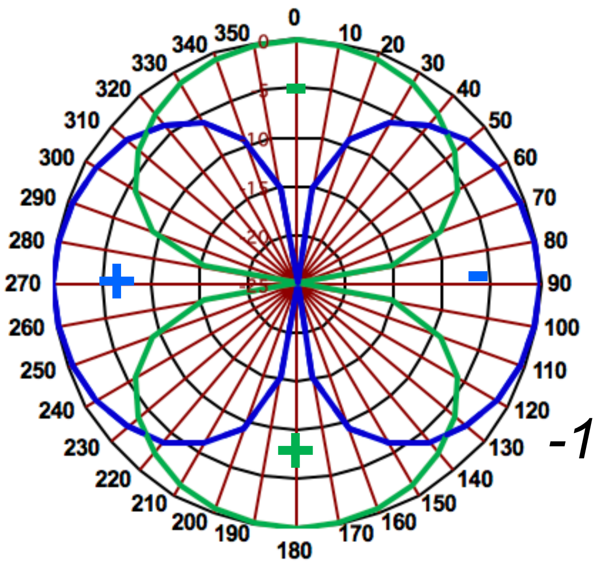
Micro OMNI = 1
Micro INFRA ≈ 0,660 (-10 dB arrière)
Micro CARDIO = 0,5
Micro SUPER ≈ 0,375 (-12 dB arrière)
Micro BI = 0

MATRICAGE



VALEURS AUX POTS

AVANT MATRICAGE



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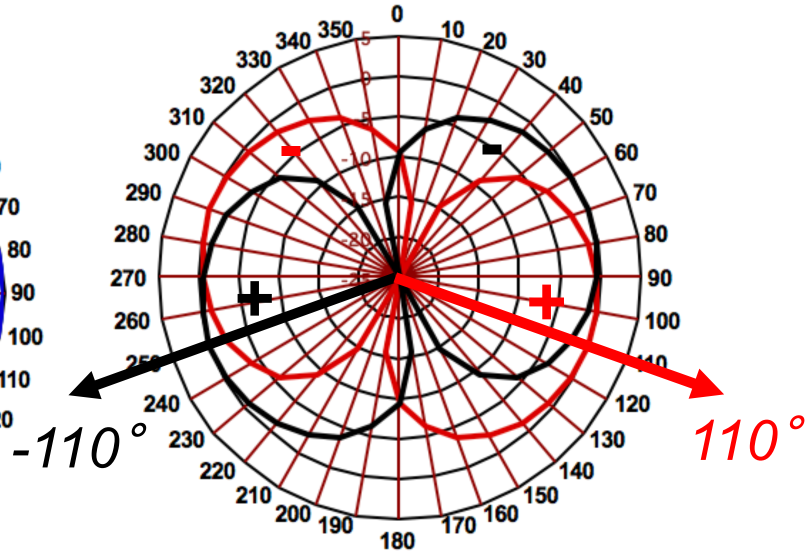


https://www.lesonbinaural.fr/EDIT/EXCEL/double_ms_simple.xls

W donne la directivité (pour 110°) :

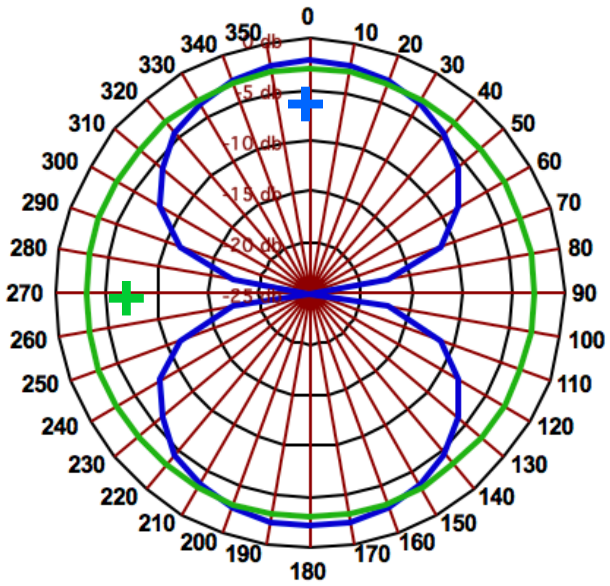
* Caractéristique du micro FRONTAL 1
1,000
* Caractéristique du micro FRONTAL 2
0,000

RÉSULTATS DE LA SOMMATION DES 2 MICROS VISANT LA MÊME DIRECTION	
* Caractéristique de directivité du micro après sommation	Niveau maximum du micro après sommation
0,474	3,5 dB

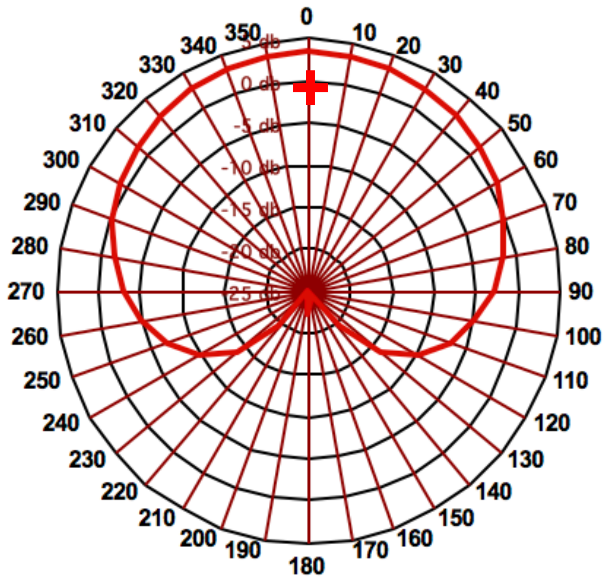
Différence de sensibilité entre le micro FRONTAL 1 et le micro FRONTAL 2
2,1 dB

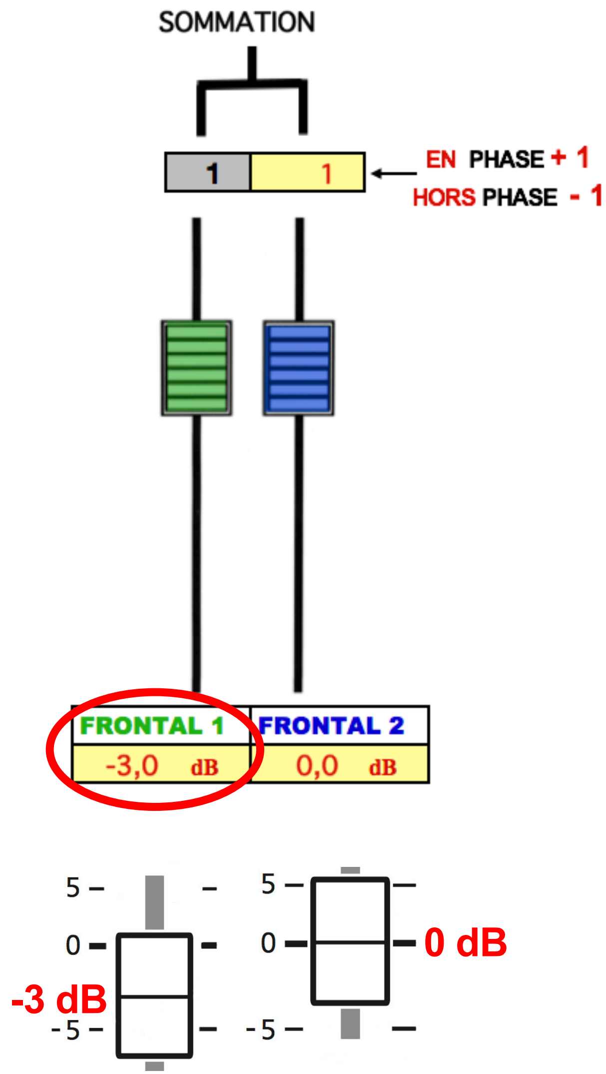
* NOTE :
Micro OMNI = 1
Micro INFRA ≈ 0,660 (-10 dB arrière)
Micro CARDIO = 0,5
Micro SUPER ≈ 0,375 (-12 dB arrière)
Micro BI = 0

AVANT SOMMATION



APRÈS SOMMATION





https://www.lesonbinaural.fr/EDIT/EXCEL/somme_2micros.xls

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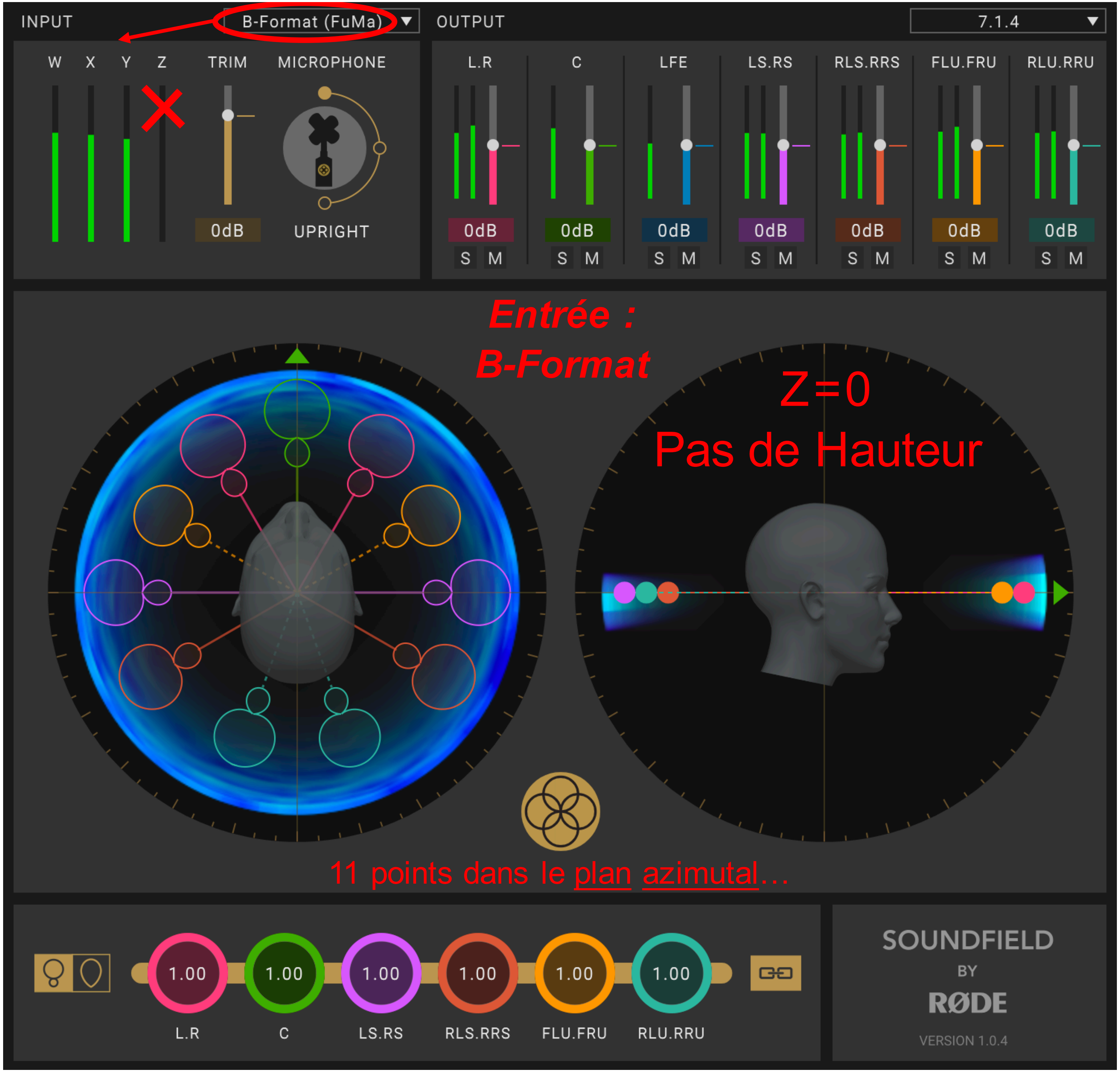
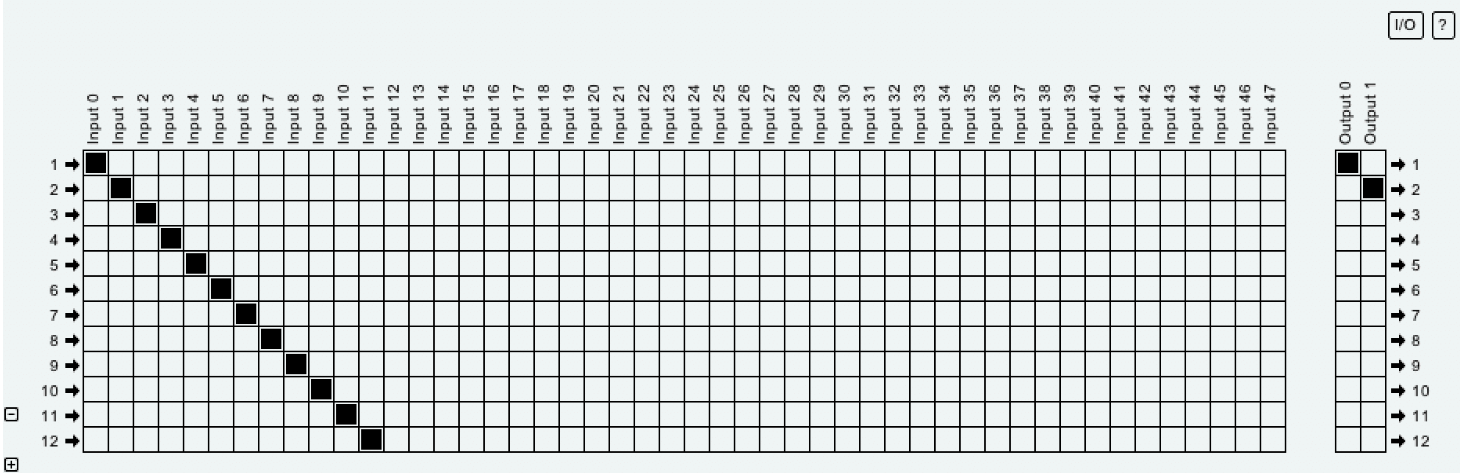




11 points dans le plan azimutal…

Élévation = 0
pour tous les points

http://www.cmap.polytechnique.fr/xaudio/mybino/

HRTF : MyBino

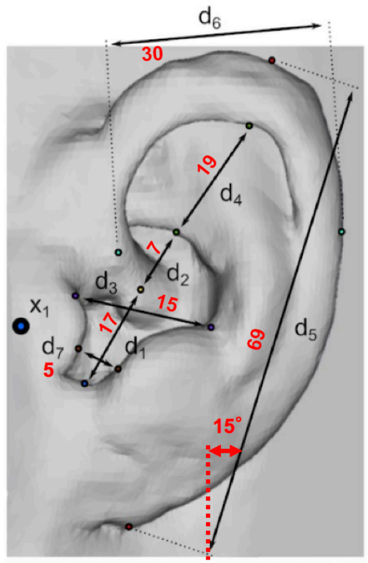
MyHead_1040.xhead

IRC_1040

ID : IRC40

Homme
Coiffure courte

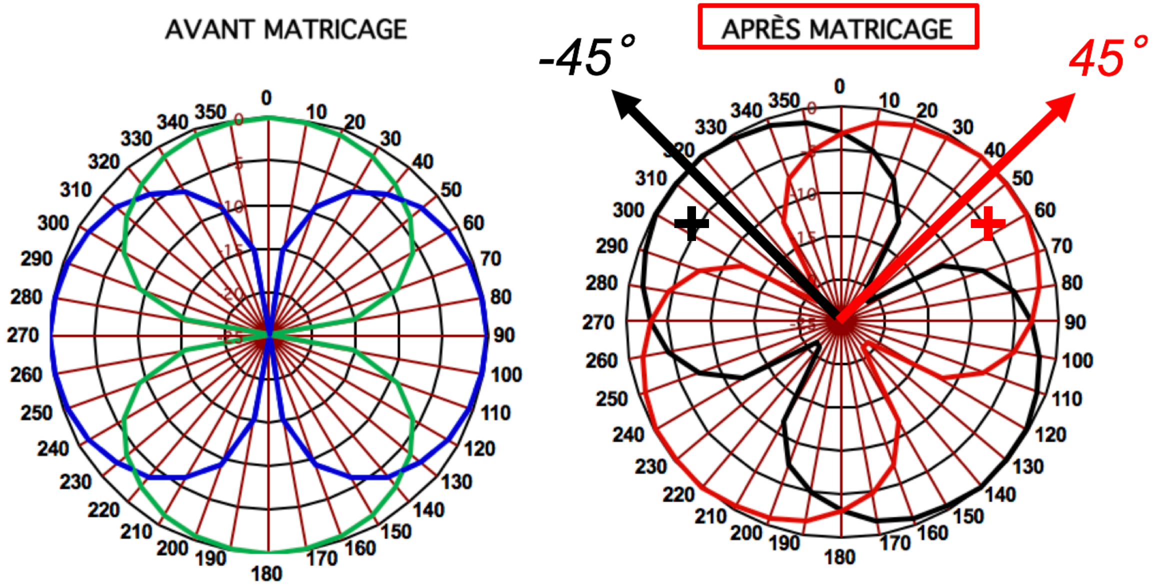
IRCAM
Automne 2002



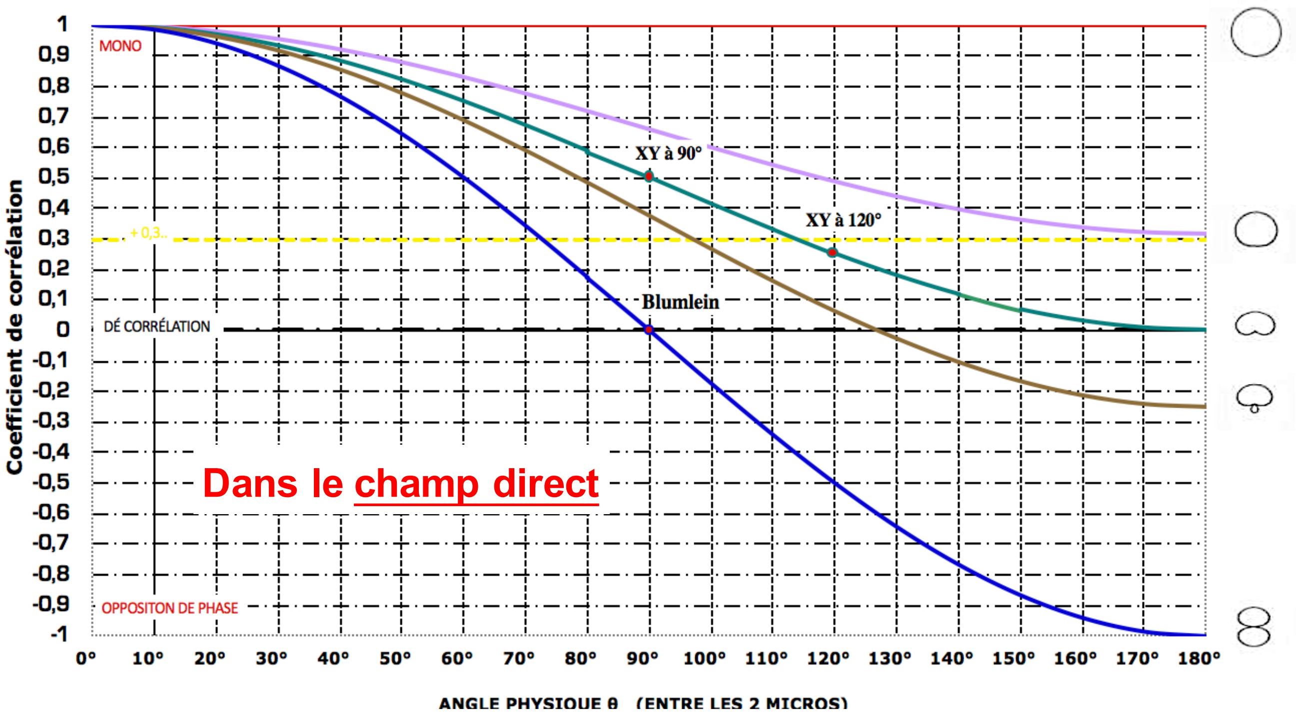
Moyenne pavillon CIPIC

d_1	d_2	d_3	d_4	d_5	d_6	d_7
18	8	18	19	64	33	5
17	7	15	19	69	30	5

Blumlein

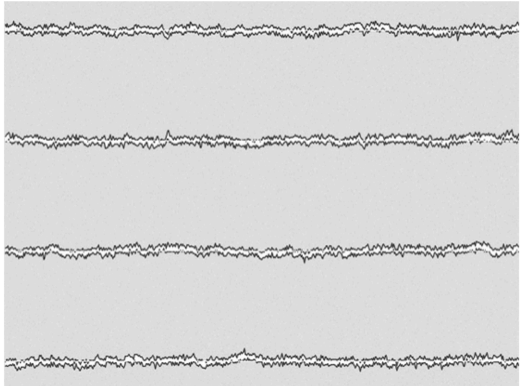


AVANT MATRICAGE
APRÈS MATRICAGE
-45°
45°



Coefficient de corrélation
MONO
XY à 90°
XY à 120°
+0,3
Blumlein
DÉ CORRÉLATION
OPPOSITON DE PHASE
Dans le champ direct
ANGLE PHYSIQUE θ (ENTRE LES 2 MICROS)

https://www.lesonbinaural.fr → Écoute : sons techniques



Bruit Rose sur 4 Pistes

Bruit Rose sur 4 pistes destiné au Multicanal en Quad
et à l'Ambisonique (courbe de réponse, équilibre, filtre...)

Dé-corrélation + 0,0 : de 0 s à 40 s
Corrélation + 0,25 : de 1 mn à 1 mn 40 s
Corrélation + 0,5 : de 2 mn à 2 mn 40 s
Corrélation + 0,75 : de 3 mn à 3 mn 40 s
Corrélation + 1,0 : de 4 mn à 4 mn 40 s

Attention au niveau -12 dBFS, coupe bas à 30 Hz.

00:00 -00:00

4 min 40 sec

Quad 4.0
L R Ls Rs
En **.WAV**
24 Bit / 48 KHz

Télécharger ©

Télécharger Bruit Rose sur 4 Pistes :

https://www.lesonbinaural.fr/EDIT/SON/bruit_rose_decorrele_correle_4pistes.wav

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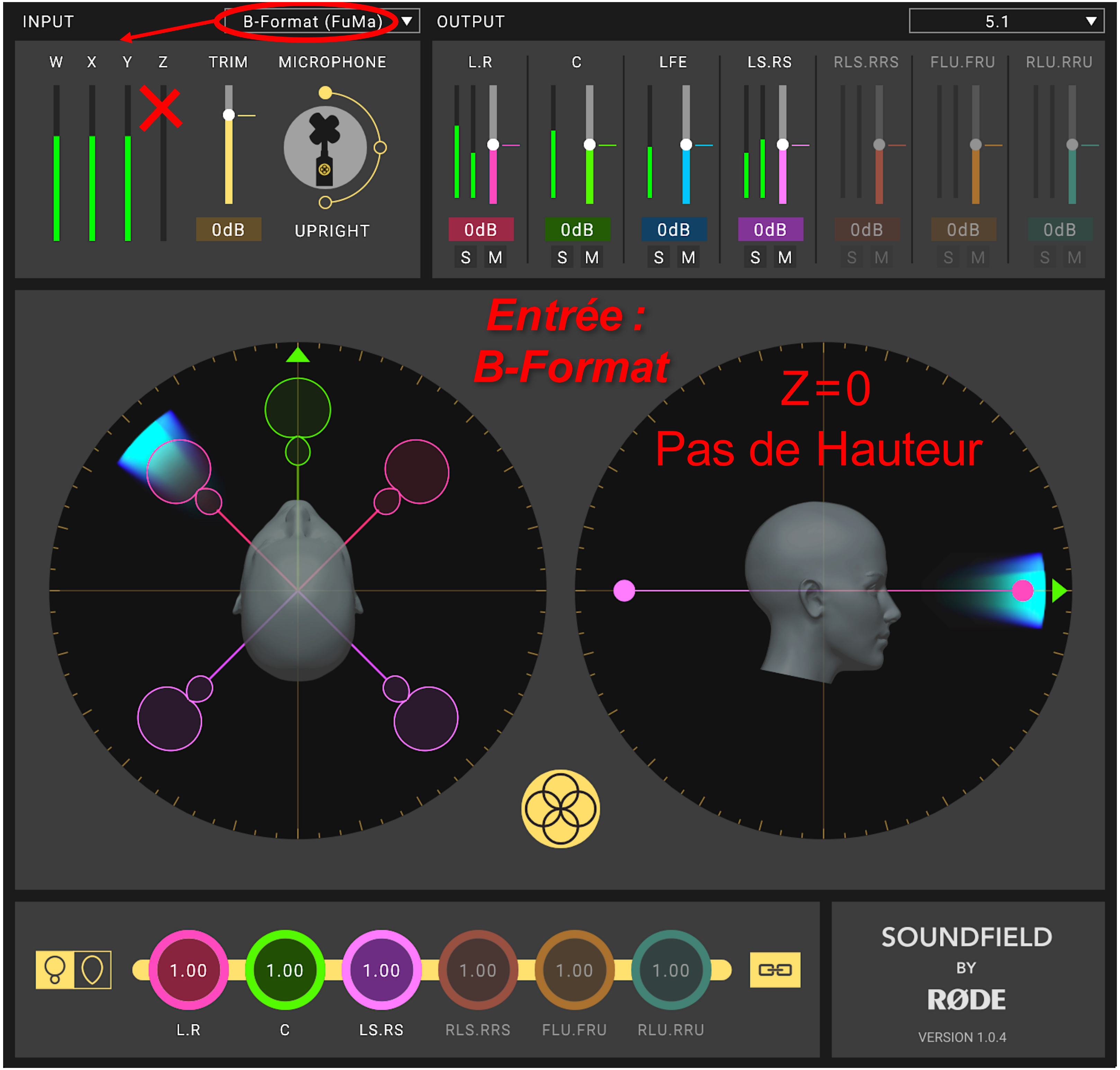
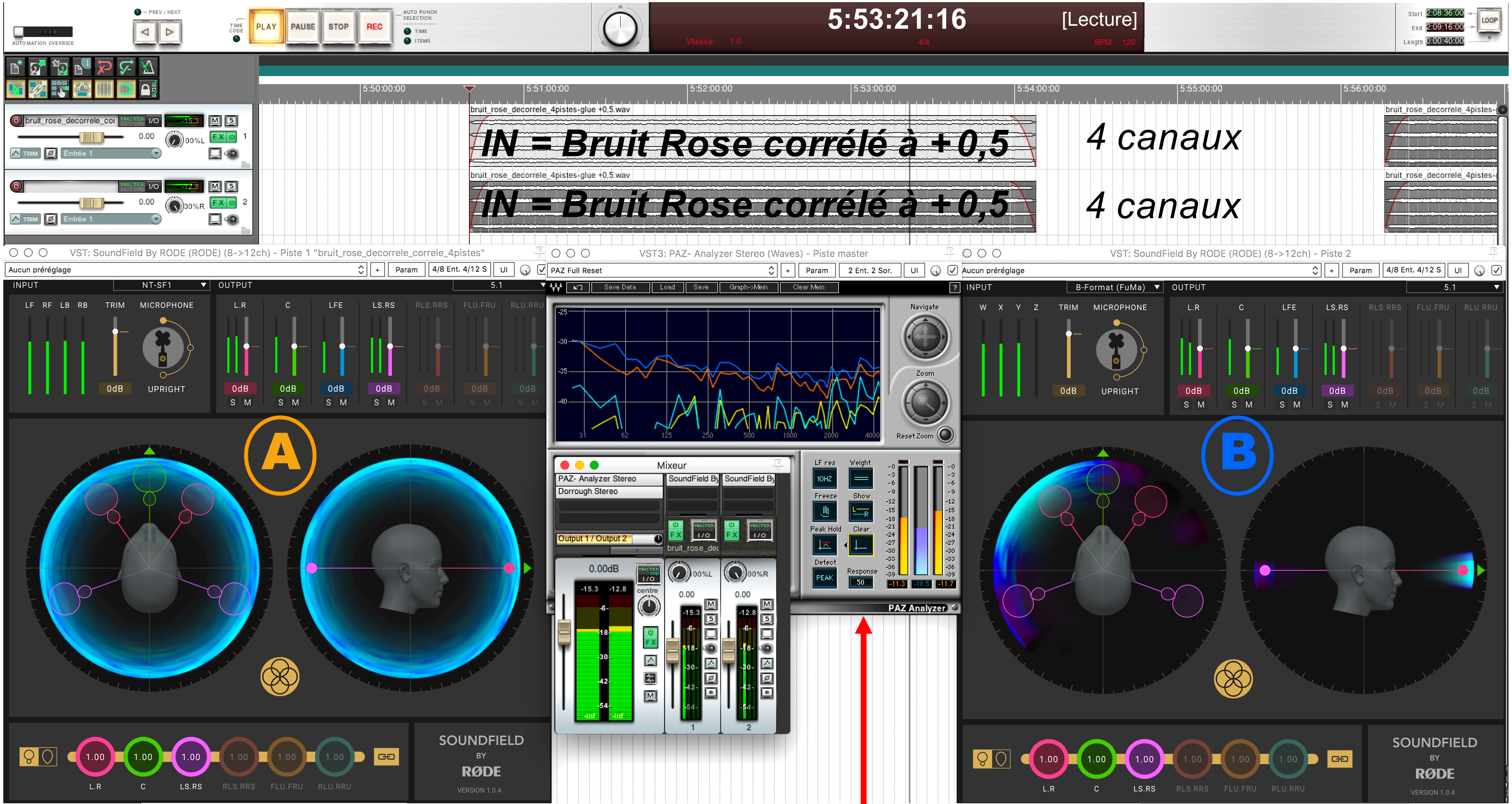
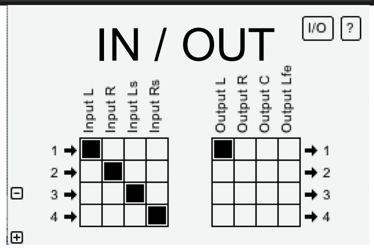
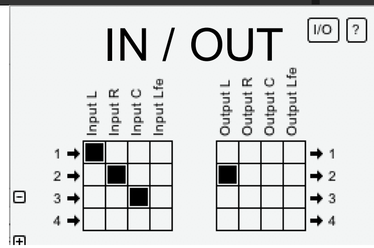


https://fr.rode.com/soundfieldplugin#footer_download

COMPARAISON entre le FORMAT A et le FORMAT B natif :







Analyse Plug-in PAZ Waves

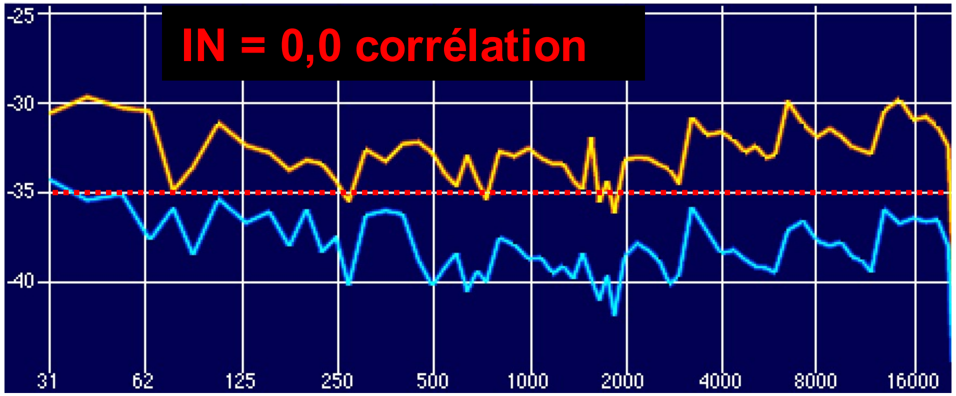
OUT = G Orange Format A

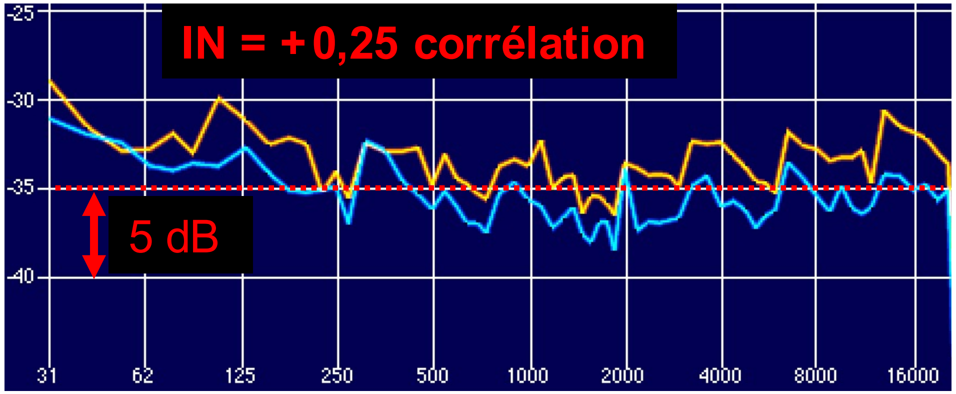
OUT = G Bleu Format B

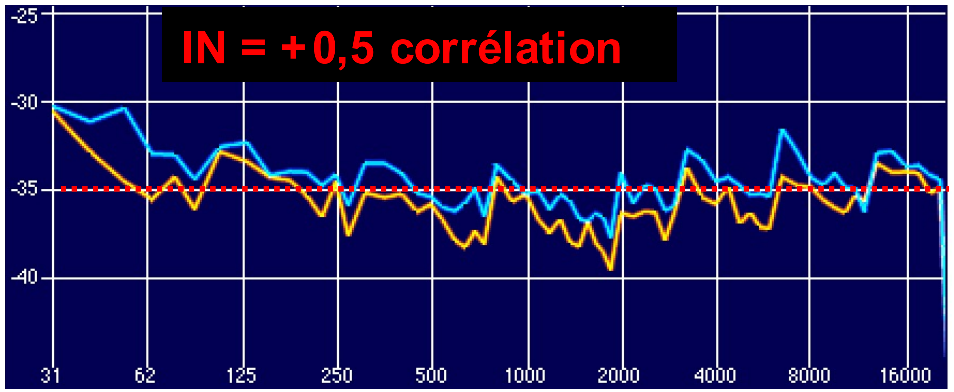
Analyse Plug-in PAZ Waves

OUT = G Orange Format **A**

OUT = G Bleu Format **B**

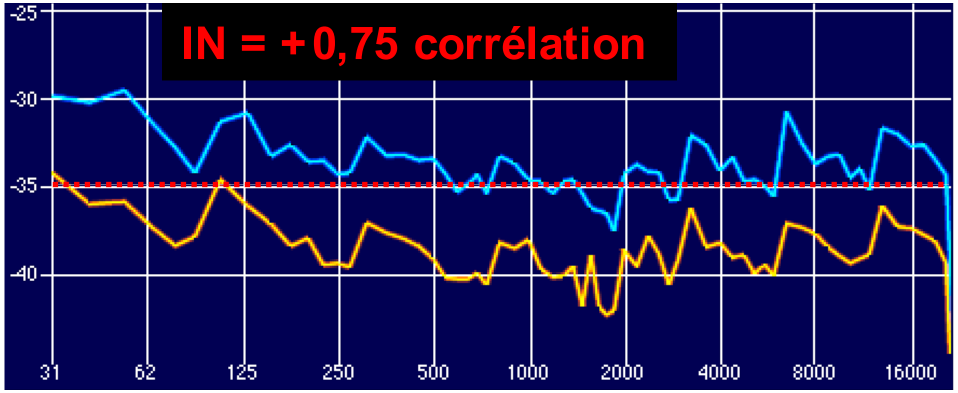


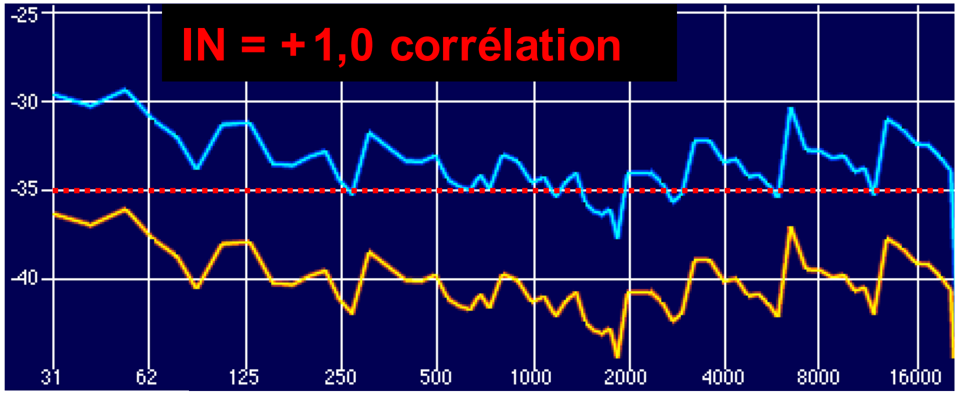




OUT = G Orange Format **A**

OUT = G Bleu Format **B**





5.1 Hyper cardio du 1er Ordre

V1.0.4

Entrée : Format A
Sortie : 5.1 (Orange)

Variation du niveau sonore en fonction de la corrélation

La corrélation ↗ :: Le niveau ↘
(de + 0,0 à + 1,0) (de 5 à 6 dB)

↳ **Baisse de niveau**
sur certaines sources…
(Basses Fréquences + corrélées)

?

Problème !!

Lié à la technologie "Tétraèdre" ?

Entrée : Format B natif
Sortie : 5.1 (Bleu)

Variation du niveau sonore en fonction de la corrélation :

La corrélation ↗ :: Le niveau ↗
(de + 0,0 à + 1,0) (de 3 à 4 dB)

NORMAL !

La Solution !!

Même si la composante Z n'existe pas…

YouTube Rechercher

Nine Inch Nails - 1 000 000 - Expérience 360 Ambisonic VR

826 vues https://www.youtube.com/watch?v=nmtoUkalxiU

MatthiasDuyck
Ajoutée le 11 déc. 2017

Cette expérience ambisonique à 360 degrés a été créée avec une simple scène Unity3D, les tiges libérées pour remix par Nine Inch Nails et le logiciel de station de travail FB360. Il permet à l'utilisateur de regarder autour de lui et d'entendre chaque joueur comme s'il se tenait autour de lui. Fonctionne sur l'application Youtube mobile et Windows avec Chrome, Firefox, MS Edge et Opera.

Catégorie **Musique**

Musique utilisée dans cette vidéo

En savoir plus

Écoutez de la musique sans publicité avec YouTube Premium

Titre	**1,000,000**
Artiste	**Ongles de neuf pouces**
Album	**Le glissement**
Auteurs-compositeurs	**Trent Reznor**
Concédé sous licence à YouTube par	**Audiam (Label) (au nom de The Null Corporation); Kobalt Music Publishing, AMRA, UMPI, LatinAutor et 7 sociétés de gestion des droits musicaux**

VR 360°
en 2D

La composante Z n'existe pas…

En n'a-t-on besoin ?

JOSEPHSON

C700 Users Guide

Josephson Engineering, Inc.
329A Ingalls Street
Santa Cruz, California
+1 831 420 0888
www.josephson.com

This Guide was previously published as the "Series Seven Users Guide" but since there are now more microphones in the Series Seven besides the C700, we decided to revise the title to avoid confusion.

C700/Series Seven Users Guide

Josephson C700 microphones provide extraordinary flexibility for the user. Any directional pattern from omni to figure-8 may be derived, and with the C700S, an unlimited number of "virtual microphones" pointing in different directions can be generated using the side-facing channel produced by the side-facing figure-8 capsule. This guide is intended to help the user understand the basic concepts of multiple capsule mid-side stereo and surround techniques, as made possible by the C700. One approach to explaining this idea is to reduce it to the mathematics of monopole and dipole transducers (but we'll save the math for the appendix, it's not needed to fully understand and use the microphone as an instrument.)

A major benefit of recording with the C700 is the ability to capture and save the raw audio components during a session, which can then be used to generate any number of directional patterns in playback. For a mono track, this allows the directional pattern to be adjusted during a track as a performer moves around, for instance.

This User's Guide applies to the both the C700A and the C700S. The only difference between the microphones is that the C700S has an additional channel for side information, that allows the direction of the main microphone pattern to be changed. We are mentioning only a few of the possibilities here; once you have a good understanding of how the patterns are added together to form new patterns, your own creativity and experience will take over in suggesting other mixtures of these channels that will produce other patterns.

The key concept to learn is that the microphone produces a separate output for each of its capsules. The user mixes these outputs together to derive any desired directional pattern. In the C700S, there is a third output, and adding this output into the mix allows the resultant directional pattern to be steered anywhere on the horizontal plane around the microphone. We have made a control console to derive patterns in the field, but we have found it much more effective to record the raw signals and matrix them afterward. The diagram of the control console is included in the Appendix for reference.

The microphone is rated for standard P48 phantom power, please note that all outputs of the microphone must be connected to phantom power before it will function properly.

Omnidirectional, or pressure microphone (W channel)

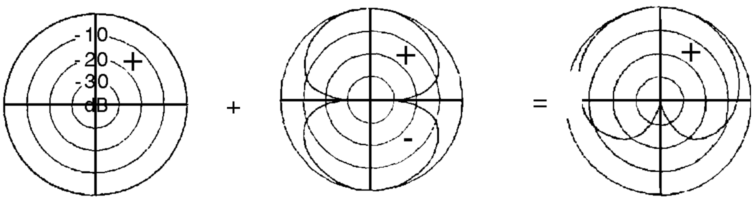
Omnidirectional microphones are the simplest. The moving element or diaphragm is open to sound on one side, and sealed on the other. Sound pressure causes the diaphragm to move inward, regardless of the direction. For low and mid-frequency sound waves, the wavelength is much larger than the size of the microphone – so the pressure wave simply flows around the microphone, pushing in on all surfaces regardless of its origin. These are called pressure microphones because they mainly respond to sound pressure. We use a small single diaphragm omni capsule for the W channel; this is the best way to assure superior off-axis tracking of the response pattern with minimal response changes.

Figure-8, or gradient microphone (X channel)

Figure-8 microphones have moving elements that are open to sound both front and rear. Both sides are equally sensitive. Sound pressure coming from the front causes the a positive electrical output. Sound pressure coming from the rear causes a negative electrical output. Sound coming from the side pushes equally in both directions, so there is no output. Figure-8 microphones are sometimes called pressure gradient or velocity microphones, because their output can be proportional to the gradient or difference between front and back pressure. We use a large dual-diaphragm capsule for the figure-8 signal because its symmetrical construction and high sensitivity produce a uniform front-back pattern with very tight nulls at the sides and a reduced noise floor.

Making a cardioid from an omni and a figure-8

The C700A and C700S include an omni microphone and a forward-facing figure-8 microphone. We call the omni or pressure signal "W" and the front-facing figure-8 "X". The microphone outputs are directly driven by the W and X elements. A whole family of directivity choices is available by mixing W and X. Mixing them at equal levels produces a cardioid. To understand this, remember that sounds arriving from the rear produce an output that's out of phase with the output that would result if they arrived from the front. If two equal but out-of-phase signals are mixed together, the result is zero. If one signal is a little bigger than the other, the result of mixing is simply the difference between the two signals.



W signal plus X signal equals cardioid

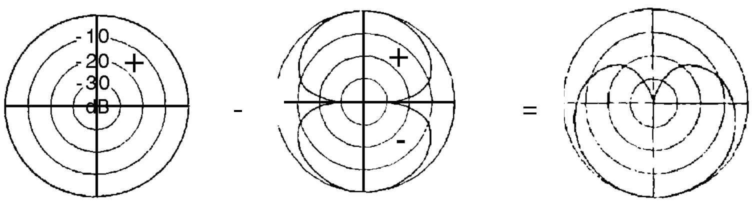
In the cardioid case, for sound coming from the front, the output from W and X are equal. Add them together, and the sum is double the value of the individual signals because the omni signal adds to the signal from the front side of the figure-8. For sounds arriving from the side, the W microphone still picks up with uniform sensitivity but the X microphone has no output, so the summed output is the same as for the W microphone alone. For sounds arriving from the rear, the omni and figure-8 signals are again equal but now out of phase, so the summed signal is zero. Note the + and – symbols, which are there to remind you that the front side of the figure-8 is in phase with the W signal, while the rear side is out of phase.

Other patterns

All of the possible patterns can be imagined by thinking of the X (figure-8) pattern, and what happens to the signal if you add some W (omni) to it. If the X signal is constant, and a small amount of W is added, the front lobe of the figure-8 grows a little because the signals add, and the rear lobe shrinks a little, because the W signal cancels the out-of-phase rear lobe of the X signal. When the W signal is increased to a point 10 dB below the X signal, the pattern has changed to a hypercardioid, with the rear lobe about 10 dB reduced from the original X signal. Adding W to X also moves the position of the figure-8 null, which is normally at 90° and 270°. Remember, when the W and X signals are equal, the side nulls are moved all the way back to 180° and merge to form a cardioid pattern. Whenever the W signal is less than X in the mix, the null will be somewhere between the sides and the rear. If the W signal is larger than the X signal, there will never be a complete null, but rather a near-omni pattern that is weighted toward the front, assuming that W and X signals remain in phase. There are different names for these patterns, including "hypoardioid" "wide cardioid" and "subcardioid." Hypercardioid is usually defined as 9.5 dB more of the X signal than the W, supercardioid occurs when X signal is 4.6 dB higher than the W.

Reverse direction

What happens when you reverse phase of the X signal? All of the same patterns described so far, but facing toward the rear of the microphone rather than the front. There is of course some change in high frequency response due to the fact that the W capsule is facing forward, but for low and middle range frequencies it is truly omnidirectional. This operation can be described either as the W signal *minus* the X signal, or as the W signal plus the *minus* X signal.



W signal minus X signal equals reversed cardioid

Summary of Summing

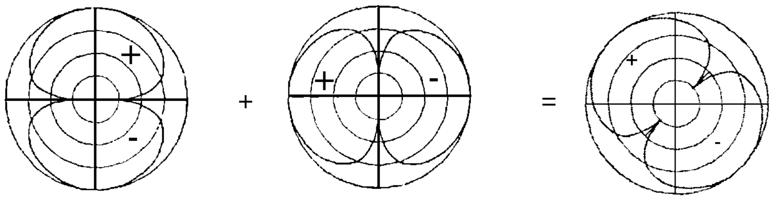
- Control the pattern by selecting the ratio of W to X

Steering (C700S only)

The C700S includes another figure-8 capsule at 90° to the "X" capsule, providing a signal from a side-facing microphone capsule called "Y" to allow the "X" figure-8 signal to be steered in any direction. If signals from two coincident, orthogonal figure-8 microphones are summed in any ratio, the result is always still a figure-8 pattern, but pointing in a different direction. Remember that all the patterns derived with the W and X signals were facing along a front-back axis. Now consider what would happen if this axis were rotated. The Y signal faces left, so patterns derived with all Y and varying amounts of W added will be about the same as the patterns derived with X and varying amounts of W – only now they point 90° to the left. If we use a mixture of X and Y, the resulting pattern will be pointing anywhere from 90° left (all Y) to 45° left (equal proportions of X and Y) to straight in front (all X). If we continue around and invert the phase of the Y signal, it's the same as having the Y capsule pointing right, so we can now derive all the patterns but pointing anywhere from 0° to 90° right. Continuing around, if we invert the phase of the X channel, the patterns are facing toward the left rear, and if we invert the phase of both the X and Y channels, the patterns face toward the right rear.

Summary of Steering

- Virtual direction of X may be changed by adding Y
- Any pattern created with W and X is therefore rotated by adding Y



X signal plus Y signal equals left-front facing figure-8

To recap, we can rotate the direction of the figure-8 signal any amont by adding a different amount of Y to X. The resulting pattern, regardless of the relative amounts of the two signals, will always be figure-8. We can call the summed X and Y signal "D." The front lobe of the D signal will be pointing somewhere on a 360 degree circle according to the relative phase of the components:

(X) + (Y) D = LF	(X) + (-Y) D = RF
D = LR (-X) + (Y)	D = RR (-X) + (-Y)

MS equivalent

If you are familiar with Mid-Side or MS stereo, you will recognize the Y channel as being the same as the S channel in MS. Any MS technique can be realized with the C700S. Use the W and X channels summed to produce the forward-facing M signal of the desired pattern, and the Y channel for the S.

XY equivalent

The most basic intensity stereo pickup consists of cardioid microphones spaced 45° either side of the center line. This is often called "XY stereo." Note, this is a different use of "X" and "Y" than the names we've adopted for the C700 output channels. XY stereo left and right channels can be created from the M and S signals exactly as described: left is M+S, right is M-S.

Experiment

Start with the basic X signal and move the sound source around to the side of the microphone. Notice the sharp null at 90°. Sounds that have asymmetrical waveforms will sound different (as with all figure-8 microphones) when moved to the rear of the mic, due to phase reversal. This is particularly true if the sound source is the person who's listening to the output through headphones, because of mixing the direct in-the-head sound paths with the headphone path. Begin adding a little W signal, for example at 20 dB below the X signal. Notice that the sound in front has changed character; the proximity effect is reduced somewhat. The side-facing nulls have moved back 10-20° and the rear sensitivity is reduced. Continue adding W to the X signal, and the null will continue to move toward the back. Adjust the ratio of W and X to be equal, and note that the null is at 180°.

Tracking

We recommend that the raw W and X signals be recorded on a tracking master (and the Y, if a C700S microphone is being used). That way, all the directional pattern and angle choices can be made in mixdown and your options for pattern selection remain.

For vocals, this can be particularly powerful. After the session, you may decide that using a more omnidirectional pattern with the singer up close to the microphone is preferable to working at a greater distance with a more directional pattern. The ratio of direct to "room" or ambient/reverberant sound can be controlled either way – by working at different distances or by controlling the pattern of the microphone.

Equalization and Proximity Effect

Another powerful tool available to the C700 user is selective equalization of the omni and figure-8 capsules. Traditional single-output microphones have a fixed set of characteristics including directional patterns that vary with frequency and distance. You might decide for instance that you need a tight hypercardioid pattern above 2 kHz for control of high frequency ambient sound, but have a vocalist who is moving around a lot causing low frequency response changes due to proximity effect. The desired pattern might nominally be a cardioid in the mid-band, but tending to omni at the low frequencies and hypercardioid at high-mid frequencies. To accomplish this, you would roll off the low frequencies and boost the mid-high frequencies of the X signal.

Stereo on a Simple Mixer

The simplest way to use a C700S for stereo with a mixer requires only a wye cord built with one of its outputs reversed in phase. See Appendix C1 for a wiring diagram, this cable is not included with the microphone. Four mixer channels will be needed. The W and X channels are connected to the first two mixer channels, which are panned center. The Y channel is connected to an inverting wye cord; the straight-through connector is connected to the third mixer channel "Y" which is panned left, the inverted connector is connected to the fourth mixer channel "-Y" which is panned right. Start with just the W and X channels, trim the channel gain controls so that a cardioid (null at 180°) results when the faders are at equal positions. Confirm this by placing a sound source at the null and adjusting W and X channel gain for a deep drop in pickup toward the rear. Now move the sound toward the front of the mic and begin to increase the level of the Y and –Y channels. Adjusting the relative levels of Y and –Y will shift the image left and right ("pan" or "balance") and adjusting the ratio of the Y signals to the W/X signals will adjust the "width" or "focus."

Using a basic mixing console or digital audio workstation, you can derive all of the channels needed for stereo, center-channel-stereo, 5.1 or 7.1 surround. In practice, the mixdown or balance engineer will want to adjust the ratios to produce a convincing sound image, but these basic relationships for surround are a good starting point:

Left Front = W+X+Y	Left Rear = W-X+Y
Center Front = W+X	Right Rear = W-X-Y
Right Front =W+X-Y	Low Frequency Effects = W (lowpass filtered)

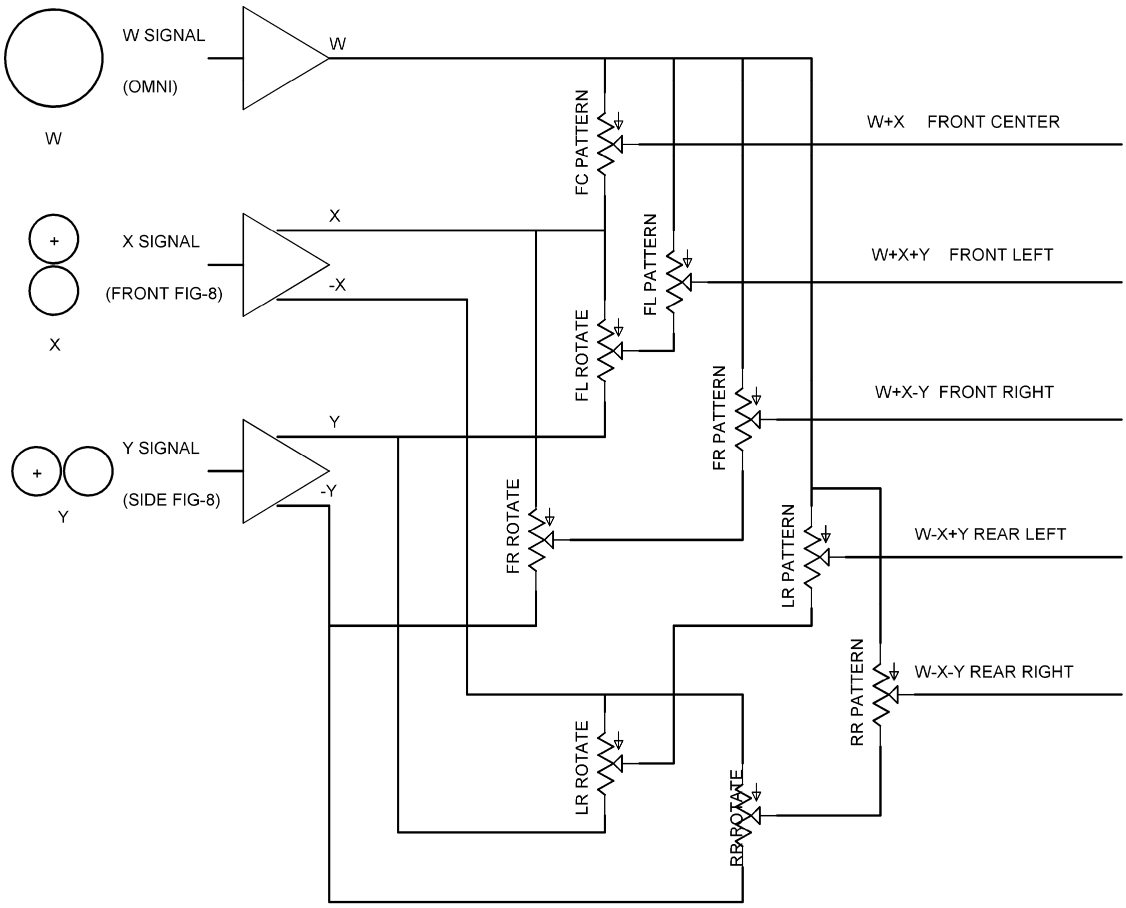
Mixing for Surround

A practical way to derive surround channel mixes from raw C700S W, X and Y signals uses a mixer or DAW with multiple aux busses. Three busses are sufficient if the channel strips have the capability of reversing phase from the busses; if they don't, you'll need five busses, one each for W, X, -X, Y and –Y.

If you use digital processing to produce the "inverted" signals, be certain that the DSP doesn't introduce unwanted phase anomalies in the signal due to changes in processing latency. Many digital workstations have variable delay depending on the amount of processing being done. This would cause shifts in stereo imaging due to differences in phase between the channels.

Appendix A - C700S Control Console (not included)

Josephson designed a microphone preamplifier/control console for use with the C700S. It was never put into production, as users have opted to record the tracks independently and mix down using their normal processing tools. The block diagram of the console is shown here for reference, as it explains the process in analog terms. Five outputs are provided, each with independent pattern and rotation controls. A sixth output, for "low frequency effects" may be derived from the W channel through a low-pass filter.



Appendix B – For the mathematically inclined

The omni or W microphone signal behaves as a monopole scalar pressure transducer, unaffected by direction (at low and mid frequencies, where the size of the microphone housing isn't a significant fraction of a wavelength). The figure-8 (X and Y) microphone signals behave as dipole transducers with a response that varies with cosine of the arrival angle. In the C700S, we use a combination of the two figure-8 signals X and Y to yield a new figure-8 pattern D pointed at a defined angle θ. We use capital letters to refer to the signals themselves and lower-case to refer to the proportions of each signal in a mixture.

For these formulas, consider the output as a combination of W and D signals where $w+d=1$. Some common ratios are

Omni: all W
Subcardioid/hypocardioid/wide cardioid: 0.66W+0.33D
Cardioid: 0.5W+0.5D
Hypercardioid: 0.33W+0.66D
Supercardioid: 0.25W+0.75D
Figure-8: all D

Steering or rotation of D is achieved by adding X and Y signals. The main axis of the pattern is located at an angle ϕ relative to the front of the microphone. ϕ changes linearly from 0 to 90° by varying the ratio of X to Y signals in the mix. For a ratio of x and y such that $x+y=1$, $\phi=-90y$.

ϕ	X	Y
90º left	0	1
45º left	.5	.5
0 º	1	0
45º right	.5	-.5
90º right	0	-1

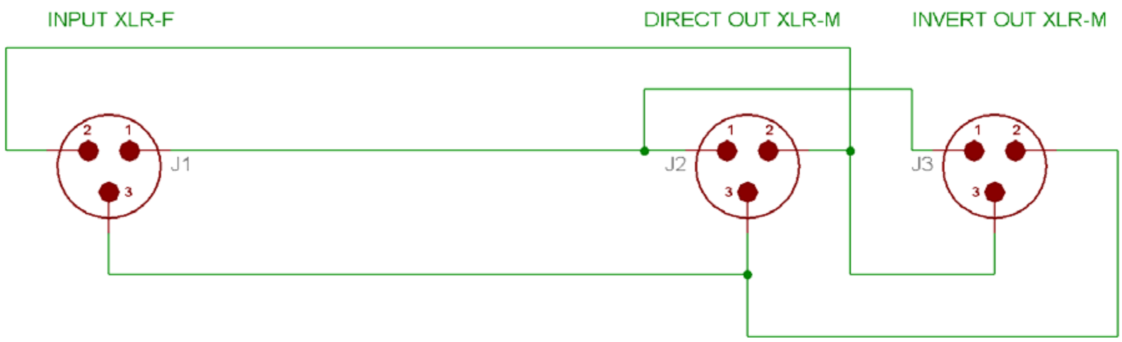
ϕ is not restricted to the front quadrants. By reversing the phase of the X signal, it can be pointed to the rear. ϕ lies within one of the four quadrants depending on the phase of the X and Y components. In the C700A there is no Y signal, so ϕ is fixed at 0 or 180° depending on whether X or –X is used.

X+Y LF	X+ -Y RF
LR -X + Y	RR -X + -Y

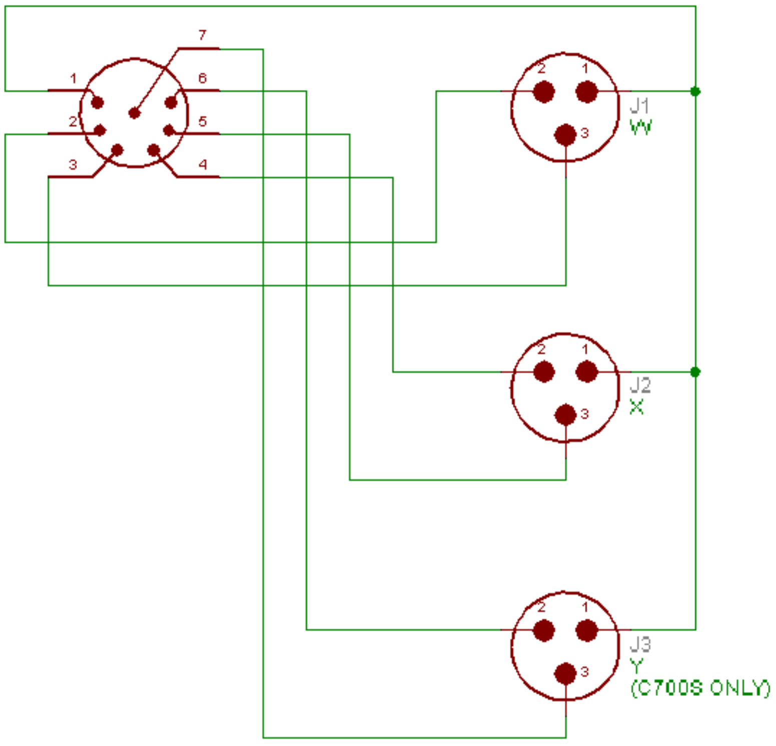
The response *r* at azimuth angle θ (where θ is the angle from the main axis of D) is found with the formula $r(\theta) = w + d \cos(\theta)$.

Appendix C1 – Recording Direct to Stereo (cable not included)

It is possible to make a basic stereo recording with an ordinary mixer without using the advanced capability of the C700S, see "Stereo on a Simple Mixer" above. The phase-reversing wye cord that's needed for this is shown below.



Appendix C2 – Output Adapter Cable for C700 (furnished with microphone)



This is the standard output breakout cable furnished with the microphone.

Warranty

Josephson microphones are warranted to be free of defects for five years from the date of original purchase. If purchase documents are not available, the warranty period begins when the microphone was shipped from the factory. Josephson Engineering will, at its option, repair or replace any microphone that fails, providing that it is returned to the factory prepaid and has not been abused or altered.

There are no user-serviceable parts inside Josephson Engineering microphones. Disassembling a Josephson microphone will void its warranty.

For service information please contact Josephson at 831-420-0888. Repair shipments may be sent to:

Josephson Engineering, Inc.
329A Ingalls St
Santa Cruz CA 95060

RoHS Certification

Josephson Engineering, Inc. certifies that the C700A and C700S microphones bearing the CE mark conform to the applicable requirements of the European Union directives as follows:

Machinery 93/68/EEC	Exempt – passive sensor
Low Voltage 93/68/EEC	Exempt – passive sensor
EMC 93/68/EEC	Exempt – passive sensor
RoHS 2002/95/EEC	Compliant for Hg, Cd, Cr6, PB, PBDE and Pb

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BPT

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... ready for 5.1 and beyond ...

Operating Instructions

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www.nevaton.eu

https://www.for-tune.de/shop/nevaton-bpt-surround-mikrofon.html?language=de

GETTING STARTED: Practical Application of the BPT-Microphone (case studies)

Recording with the BPT is easy: just position the microphone like you would do with any other main- or spot-microphone. What you need to know on your first recording session with the BPT is the meaning of the color-code of the short break-out cable Neutrik 7-pin XLR to 3 x 3-pin XLR, which comes with the microphone - so here it is:

'Break-Out Cable' COLOR CODE (for upright position of the BPT):

GREY (or BLACK) 3-pin XLR: carries signal of capsule 1, which is "L"

RED 3-pin XLR: carries the signal of capsule 2, which is "R"

YELLOW 3-pin XLR: carries the signal of capsule 3 "Center", which is the top capsule

Please do not forget that when the BPT is positioned 'upside down', the signal content of capsules 1 and 2 will be reversed: capsule 1 will deliver "R" and capsule 2 will carry a "L" signal.

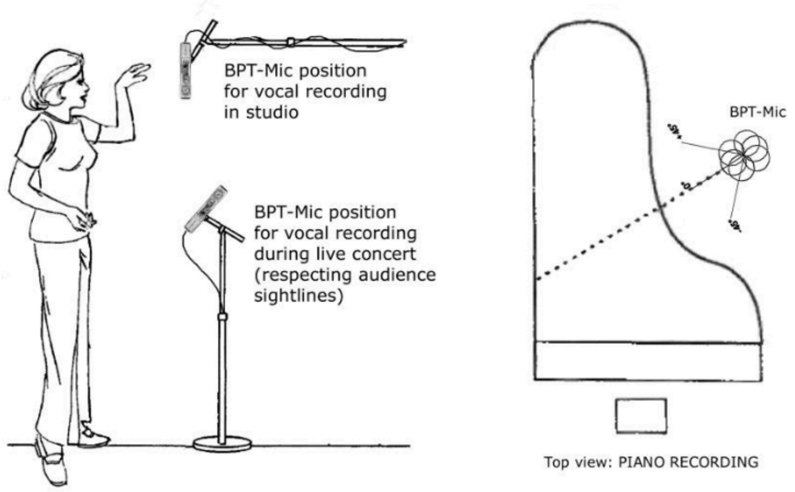


Fig. 1 (left): Of course the BPT microphone gets positioned like any other mono or stereo microphone …. The big ADVANTAGE is that you can record a MONO (center capsule) and a STEREO signal (L+R capsule) at the same time ! This gives you many more options and creative possibilities on mix-down. When recording, you could set all three capsules to cardioid, or choose to use the L&R capsule as "Blumlein-Pair" and switch the center capsule to cardioid, hypercardioid, fig-8 or even omni … On the **(right)** you see how the BPT-microphone can be set-up for piano recording. You could try a similar setup in front (or overhead) of a jazz-drum kit, a violoncello … or whatever other instrument you need to record …

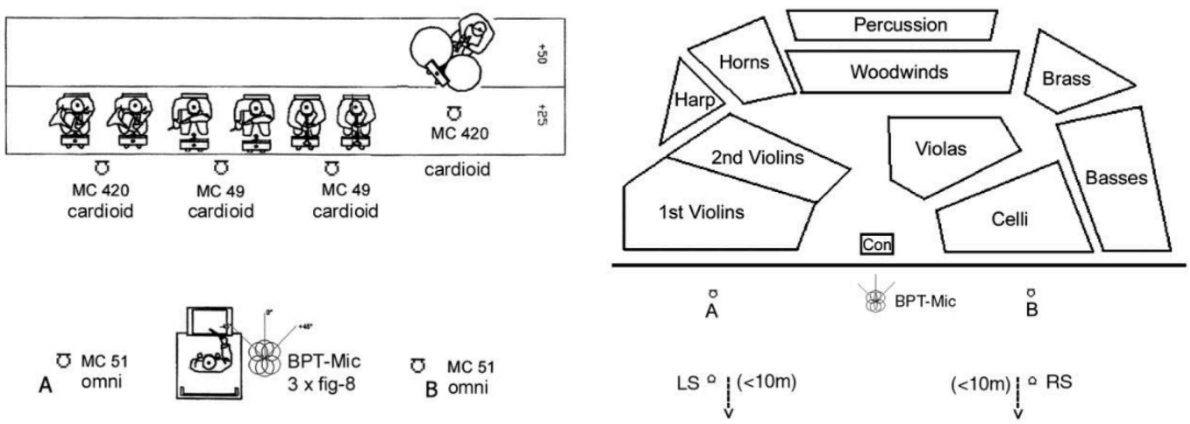


Fig. 2 (left): For the recording of a musical ensemble the BPT can become the main-microphone, which can be accompanied by large-AB style "outriggers" (omnis) in order to create a stronger "spatial impression" for the recording. Of course this AB-BPT main system can be accompanied with various spot microphones (here: chosen from the Nevaton microphone palette ☺) **(right)** The same setup works very well also for a large symphonic orchestra. In this case a surround recording is pictured, with two cardioids at several meters distance from the front microphones, facing the rear wall.


Rotation de 45°
Nevaton BPT
B-Format
ou 5.1
W (Omni)
Y (Bi)
X (Bi)
© Nevaton

1. Preamble – "Why build a 3 capsule microphone ?"

As the name already indicates, the "Blumlein-Pfanzagl-Triple" (BPT) makes use of a third figure-of-eight capsule, which is added to the well known Blumlein-Pair of two figure-of-8 capsules, which have an included angle of 90° and are oriented at +/- 45° towards the sound source. The new capsule is pointed directly towards the middle of the sound source at 0°.

Why add a third capsule at all, as the Blumlein-Pair already captures 360° ? The Blumlein-Pair arrangement is appreciated by many as 'very naturally sounding' (see [Streicher and Dooley, 1985]), which is attributed also to the fact that the signals of the two channels are largely decorrelated (meaning: very different) also at low frequencies, which is important for good spatial impression. This is a unique feature which is almost exclusively reserved to crossed figure-of-eights (when it comes to 'one-point' or closely spaced microphone techniques), as all other microphone patterns (with hypercardioids as an exception) do not provide similar signal-separation for low frequencies.

As a one-point recording technique with highly directional microphones the BPT-mic is characterized by providing precise localization.

Tomlinson Holman described this behaviour, which can also be found with figure-of-eight microphones, with the following words: „ … The system [i.e. Blumlein-pair] aims the microphones to the left and right of center; for practical microphones, the frequency response at 45° off the axis may not be as flat as on axis, so centered sound may not be as well recorded as sound on the axis of each of the microphones …" (see [Holman, 2000]).

This is of course one of the main reasons why it makes very much (sonic) sense to add a third figure-of-eight microphone or capsule to the "Blumlein-Pair" arrangement. In addition, this gives the sound engineer a high degree of freedom, of how broad / (i.e. 'stereophonic') or narrow ('monophonic') he would like the sound image to be, by simply varying the level of the Center-capsule in relation to the L and R capsule.
For normal recording applications, the level of the center microphone will usually be set between -6 dB to -3dB in relation to the L and R microphone.
Deviations of this are – of course – possible and for some applications it might make sense to set it at -10dB or – if it is necessary to emphasize the center of the sound-source – it may be set to 0dB or even more.

In order to shield off unwanted signals from the rear it is preferable to place an acoustic barrier (for example in the form of a sound-absorptive acoustic panel) behind the BPT microphone. This helps to achieve a better direct-/diffuse sound (i.e. direct signal /reverb) ratio and enables the sound engineer to move the system further away from the sound-source thereby usually achieving also a better balance between instruments (or instrumental groups) in case of larger ensembles.
(Rem.: If the absorptive panel cannot be used for optical reasons, the three capsules of the microphone can be switched to cardioid pattern and attenuate sound from the rear in this manner. However – if possible, the use of an absorptive panel is recommended, as this will result in a much clearer sonic image achieved in the recording, since diffuse sound arriving from the back is being eliminated more effectively)

2. Using the Center-Capsule C to “zoom in” on the sound source

Rising the level of C ‘focuses’ or ‘zooms in’ to the centre of the sound source, while – as a side effect, desirable or non-desirable – increasing the ‘monophonic’ part of the overall signal. Theoretical calculations (using the “Microphone Assistant V.2” software - see [Wittek, 2002]) have shown that if C is set at the same level as the L and R microphone the result will be a gradual increase and boost of volume by 3 dB towards the middle of the sound-source.
This can be of advantage in connection with broadcast-applications (news, location recording, etc.), as it is common practice to work with (highly) directional microphones in order to create an “acoustic focus”.
If the center capsule’s signal is adjusted to a relative level of -10dB in relation to the L and R capsule, almost no emphasis or gain is added to the sounds coming from the centre of the sound-source.

It is of advantage to use the additional fig-8 center-capsule, as it helps to achieve a more stable sound-image especially if sound sources with complex (i.e. frequency dependent) radiation characteristics (as is the case with many music instruments) need to be recorded.

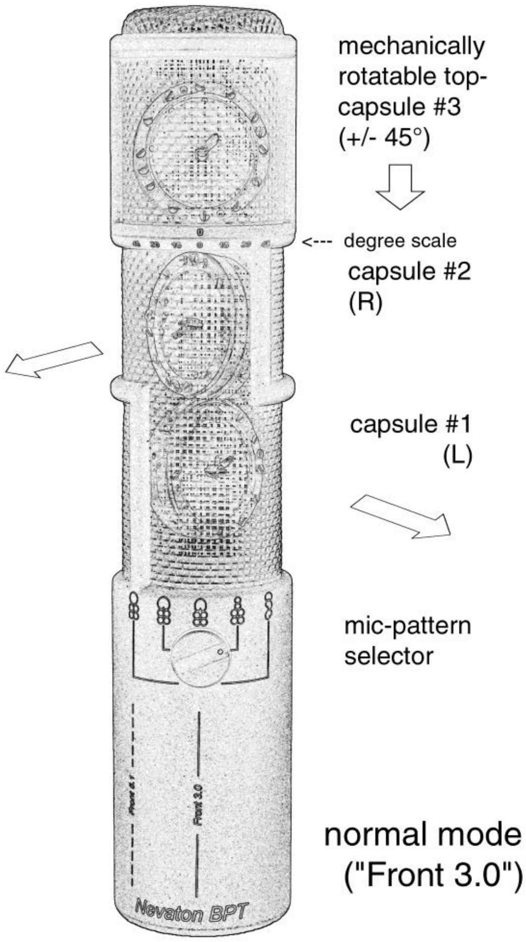


Fig. 3: BPT-mic (normal mode)

2. Use of a HPF on the Center-Capsule (for enhanced de-correlation)

Results of various researchers have shown that low signal-correlation (cross correlation) is necessary at frequencies below approx. 200Hz in a stereo (or binaural) signal in order to have good spatial impression as a listener (see –among others - [Hidaka et al, 1995] and [Griesinger, 1998]).
Due to the 90° physical offset angle between the L and R capsule, which have fig-8 characteristics, the signal of these two will be completely de-correlated over the entire frequency response (to be precise: for diffuse sound signals, arriving from all directions). Adding the signal of the center capsule (by mixing it in on the stereo-bus, or routing it to the center-speaker in a 5.1 surround setup) signal perception will certainly become more correlated over the entire frequency band. For the sake of good spatial impression, correlation for bass-frequencies should be kept at a minimum: the simplest way to achieve this is by setting a High-Pass-Filter (HPF) on the signal of the Center capsule, with a cut-off frequency somewhere in-between 90 and 160Hz, depending on the frequency response of the sound-source, the acoustics of the room, etc.

This can be done on the mixing desk, or already while recording by use of the High-Pass Filter-control (fc=90Hz) on the rear of the BPT-mic set to the position on the right.

3. Choosing the right distance between BPT and sound-source

Concerning the normal "Blumlein-pair" microphone technique Tomlinson Holman stated that "… The system makes no distinction between front and back of the microphone set, and thus may have to be placed *closer* than other coincident types … ". There is certainly a point to it, as in a normal concert situation the rear lobes of the fig-8 microphones will mainly pick up reverb from the hall and therefore the direct/diffuse sound ratio will be affected in a negative way.

If the natural mix of these two signal components is too reverberant, an absorptive panel should be placed behind the BPT.

When using an absorptive panel, the soundfield of the rear hemisphere is kept away from the BPT and therefore the critical distance (i.e. reverberation radius; see appendix) is effectively doubled for the microphone, which means it can be moved away from the sound-source by the amount equivalent to the reverberation radius, while the direct/diffuse sound ratio will remain the same (due to the addition of the absorptive panel behind the BPT).

A second important factor needs to be considered for the placement of the BPT microphone, which is the physical opening angle of the L and R fig-8 capsules (angled at +/-45° relative to the sound source). Even though the Blumlein-Pair is said to have a correct recording angle of 120° (see [Streicher and Dooley, 1985]), it is certainly preferable if the main body of the sound source stays well within the included angle of 90°, as otherwise there is the danger that a sound source in front on the far left might be picked up by the rear lobe of the right fig-8 capsule of the Blumlein Pair and therefore will be replayed not only from the left side (left capsule signal), but also from the right side, but with a phase inversion.

In [Faulkner, 1981] the question of optimal microphone positioning in order to achieve a good balance between all instrumental groups of an orchestra in a concert location has been discussed. As sound engineer Tony Faulkner points out, the "old rule of thumb" to position the main (stereo) microphone at a distance of about half the width of the orchestra away from the sound-source can be enlarged to the whole width in case of highly directional microphones, which is the case of fig-8 microphones (and even more so, if an absorptive panel is used). Therefore a microphone position in the 6th up to the 9th row, as pictured in fig.4, might not only be "good seat" position for a listener, but also for a Blumlein-Pair or BPT-mic.…

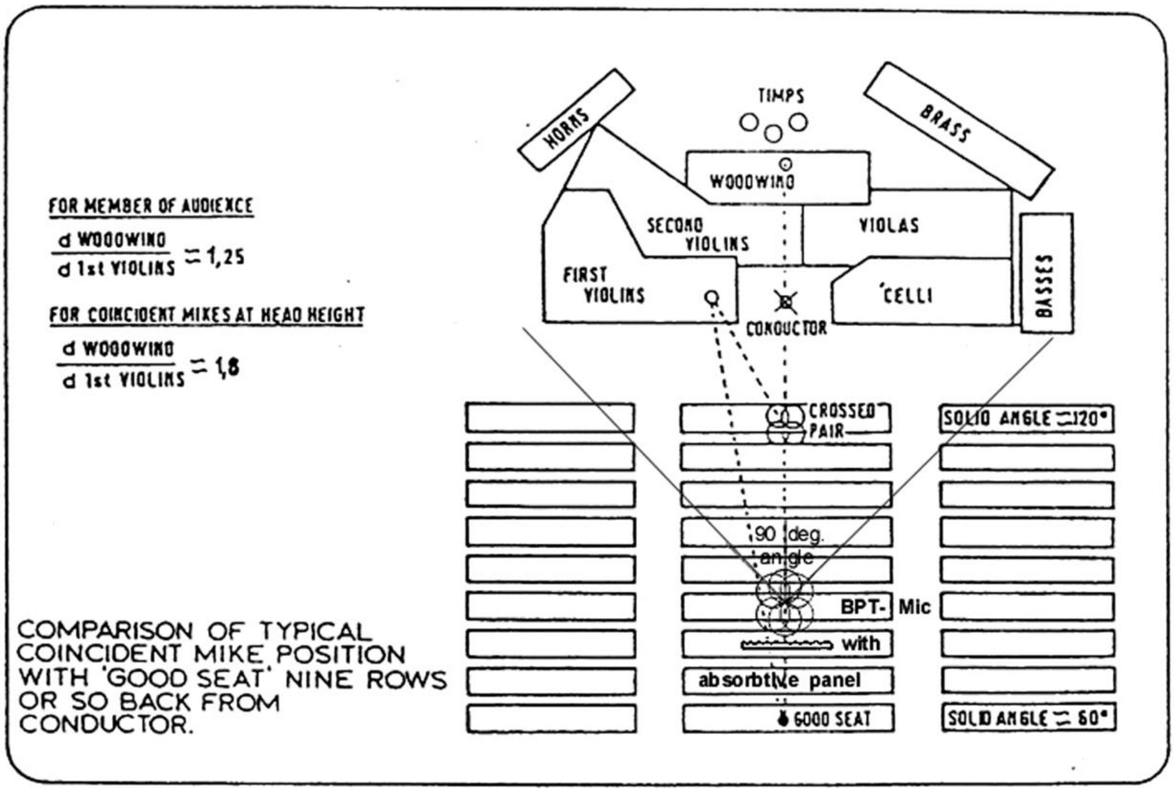
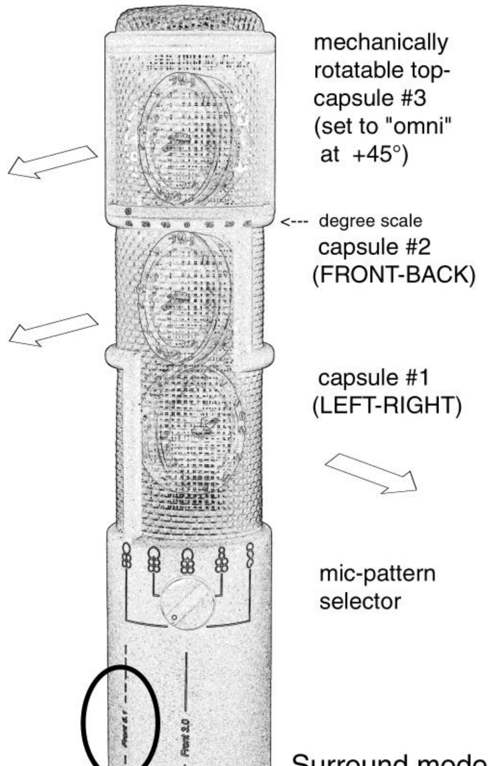


Fig. 4: Microphone position vs. listener position in a concert hall [adapted from Faulkner, 1981]

4. RECORDING IN SURROUND Version 1: The BPT in 5.1 mode (see **Fig. 5** ->)



As stated by audio engineer Tomlinson Holman "… Two channel stereo can produce the sensation of looking into a space beyond the loudspeakers; multichannel stereo can produce the sensation of being there ..." [Holman, 2000]

This is a strong statement for making surround recordings and the BPT makes this quite easy, as no external processor is needed:

If you want to create a 5.1 surround signal with just one BPT-mic, the pattern selector needs to be switched to total left position. In this mode the top capsule (# 3) is in omni mode (and rotated to +45°), while capsules #1 and #2 remain in fig-8 mode. As the 5.1 signal is going to be derived by using MS-decoding, the BPT-mic as a whole needs to be rotated by 45° so that the lowest capsule (#1) becomes a figure-8, which picks up sound to the LEFT and RIGHT of the microphone, while capsule #2 picks up the FRONT and BACK sound-field.

To help visually with the right orientation of the mic-body, there is dashed line indicating the correct 0° alignment of the microphone: ----FRONT 5.1 -----

As can be seen in the scheme of figure 4, the 5.1 surround signal is then created by combining (mixing, i.e. summing) the signals of the omni-capsule (#3) with the fig-8 capsules (#1 and 2#) in an appropriate way.

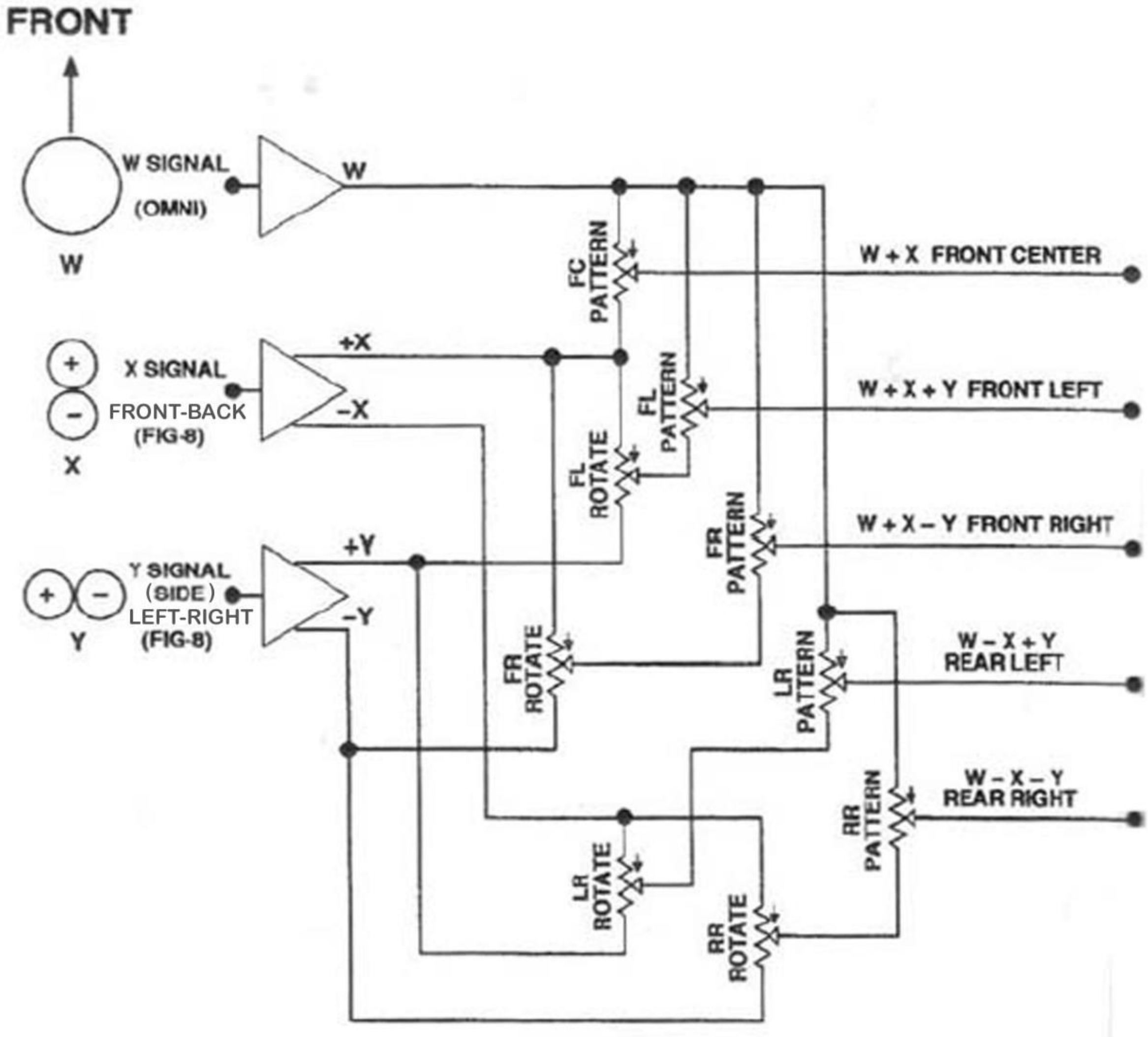


Fig. 6: signal processing schematic for 3-capsule microphone (adapted from [Eargle, 2004])

For the 5.1 "Center-channel" the signal of the omni-capsule (#3, "W") and front-back capsule (#2, "X") need to be combined: W + X
For "Front Left" all the signals W + X +Y need to be summed.
For "Front Right" the same three signals, but on Y (capsule #1) the phase needs to be reversed by 180°: W + X - Y
For "Rear Left" (also called 'Left Surround' or LS) we need to perform: W – X + Y
For "Rear Right" (also called 'Right Surround' or RS): W – X - Y
The processing as seen in fig. 4 – which might seem complicated at first – can very easily be achieved on a normal mixing desk: on digital mixing desks, which allow to assign the signal-input of a channel A/D-converter to multiple channel strips, this is particularly easy.

The graphic below shows a routing assignment on an 8-bus console which mixes the W, X and Y signals on the appropriate busses for 5.1 recording or reproduction.

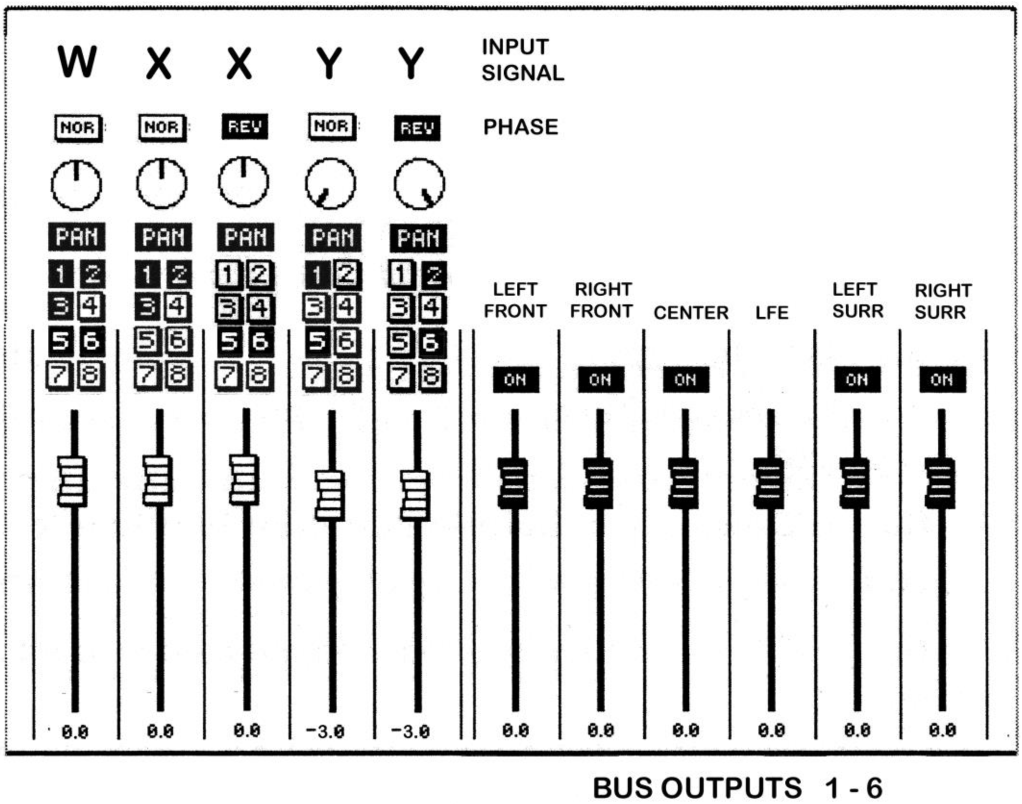


Fig. 7: BPT-microphone signal MS-decoding on an 8-bus mixing desk

The routing scheme above also makes use of the PAN function on a mixing desk, which is not really needed if the desk allows “direct routing” to single busses. In that case all input faders can be set to 0dB and the PAN-control at center.
The -3dB level attenuation on the fader needs to be done if for L/R-routing on stereo-busses the PAN-control needs to be employed, as a signal which is panned in this way has a 3dB gain in respect to signals which are routed to both bus-channels with the PAN-control set at center (due to the ‘panning law’ which is normally employed in mixing desks).

Creating an LFE-signal:
The signal-routing layout shown in fig. 7 does not provide an LFE-signal; should this be needed, the easiest way to generate one would be to bring up the W (omni) signal on a free input-channel, then low-pass filter it with a cut-off frequency set somewhere between 90 - 120Hz, route it to bus #4 and balance it in level wise, according to taste.

5. RECORDING IN SURROUND - Version 2: two BPT’s in “Back-to-back” mode (2 x 3.0)

There is a second possibility to achieve a 5.1 surround recording with the BPT microphone, which offers better channel separation (especially between front and rear signals) by use of 2 BPT microphones and (at least one) acoustic baffle and absorptive panel in-between.
In this case for both front and rear a BPT-mic in 3.0 mode is being used, which directly deliver the signals for L, Center, Right (Front) and LS, RS (Rear).
This version has the advantage that no signal-processing (MS-decoding) is needed.

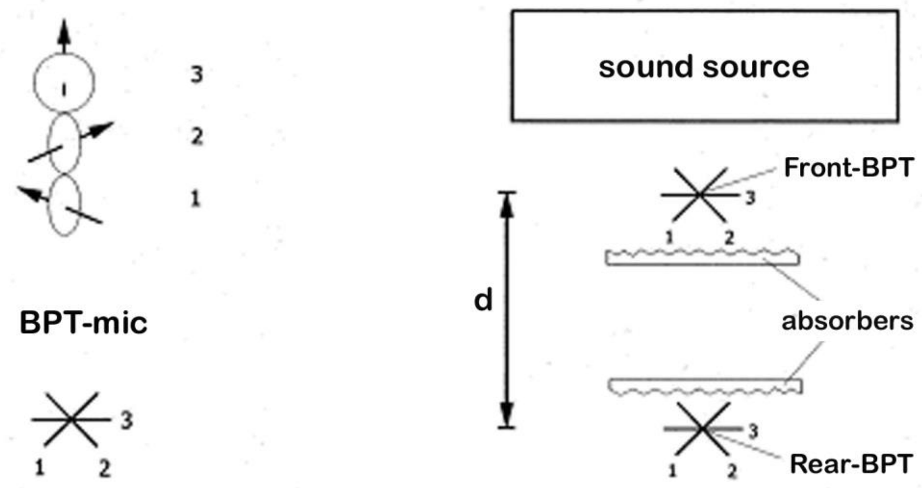


Fig. 8: Two BPT microphones in 'Back-to-back' configuration with absorber panels (for surround rec.)

The distance "**d**" between the two BPT microphones can be varied to control the amount of spatial impression contained in the recording: if **d** is in the range of 50cm to 100cm spatial impression will be less, if **d** is in the order of several meters the front and rear signals will be more de-correlated and spatial impression will be larger. However, a distance of 10m should not be exceeded between the front and rear BPT in order to keep time-of-arrival differences for sound below 30ms, as above this limit echo effects might occur, depending on the room acoustics and placement of the BPT- microphones in relation to the sound source.

6. Using the Figure-of-Eight pattern to "re-balance" the orchestra (and soloists)

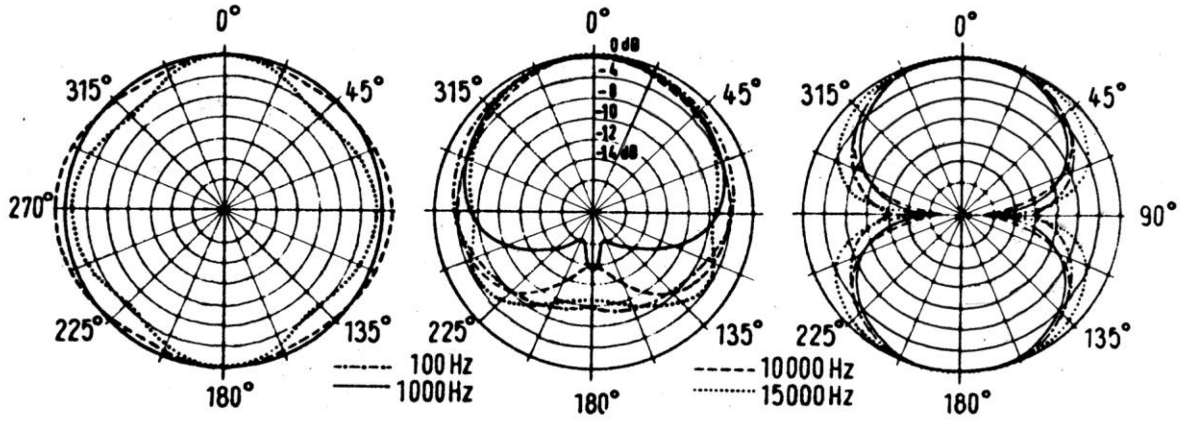


Fig. 9: Polar diagrams of a double membrane condenser mic (Neumann KM88)

As can be seen in fig. 9 off-axis level attenuation is much more pronounced with fig-8 microphone patterns, than with cardioids: while a cardioid will have only 1-2dB attenuation at 30° and approx. 4 dB at 60°, the fig-8 pattern already exhibits a 9dB level drop at 60°,

which - due to its polar pattern – increases dramatically above 60° to reach 'infinity' (total attenuation) at 90°.
This strong directional characteristic of the fig-8 pattern can be used as an advantage if one needs to re-balance the instrumental groups (including soloists) of an orchestra:

For this purpose it is recommended to point the main axis of the fig-8 towards the sound element which needs the most support level-wise. In the example of fig. 10 below, the mic is directed towards the woodwinds, but due to the much closer distance to the first desks of the sting section, these will still have an advantage of 2dB over the woodwinds.

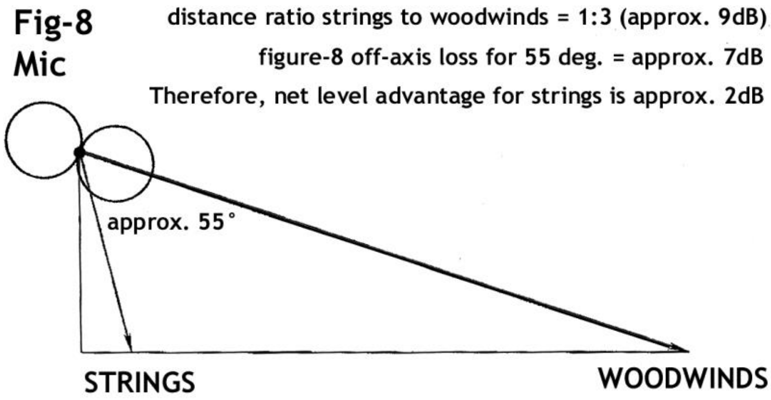


Fig. 10: vertical attenuation (angle of incidence) as a creative tool for level balancing

In a typical situation, pictured in the diagram above, the microphone could be positioned approx. 3m (9feet) above the first desks of the strings, while the woodwinds might be approx. 9m (27 feet) away. Hence a distance ratio of 1:3 applies. As can also be seen in the graphic concerning the "reverberation radius" or "critical distance" in the appendix, direct sound suffers a level drop of 6dB, when the distance is doubled (2:1). Doubling the distance again (4:1) would lead to another level drop of 6dB, so a total of 12dB attenuation; a distance ratio of 1:3 therefore results in a level drop of 9dB.
[Rem.: usually, when miking the strings from above they might tend to sound rather bright and sometimes aggressive, therefore a slight level-drop (and also high-frequency attenuation) due to the off-axis position of the strings might even be an advantage. However, we might be able to achieve a more balanced overall sound by moving the BPT main-mic further out in the hall]

If we imagine a situation of a hall with a reverberation radius D_{crit} = 6m for example, we might feel encouraged to move the main microphone (BPT) further back in the hall (with or without acoustic panel behind), so it might be slightly less than 6m away from the first strings and therefore approx. 12m away from the woodwinds.
If we decide to tilt the main axis of the fig-8 down a bit so that it is directed towards the middle between the woodwinds and the first desks of the strings, we would get the same amount of off-axis level attenuation for both of them, while the level drop due to distance would result in an advantage of 6dB for the strings in comparison to the woodwinds.
The further we move the BPT away from a large sound source like an orchestra, the more rounded or 'naturally balanced' the sonic picture becomes usually and the more effectively we can use the right orientation of the main-axis to feature sound elements within the

sound source, who need that kind of support. In the absence of spot-microphones, soloists in front of the orchestra next to the conductor, or a harp further back in the orchestra might be the usual candidates which need such help ...

To give an example of the effectiveness of diffuse sound attenuation due to the use of an acoustic panel behind the BPT microphone: in the concert hall of the Salzburg Festival (D_{crit}=5,5m) – in order not to obstruct sight-lines from the balcony - the BPT mic with panel is 12m (!) away from the conductor's podium. Since the curved absorptive panel shades off more or less the rear half-hemisphere of diffuse sound, the critical distance is almost doubled, so even with such a large distance to the orchestra, the direct-/diffuse-sound ratio is still a "healthy" one and provides nice-sounding results for high-quality documentary recordings ...

7. Flat absorptive panel vs. curved panel

As already mentioned above, in many recording situations the final result will benefit from the use of an absorptive acoustic panel behind the BPT-microphone in order to re-balance the direct- to diffuse-sound ratio (Rem.: … for use of the BPT in '3.0 mode'. If the BPT is used in the '5.1 mode', of course the rear side should not be covered or shielded off potential sound sources or diffuse sound).

A curved acoustic panel will usually be more effective, as with the same surface it manages to 'protect' a larger angle of the microphone from picking up unwanted sound. On the other hand, some focusing effects may be noticed with a curved (rounded) acoustic panel, if the absorptive materials are not fully absorptive over the whole frequency range (which is hard to achieve sometimes if its mechanically supportive elements are made of metal).

In practical experiments it has been found that curved absorptive panels, with diameters useful in connection with the BPT microphone (diameter approx. 40cm), may produce unwanted reflections (and therefore sound coloration due to comb-filtering effects) in the frequency band roughly between 500 – 700 Hz. In this case a band-filter tuned to about 600Hz with Q=3 to 5 and an attenuation of -4 to -8dB (inserted on at least the L and R BPT-mic signal) will help to eliminate unwanted sound coloration artifacts.

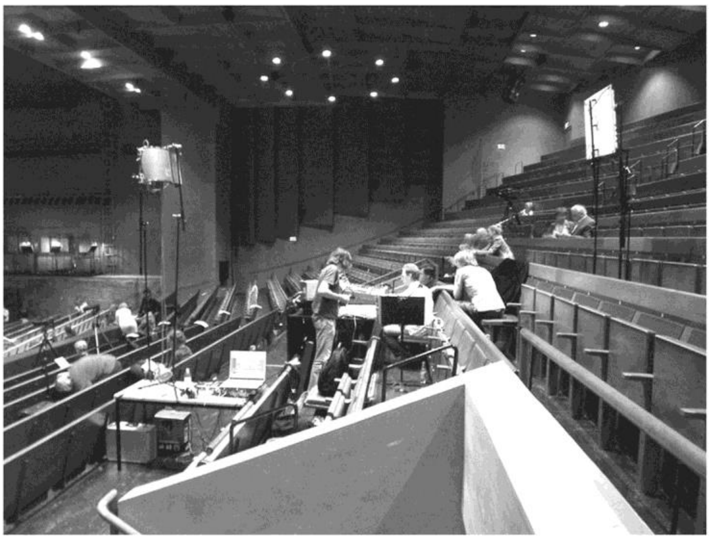


Fig. 11 - (left): BPT with rounded panel; **(right):** BPT surround-recording with 2 BPT-mics in "back-to-back" mode in auditorium (approx. 5m spacing) with rounded panel (front) and flat acoustic panel (back)

APPENDIX:

Definition of "critical distance" D_{crit} :

D_{crit} [m] = 0.057 sqrt (V [m^3] / RT60 [s])

or

D_{crit} [ft] = 0.03147 sqrt (V [ft^3] / RT60 [s])

with 1ft = 0.3048m or 1m = 3.2808 feet

D_{crit} …. "Critical distance" (i.e. reverberation radius)

sqrt …. square root

V …. Volume of the room

RT60 … reverb time of the room in seconds

The "critical distance" (or reverberation radius) from the sound source is reached, when diffuse sound reaches the same level (volume) as the direct sound.

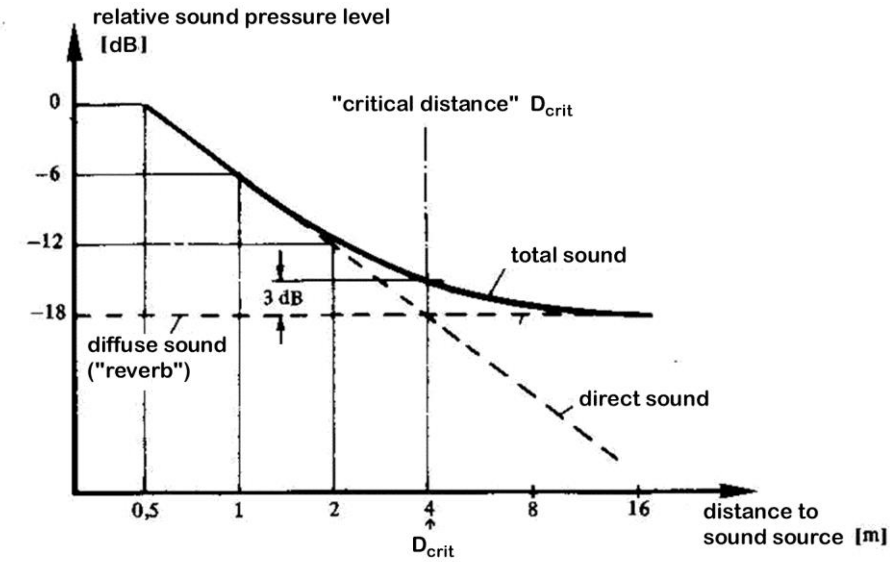


Fig. 12: "Critical Distance" (D_{crit}): Relation between SPL and distance to sound source for direct- and diffuse-sound in a room

Hall	**Volume** [m3]	Volume [ft3]	**Seats**	**RevTime** [s]	**D_crit** [m]	D_crit [ft]
Ottobeuren (church, Germany)	130000	4590300	n.a.	7,00	7,77	25,48
Royal Albert Hall, London	86650	3059612	5080	2,40	10,83	35,53
Amsterdam Concertgebouw	18700	660297	2206	2,00	5,51	18,08
Boston Symphony Hall	18740	661709	2631	1,80	5,82	19,08
Vienna Musikverein	14600	515526	1680	2,05	4,81	15,78
Teatro della Scala (opera, Italy)	11252	397308	2289	1,24	5,43	17,81
Eisenstadt Castle - concert-hall	6800	240108	n.a.	1,70	3,60	11,83
King's Theatre, London	4550	160661	n.a.	1,55	3,09	10,13
Brahmssaal, Vienna Musikverein	3390	119701	604	1,63	2,60	8,53
Esterháza Castle - concert hall	1530	54024	n.a.	1,20	2,04	6,68
control room (typ. rec. studio)	118	4167	n.a.	0,24	1,26	4,15

Table 1: Reverb time and resulting critical distance for performance spaces of different size

High Pass Filter Control
on rear side of BPT-Microphone:

On the rear side of the BPT a filter-selector can be found:
A High-Pass Filter with corner frequency at 90Hz can be switched in for all three capsules, if needed.
Switch in left position, all three capsules linear.
Switch in center position: HPF for all 3 capsules
Switch in right position: only top capsule #3 has HPF activated

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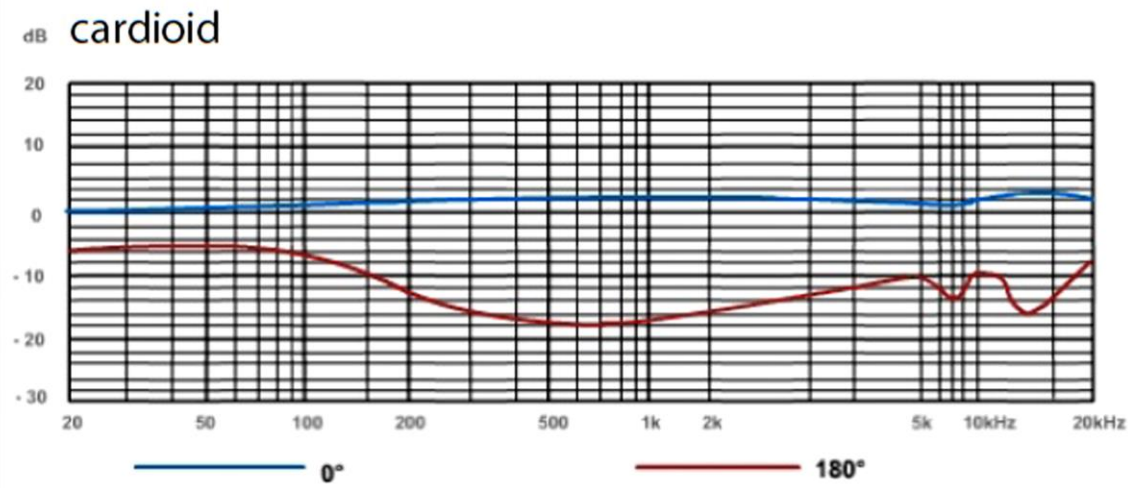
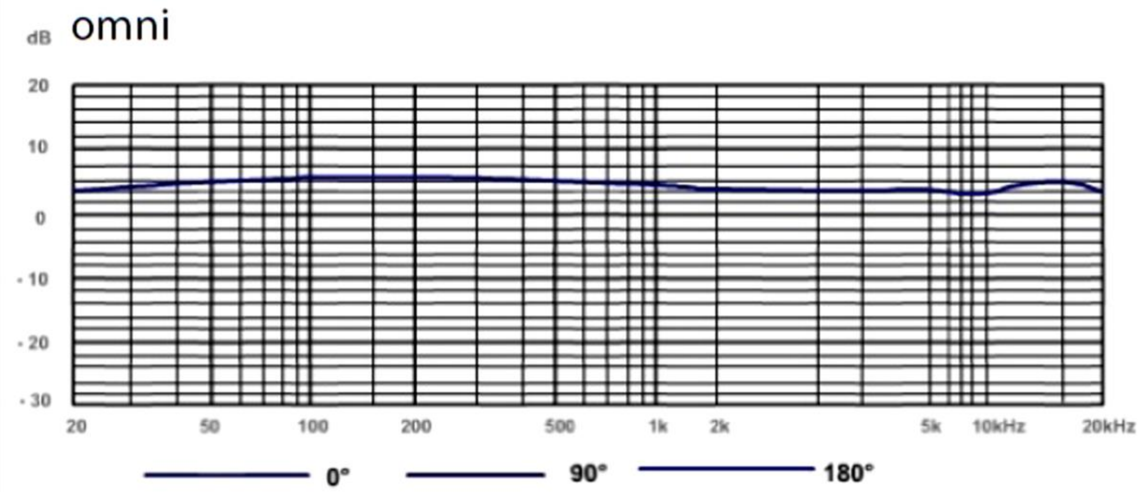
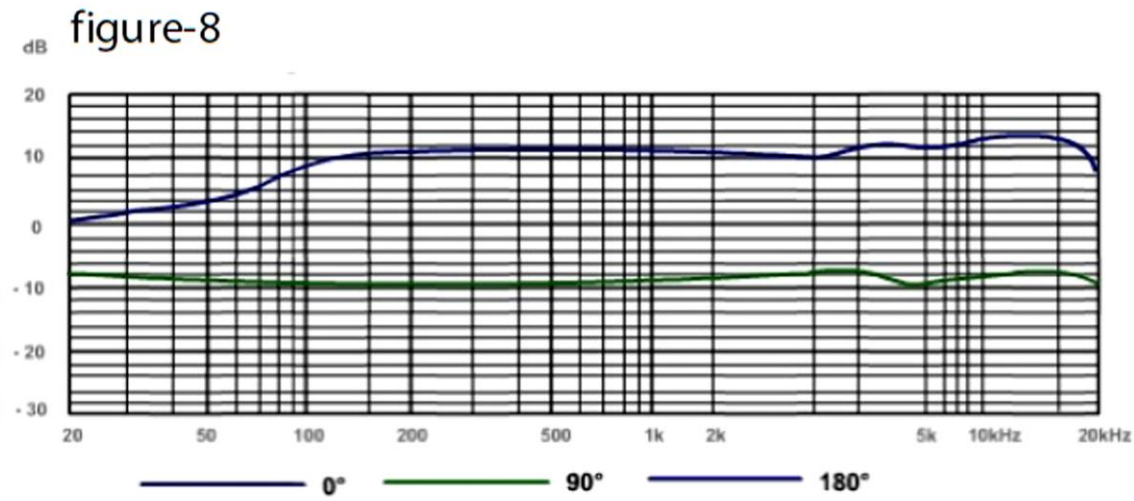
NEVATON "BPT"-Microphone Technical data:

- Frequency Range: 20-20000Hz
- Free Field Sensitivity at 1000Hz and 1000 Ω: 20 +/-2 mV / Pa (or better)
- Equivalent SPL DIN/IEC 651: 4 dB(A) for cardioid, omni and 7dB(A) for fig-8
- Max. Sound Pressure Level [for THD 0,5%]: 142dB
- Dynamic range: 137 dB (135 dB for fig-8)
- Nominal output impedance: < 50 Ω
- Recommended Load Impedance: >= 1000 Ω
- Capsule diameter: 33 mm
- Microphone body diameter: 48 mm
- Length: 245 mm
- Weight: 820 g
- Phantom power (48V +/- 4V)
- Current consumption: 7 mA per capsule
- Relative Humidity: operating range up to 80% (at +20°C)
- Connector Type: XLR, 7-pin (male connector built into microphone)
- Pin assignment: 1= GND, pin 6+7 = signal „Left Capsule (#1)“ +/-, pin 4+5 = signal „Right capsule (#2)“ +/-, pin 2+3 = signal „Center capsule (#3)“ +/-

NEVATON "BPT"-Microphone Frequency responses:







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