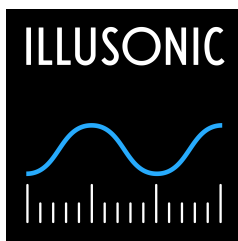


## **ILLUSONIC traitements**

***Modèle paramétrique au B-Format***

***Plugin Ambisonique :***  
**A/B – Format Decoder**

<https://www.illusonic.com/fr/landing-pages/test-balloon-plug-ins/>



# AES Dublin 2019 Paper Session P14

**P14-2 Décodage au format B basé sur la formation de faisceaux adaptatifs** - *Alexis Favrot*, Illusonic GmbH - Uster, Suisse; *Christof Faller*, Illusonic GmbH - Uster, Zurich, Suisse; EPFL - Lausanne, Suisse

Les signaux au format B peuvent être décodés en signaux avec une directivité de premier ordre. Pour le décodage stéréo et multicanal, il serait souhaitable d'avoir plus de séparation des canaux que ce qui est réalisable au premier ordre. DirAC (codage audio directionnel) et HARPEX (expansion des ondes planes à haute résolution) permettent une séparation des canaux plus élevée en utilisant un modèle paramétrique au format B pour estimer les ondes planes et le son diffus, et les rendre de manière adaptative. Une limitation est que les modèles à ondes planes et diffuses sont trop simples pour représenter des signaux complexes au format B. Nous proposons un décodeur au format B, où chaque canal est généré par un formateur de faisceau adaptatif indépendant au format B. Chaque faisceau est généré indépendamment des autres faisceaux, contournant la limitation lors de l'utilisation d'un modèle de signal au format B unique.

[https://www.lesonbinaural.fr/EDIT/DOCS/favrot\\_faffer.PDF](https://www.lesonbinaural.fr/EDIT/DOCS/favrot_faffer.PDF)



## Audio Engineering Society Convention Paper

Presented at the 146<sup>th</sup> Convention  
2019 March 20 – 23, Dublin, Ireland

This convention paper was selected based on a submitted abstract and 750-word precis that have been peer reviewed by at least two qualified anonymous reviewers. The complete manuscript was not peer reviewed. This convention paper has been reproduced from the author's advance manuscript without editing, corrections, or consideration by the Review Board. The AES takes no responsibility for the contents. This paper is available in the AES E-Library (<http://www.aes.org/e-lib>), all rights reserved. Reproduction of this paper, or any portion thereof, is not permitted without direct permission from the Journal of the Audio Engineering Society.

### B-Format Decoding Based on Adaptive Beamforming

Alexis Favrot<sup>1</sup> and Christof Faller<sup>1</sup>

<sup>1</sup>*Illusonic GmbH, Bahnstrasse 23, 8610 Uster, Switzerland*

Correspondence should be addressed to Alexis Favrot ([alexis.favrot@illusonic.com](mailto:alexis.favrot@illusonic.com))

#### ABSTRACT

B-Format signals can be decoded into signals with first order directivity. For stereo and multi-channel decoding, it would be desirable to have more channel separation than what is achievable by first order. DirAC (directional audio coding) and HARPEX (high resolution plane wave expansion) achieve higher channel separation by means of using a parametric B-Format model to estimate plane waves and diffuse sound, and adaptively rendering those. A limitation is that plane wave and diffuse models are too simple to represent complex B-Format signals. We propose a B-Format decoder, where each channel is generated by an adaptive B-Format beamformer. Each beam is generated independently of the other beams, circumventing the limitation when using a single B-Format signal model.



## Audio Engineering Society Convention Paper

Presented at the 144<sup>th</sup> Convention  
2018 May 23 – 26, Milan, Italy

This convention paper was selected based on a submitted abstract and 750-word precis that have been peer reviewed by at least two qualified anonymous reviewers. The complete manuscript was not peer reviewed. This convention paper has been reproduced from the author's advance manuscript without editing, corrections, or consideration by the Review Board. The AES takes no responsibility for the contents. This paper is available in the AES E-Library (<http://www.aes.org/e-lib>), all rights reserved. Reproduction of this paper, or any portion thereof, is not permitted without direct permission from the Journal of the Audio Engineering Society.

### Adaptive Non-Coincidence Correction for A to B-Format Conversion

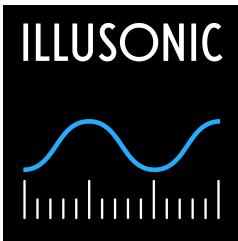
Alexis Favrot<sup>1</sup> and Christof Faller<sup>1</sup>

<sup>1</sup>*Illusonic GmbH, Bahnstrasse 23, 8610 Uster, Switzerland*

Correspondence should be addressed to Alexis Favrot ([alexis.favrot@illusonic.com](mailto:alexis.favrot@illusonic.com))

#### ABSTRACT

B-Format is usually obtained from A-format signals, i.e. from four directive microphone capsules pointing in different directions. Ideally, these capsules should be coincident, but due to design constraints, small distances always remain between them. The resulting phase mismatches between the microphone capsule signals lead to inaccuracies and interferences, impairing B-format directional responses, especially at high frequencies. An adaptive non-coincidence correction is proposed based on adaptive phase matching of the four microphone A-format signals before conversion to B-format, improving the directional responses at high frequencies, enabling higher focus, better spatial image and timbre in B-format decoded signals.



# A/B – Format Decoder v5.1.0

*A-Format* vers *9.1+6H* (ITU/SMPTE)

## ILLUSONIC

### A/B-Format Decoder

#### Decoding

Rotation: 0°    Elevation: 0°

**FOCUS**

Center: 50%    Front: 50%    Wide: 50%    Surround: 50%

Rear: 50%    Front Height: 50%    Surround Height: 50%    Rear Height: 50%

**ANGLE**

Azimuth: 45°    Front: 60°    Wide: 135°    Surround: 135°

Rear: 150°    Front Height: 45°    Surround Height: 135°    Rear Height: 150°

Diffuse gain: -4 dB    De-correlation:     Room size: 50

#### W Signal Bass

Cross-over

Gain: 0 dB    Frequency: 50 Hz

Order: Linkwitz-Riley 2nd

Invert bass

#### Outputs

**GAIN**

Center: 0 dB    Front: 0 dB    Wide: 0 dB    Surround: 0 dB

Rear: 0 dB    Front Height: 0 dB    Surround Height: 0 dB    Rear Height: 0 dB

Delay / Shelving: Surround    Delay: 20 ms    Frequency: 6 kHz    Gain: -3 dB

#### Formats

Input format: A-Format

Microphone distance: 24 mm (NT-SF1)    Microphone position: normal

Output format: 9.1 + 6H (ITU/SMPTE)

Binaural output

#### Channel ordering

Input: LF RF LB RB     Output channel test

Output: L R C LFE Lw Rw Lss Rss Lsr Rsr Lts Rts Ltr Rtr

© Copyright Illusonic GmbH, Greifensee, Switzerland, 2024. All rights reserved. v5.1.0 – Software expiration: Dec 31, 2025

- **Rotation** : ajuste la rotation de l'ensemble de la configuration du microphone virtuel de  $-180^\circ$  à  $+180^\circ$  . Le canal central indiquera exactement cet endroit. Ceci peut être utilisé pour compenser le placement optimal du microphone ou la conception du son.
  - **Élévation** : ajuste l'élévation de l'ensemble de la configuration du microphone virtuel de  $-90^\circ$  à  $+90^\circ$  . Le canal central indiquera exactement cet endroit. Ceci peut être utilisé pour compenser le placement optimal du microphone ou la conception du son.
  - **FOCUS** : Ajustez le diagramme polaire des canaux entre 0% et 100%.
- Beamforming** : Enhanced Pattern pour une plus grande directivité et séparation des canaux.
- **ANGLE** : angle d'ouverture entre la paire stéréo L / R, Ls / Rs en azimuth et en élévation...
  - **Diffuse gain** : ajuste la quantité d'énergie sonore diffuse contenue dans l'enregistrement de -18 dB à +6 dB. Le gain d'ambiance affecte directement les signaux d'entrée et modifie donc tous les canaux simultanément. Ce curseur de contrôle peut être utilisé pour ajouter plus de son de salle à un enregistrement sans ajouter de réverbération artificielle. Dans cet algorithme, le son diffus contenu dans les signaux d'entrée est extrait puis augmenté ou diminué.
  - **Décorrélation** : Vous pouvez activer / désactiver en appuyant sur le rond gris. La fonction de décorrélation affecte directement les signaux d'entrée et modifie donc simultanément tous les canaux. Cette commande peut être utilisée pour améliorer l'enveloppement d'un enregistrement. Engage **Room size**...
  - **GAIN** : vous pouvez régler le gain de sortie pour le canal C et les différents couples. Le niveau est également représenté dans le graphique polaire.
  - **Surround Delay**: retarde le signal des paires stéréo surround choisies de **off** à **100 ms**. Ceci peut être utilisé pour optimiser la localisation avant / arrière et augmente la zone d'écoute.
  - **Surround High Cut**: c'est un filtre coupe haut de premier ordre. Vous pouvez régler la fréquence de coupure de 2 kHz à 20 kHz avec la quantité de **gain** appropriée. Le filtre est appliqué aux signaux surround pour simuler une perte de haute fréquence pour les paires stéréo surround choisies, due à la dissipation de l'air.
  - **Indicateurs de niveau d'entrée**: Il reflètent le niveau des signaux d'entrée..
  - **Indicateurs de niveau de sortie**: les indicateurs de niveau de sortie indiquent le niveau des signaux décodés que le plug-in transmet au DAW.

# A/B – Format Decoder v5.1.0

- Coincident
  - 10 mm
  - 12 mm
  - 13 mm
  - 14 mm (Ambeo)
  - 15 mm
  - 16 mm
  - 17 mm
  - 18 mm
  - 19 mm
  - 20 mm
  - 21 mm (SPS 200)
  - 22 mm
  - 23 mm
  - 24 mm (NT-SF1)
  - 25 mm
  - 26 mm
  - 27 mm
  - 28 mm
  - 29 mm
  - 30 mm
- 24 mm (NT-SF1) ▾

- A.1-Format ▾
- A-Format
  - B-Format FuMa
  - B-Format AmbiX
  - Schoeps Triple MS
  - Dolby Triple MS
  - A.1-Format
  - Schoeps Double MS
  - B.1-Format FuMa
  - B.1-Format AmbiX

- Microphone position ▾
- normal ▾
- normal
  - end-fire
  - invert

Formats

Input format A.1-Format ▾

Microphone distance 24 mm (NT-SF1) ▾

Microphone position normal ▾

Output format 5.1 + 4H (ITU/SMPTE) ▾

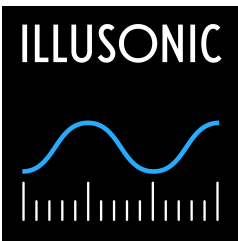
Binaural output

Channel ordering

Input LF RF LB RB Omni  Output channel test

Output L R C LFE Ls Rs Lts Rts Ltr Rtr

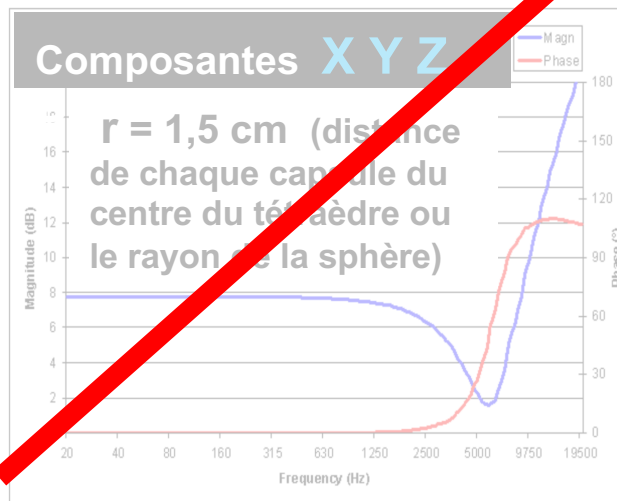
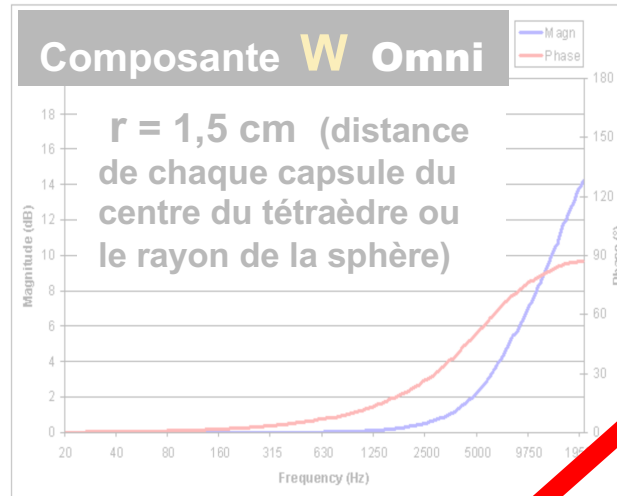
- Mono
  - Stereo
  - Quad (ITU/SMPTE)
  - 5.1 (ITU/SMPTE)
  - 7.1 (ITU/SMPTE)
  - 4+4 (ITU/SMPTE)
  - 5.1 + 2H (ITU/SMPTE)
  - 7.1 + 2H (ITU/SMPTE)
  - 5.1 + 4H (ITU/SMPTE)
  - 7.1 + 4H (ITU/SMPTE)
  - Quad (Film)
  - 5.1 (Film)
  - 7.1 (Film)
  - 4+4 (Film)
  - 5.1 + 2H (Film)
  - 7.1 + 2H (Film)
  - 5.1 + 4H (Film)
  - 7.1 + 4H (Film)
  - Cube
  - Cube + Center
  - Cube + Center + Side
  - Cube + Center + Side + Back
  - B-Format FuMa
  - B-Format AmbiX
  - A-Format
  - 7.1 (ITU/SMPTE alt)
  - 7.1 + 2H (ITU/SMPTE alt)
  - 7.1 + 4H (ITU/SMPTE alt)
  - 9.1 (ITU/SMPTE)
  - 9.1 (Film)
  - 9.1 + 2H (ITU/SMPTE)
  - 9.1 + 2H (Film)
  - 9.1 + 4H (ITU/SMPTE)
  - 9.1 + 4H (Film)
  - 9.1 + 6H (ITU/SMPTE)
  - 9.1 + 6H (Film)
- 5.1 + 4H (ITU/SMPTE) ▾



Microphone Distance

- Coincident
  - 10 mm
  - 12 mm
  - 13 mm
  - 14 mm (Ambeo)
  - 15 mm
  - 16 mm
  - 17 mm
  - 18 mm
  - 19 mm
  - 20 mm
  - 21 mm (SPS 200)
  - 22 mm
  - 23 mm
  - 24 mm (NT-SF1)
  - 25 mm
  - 26 mm
  - 27 mm
  - 28 mm
  - 29 mm
  - 30 mm
- 24 mm (NT-SF1) ▾

# Réponse en fréquence ( **Amplitude** et **Phase** ) de filtres théoriques *FW* pour la conversion du A-Format vers le B-Format



Pour la composante **W**

$$F_W = \frac{1 + \frac{j\omega r}{c} - \frac{1}{3} \left(\frac{\omega r}{c}\right)^2}{1 + \frac{1}{3} \left(\frac{j\omega r}{c}\right)}$$

**ILLUSONIC tire un trait sur 40 ans d'Histoire !!**

<http://pcfarina.tng.uni-r.it/Ambisonics.html>

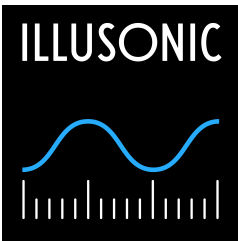
Pour les composantes **XYZ**

$$F_{XYZ} = \sqrt{6} \frac{1 + \frac{1}{3} \left(\frac{j\omega r}{c}\right) - \frac{1}{3} \left(\frac{\omega r}{c}\right)^2}{1 + \frac{1}{3} \left(\frac{j\omega r}{c}\right)}$$

$r$  = distance de chaque capsule du centre du tétraèdre en m  
 $w$  = fréquence angulaire en rad / s (  $w = 2\pi f$  )  
 $c$  = vitesse du son en m / s ( 340 m / s )

## A/B – Format Decoder v5.1.0





# A/B – Format Decoder v5.1.0

- A.1-Format ▾
  - A-Format
  - B-Format FuMa
  - B-Format AmbiX
  - Schoeps Triple MS
  - Dolby Triple MS
  - A.1-Format**
  - Schoeps Double MS
  - B.1-Format FuMa ?
  - B.1-Format AmbiX ?

Omni Microphone Bass

Cross-over

Gain: 0 dB      Frequency: 50 Hz

Order: **Passe-bas 12dB/oct**

Linkwitz-Riley 2nd ▾

Invert bass

## A-Format + Micro Omni

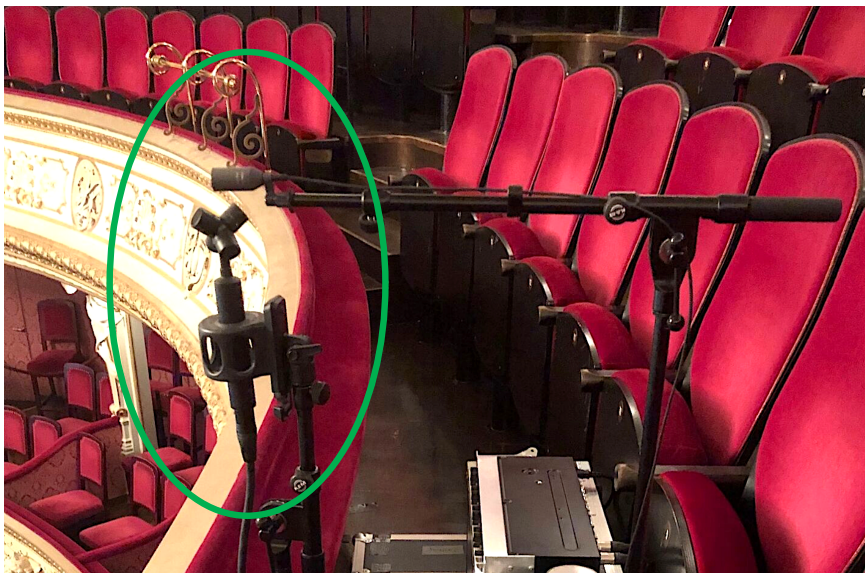
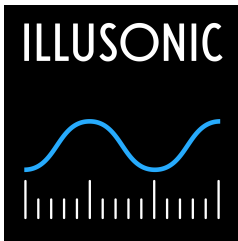


Photo Illusonic

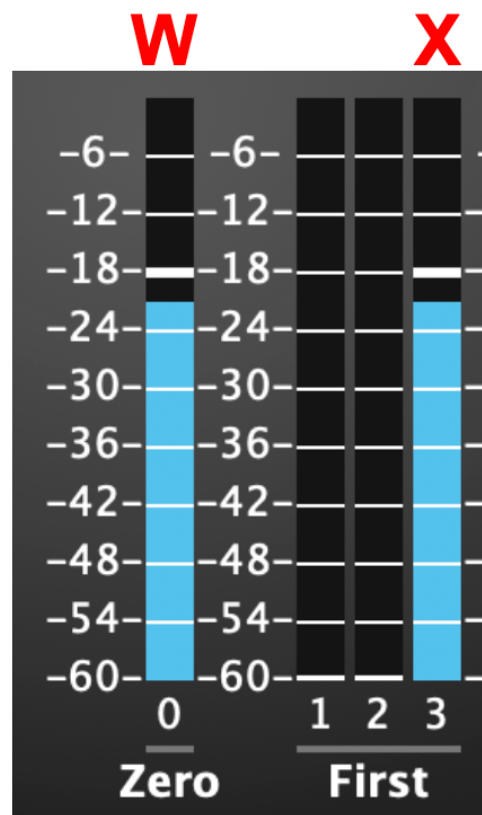
- Linkwitz-Riley 2nd ▾
  - Linkwitz-Riley 2nd**
  - Butterworth 3rd
  - Linkwitz-Riley 4th



# A/B – Format Decoder v5.1.0

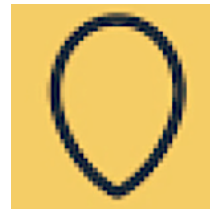
**IN : 1 KHz**

**B-FORMAT**



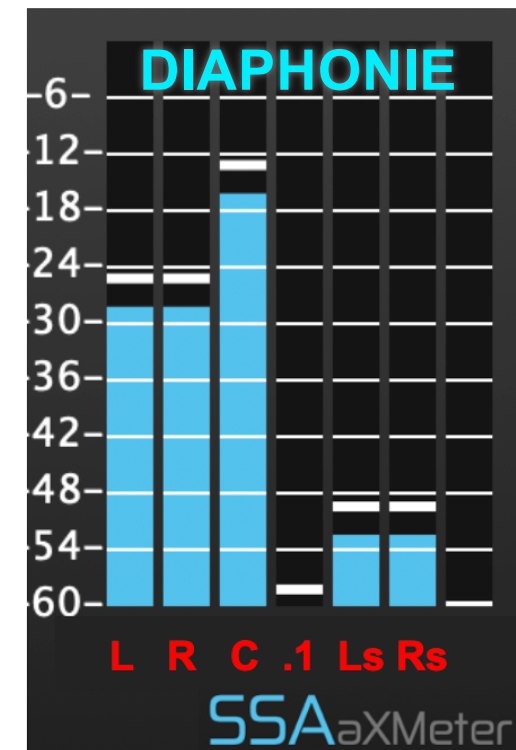
**Centre C**

**Beamforming**



**OUT : 1 KHz**

**5.1 ITU**





# A / B – Format Decoder v 5.1.0

5.1



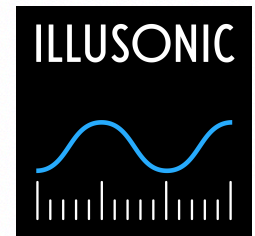
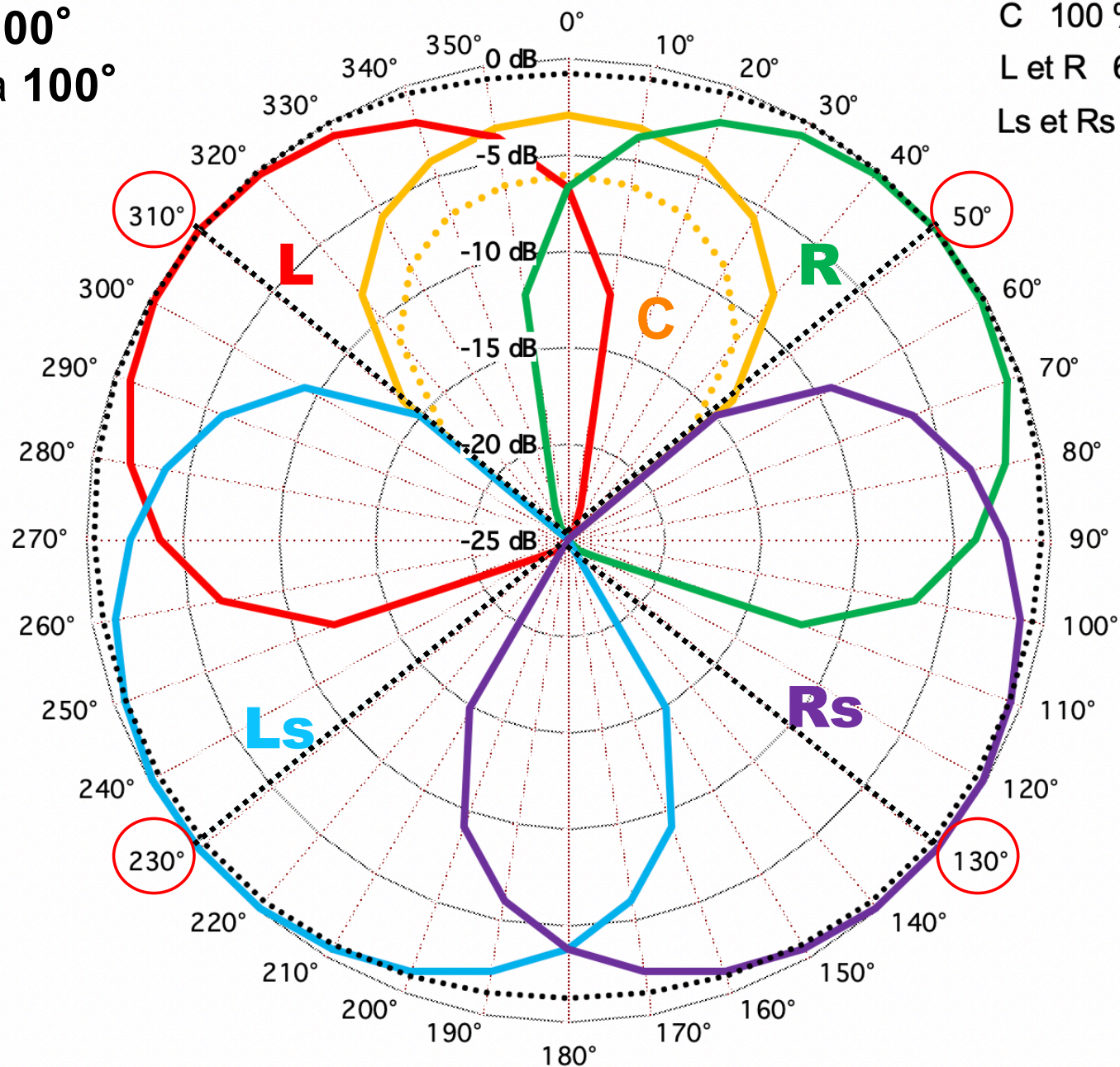
Recouvrement à -3 dB !!

C 100 % FOCUS

L et R 67% FOCUS

Ls et Rs 33 % FOCUS

L / R à 100°  
Ls / Rs à 100°



# Process Beamforming : Enhanced Pattern

1 kHz

1

A/B-Format Decoder

W Signal Bass

Gain: 0 dB, Frequency: 50 Hz

Order: Linkwitz-Riley 2nd

ILLUSONIC

Outputs

Center, Front, Wide, Surround, Rear, Front Height, Surround Height, Rear Height

Delay / Shelving, Delay, Frequency, Gain

Decoding

Rotation: 50°, Elevation: 0°

2

FOCUS

Center, Front, Wide, Surround, Rear, Front Height, Surround Height, Rear Height

50°

Formats

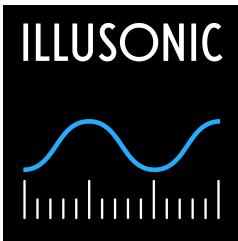
Input format: B-Format AmbiX

Microphone distance: Coincident, Microphone position: normal

Output format: Stereo

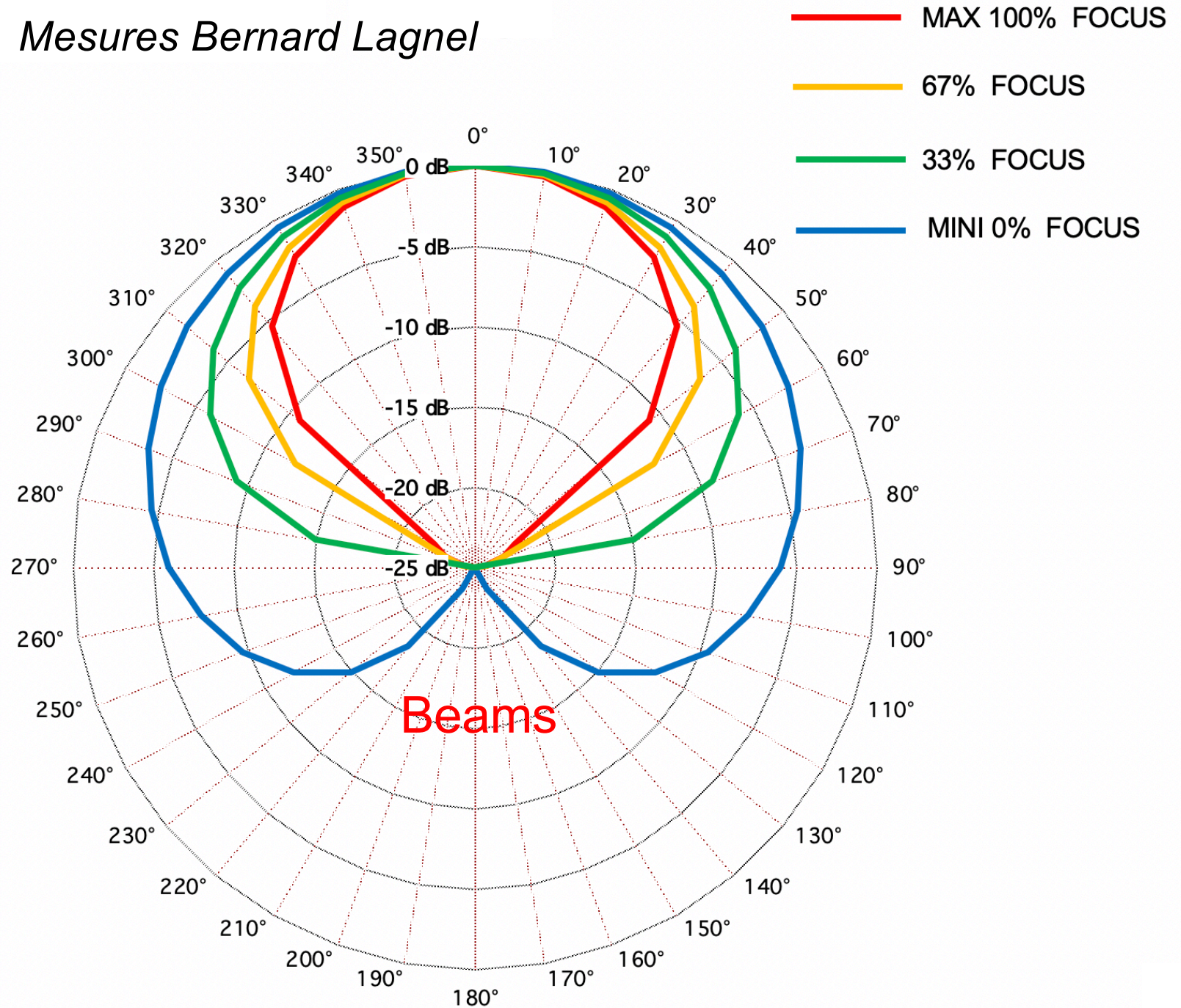
Channel ordering: Input W Y Z X, Output L R

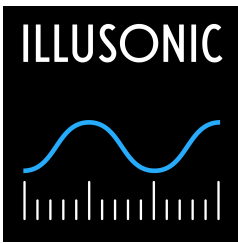
beam à	Données
0° >>	0,0 dB
30° >>	-1,4 dB
60° >>	-10,7 dB
90° >>	-27,2 dB
120° >>	-23,1 dB
150° >>	-15,2 dB
180° >>	-14,4 dB
210° >>	-15,2 dB
240° >>	-23,1 dB
270° >>	-27,2 dB
300° >>	-10,7 dB
330° >>	-1,4 dB



# Process Beamforming : Enhanced Pattern

*Mesures Bernard Lagnol*



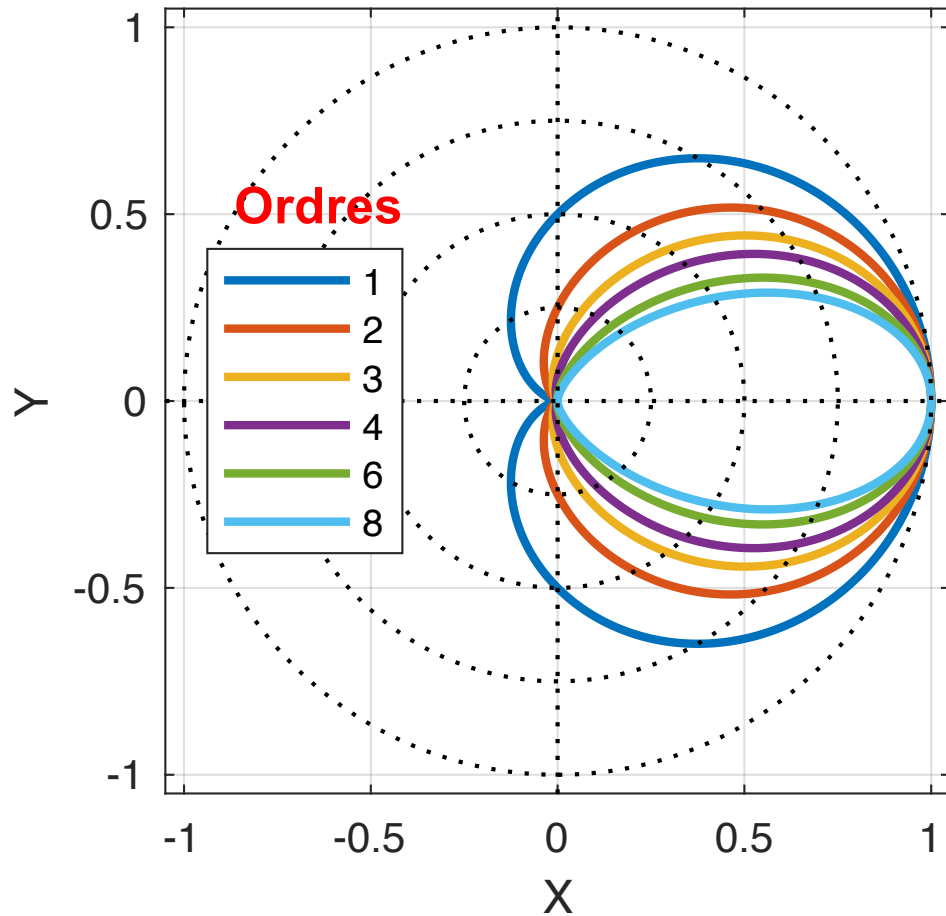


# Beamforming : Enhanced Pattern

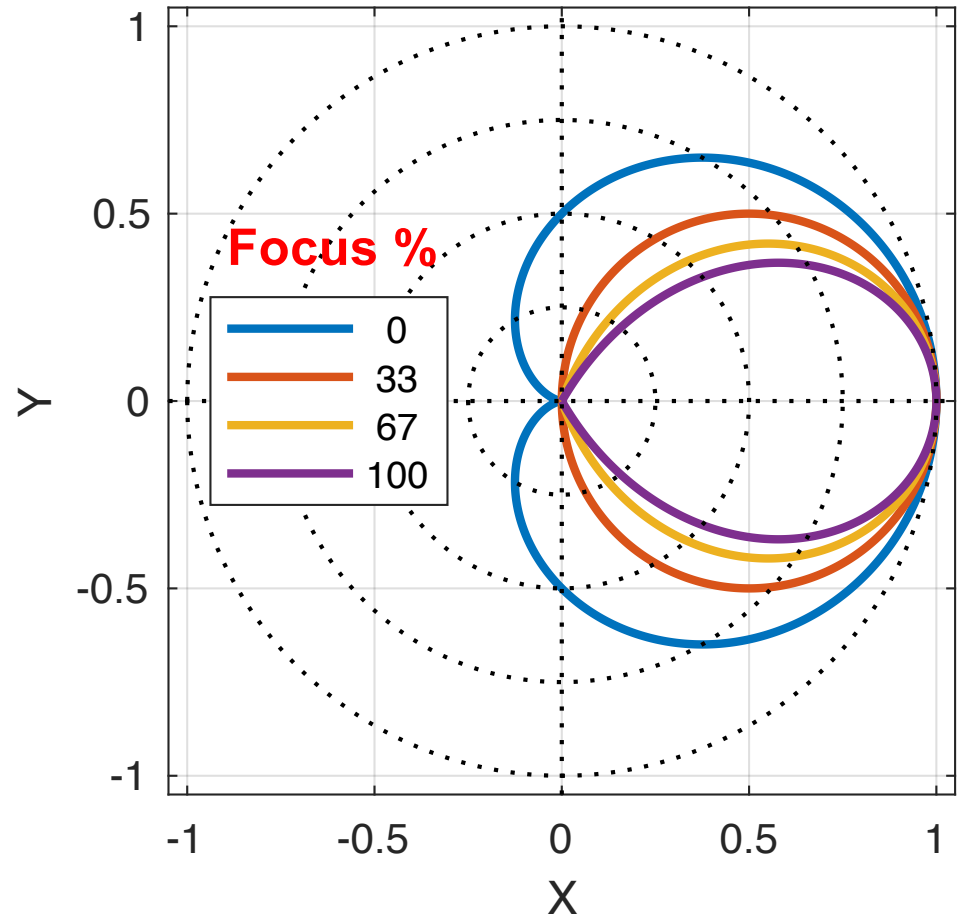
## HOA

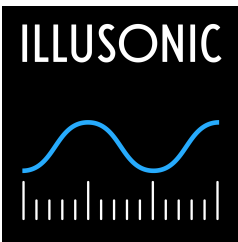
## BEAMS

Cardioid polar patterns with increasing order



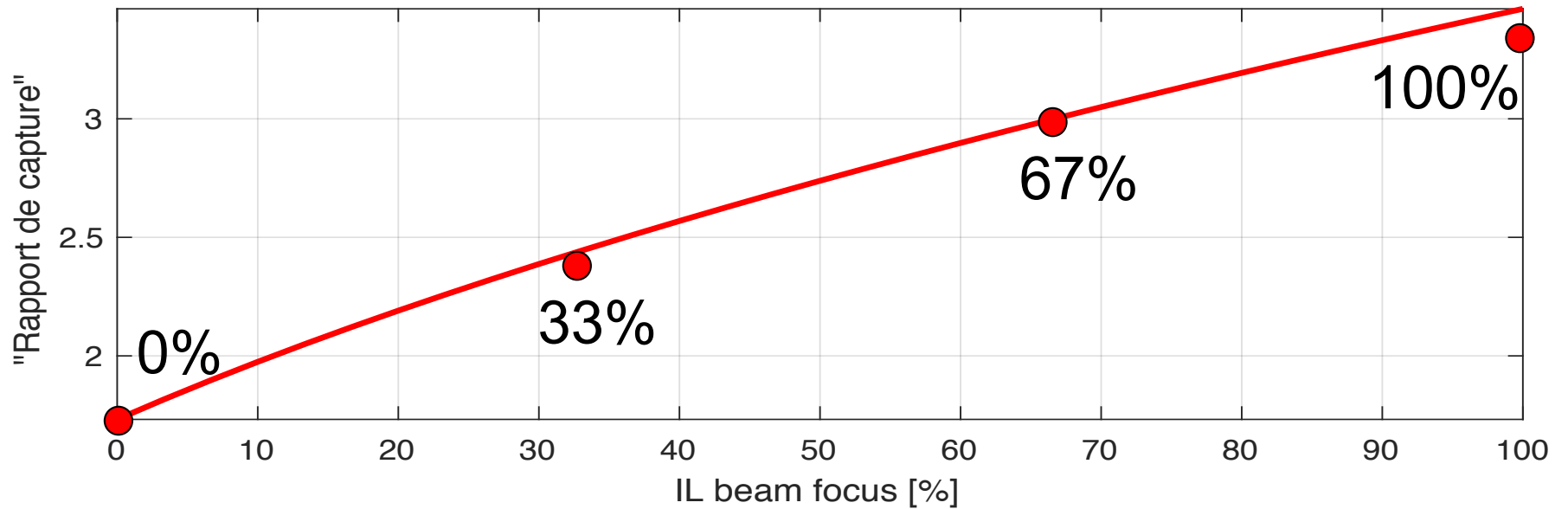
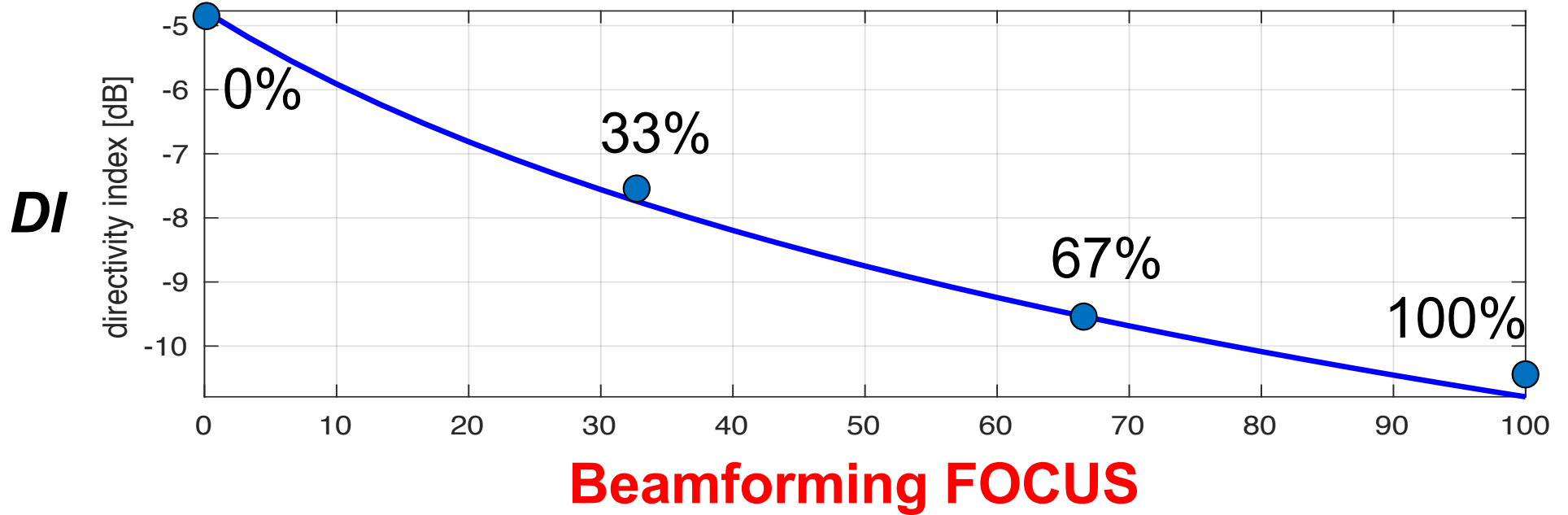
BF beam patterns with increasing focus

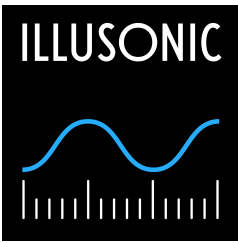




# Beamforming : Enhanced Pattern

● ● *Mesures Bernard Lagnel*



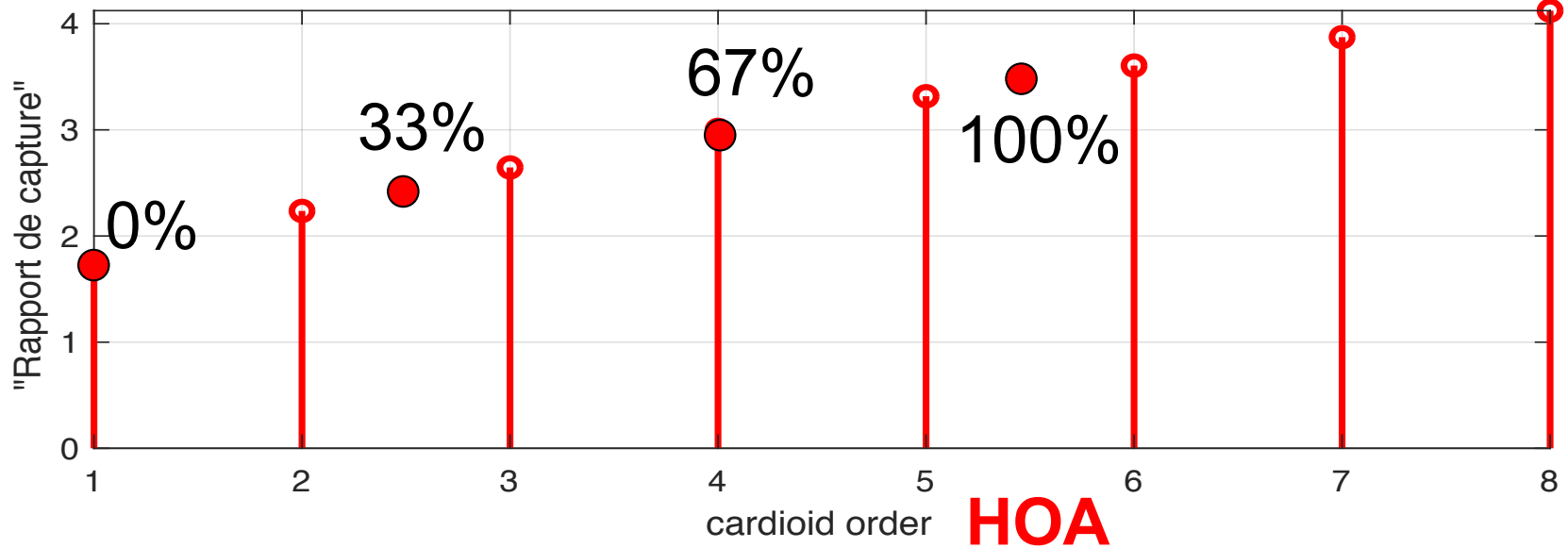
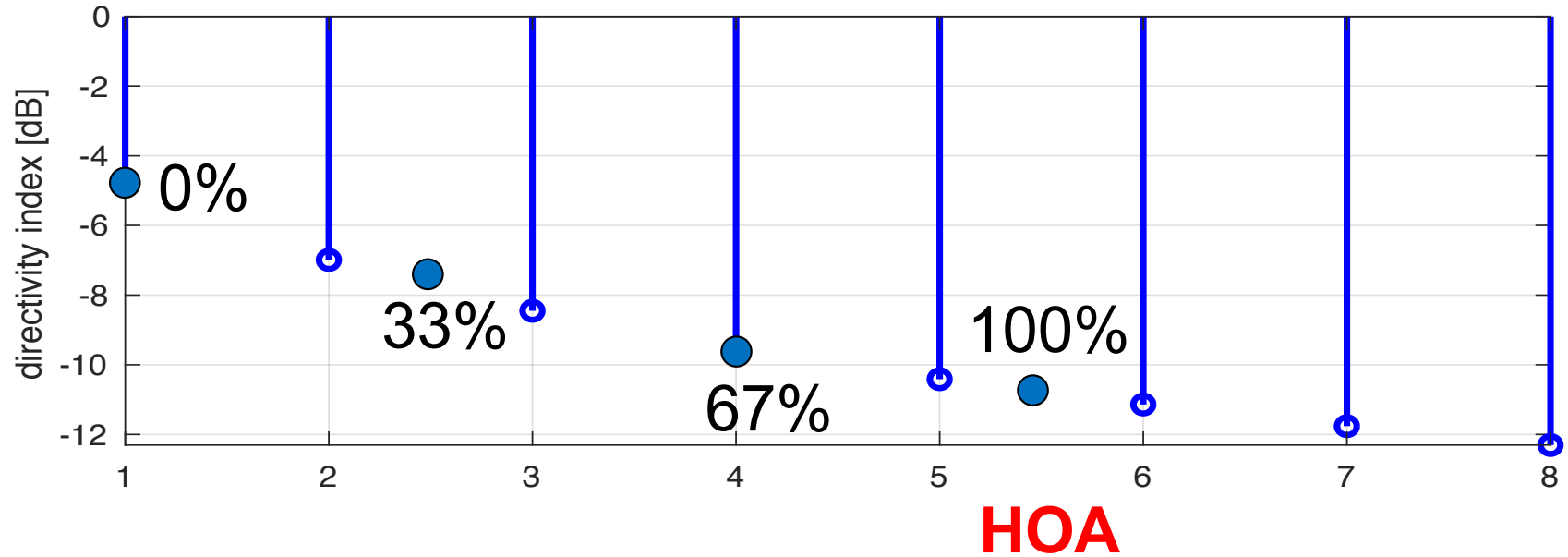


# Beamforming : Enhanced Pattern

Documents  
ILLUSONIC

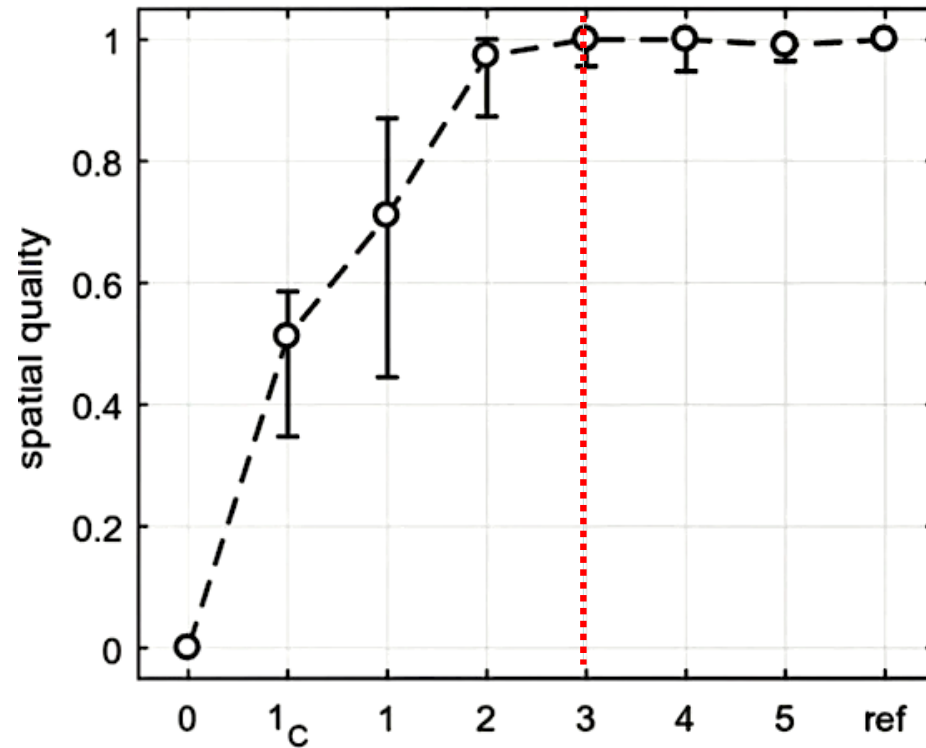
● ● *Mesures Bernard Lagnel*

**DI**



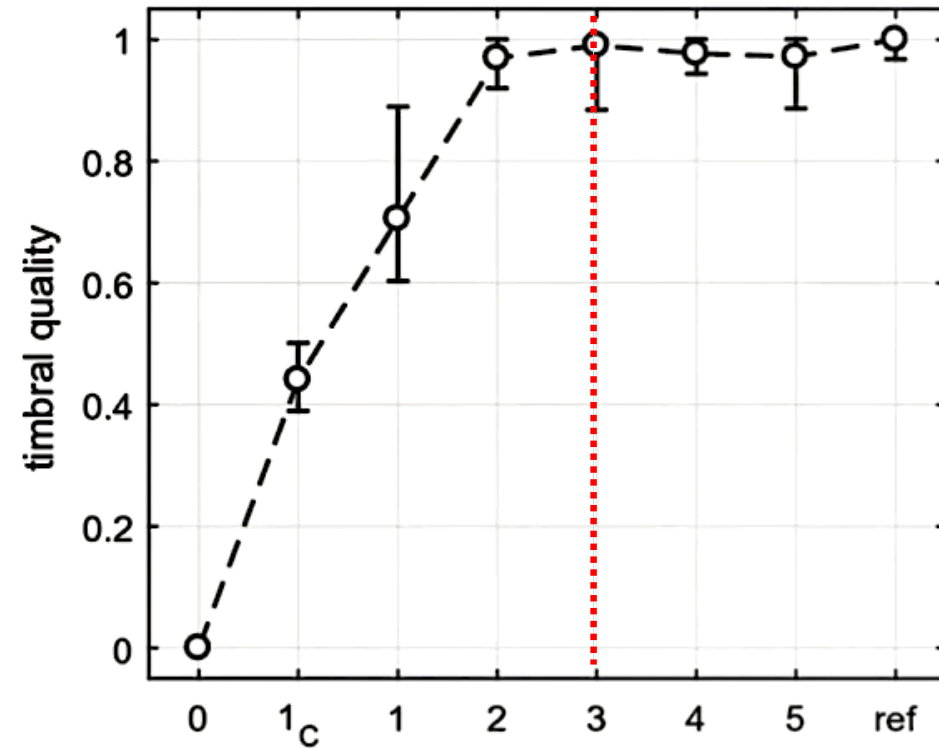
# Encoding with increasing order (headphone playback)

Ordre 3



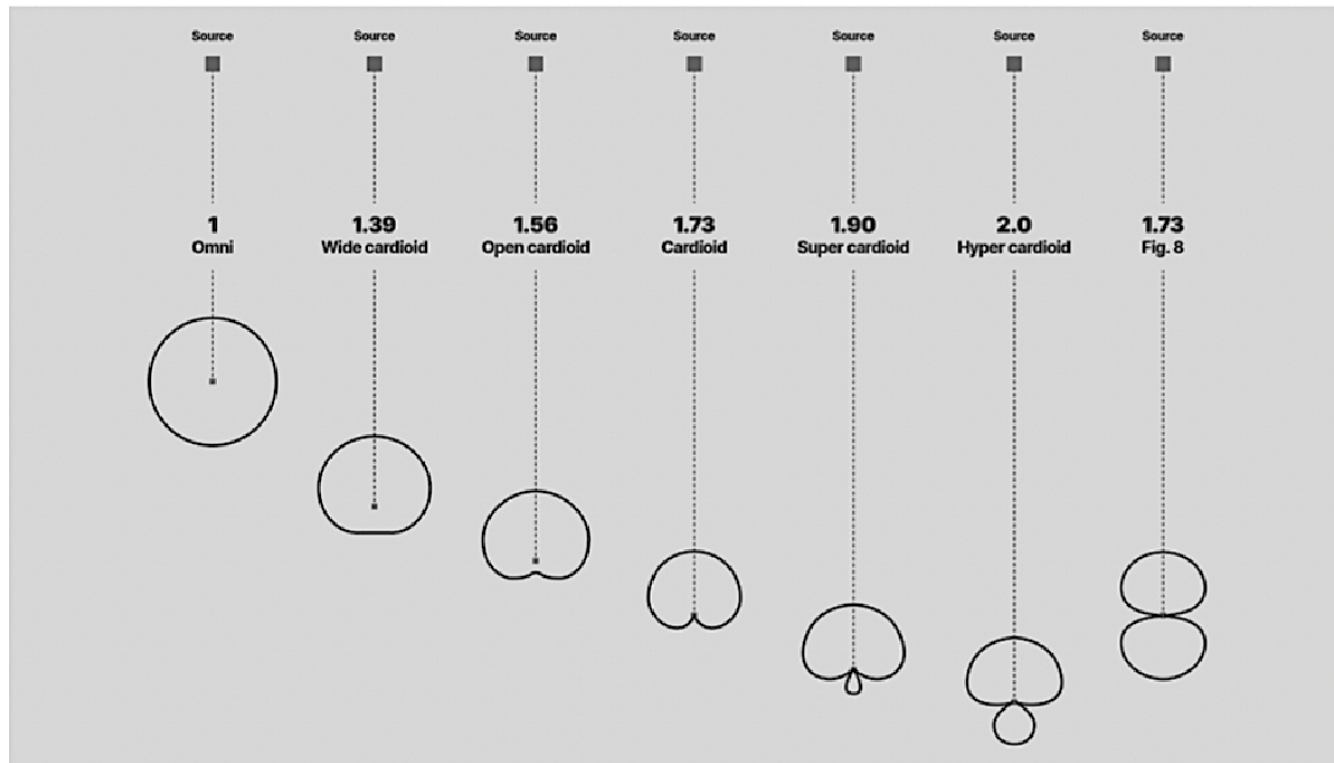
(a) Spatial Quality

Ordre 3



(b) Timbral Quality

[H.Lee, M.Frank, F.Zotter, AES IAA 2019]  
[F.Zotter, M.Frank, Ambisonics, 2019]



*Facteur de distance (DSF) des microphones de premier ordre.*

## Facteur de directivité

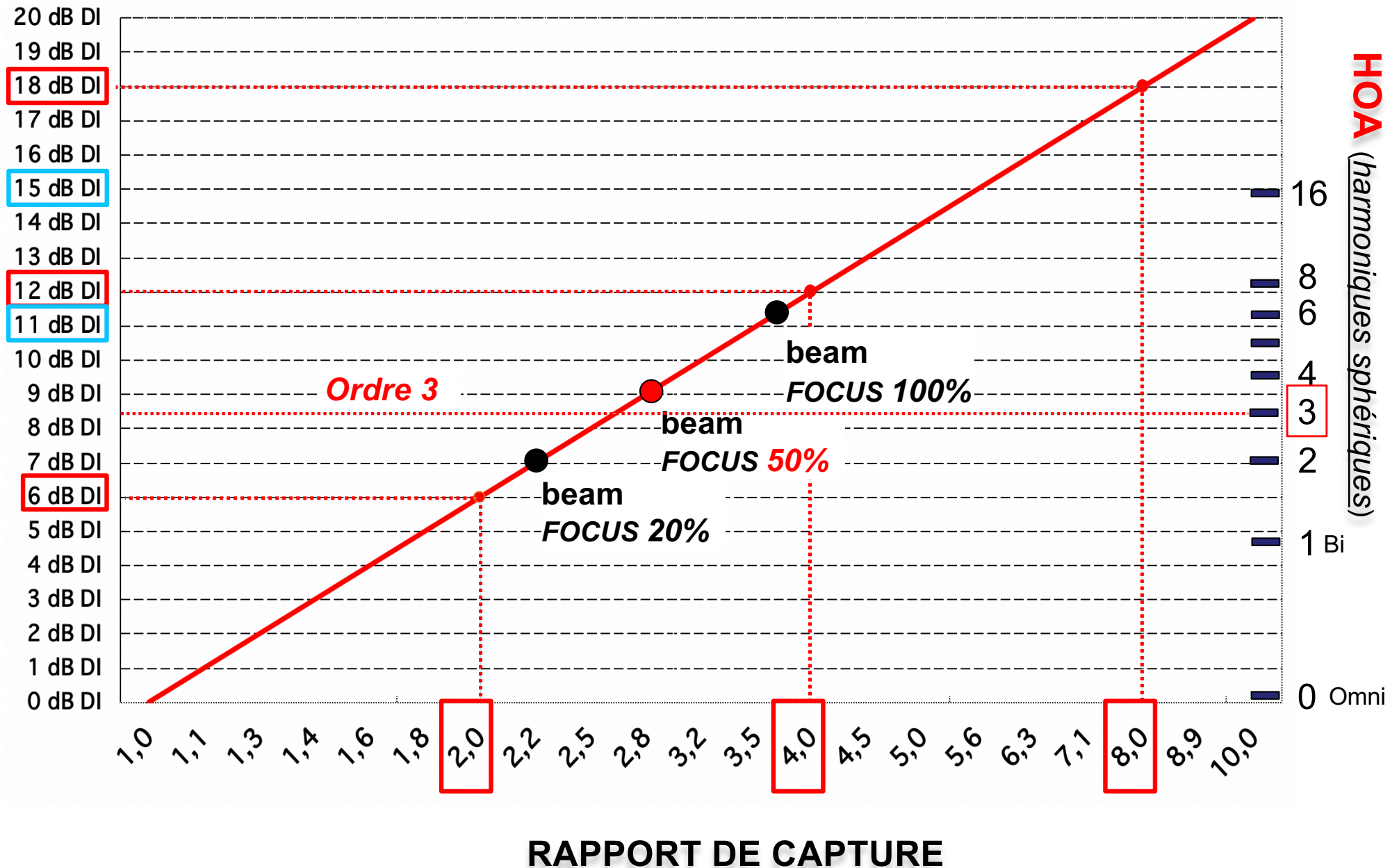
Le facteur de directivité (DF ou Q) est défini comme le rapport entre l'énergie captée dans l'axe et l'énergie captée dans toutes les directions.

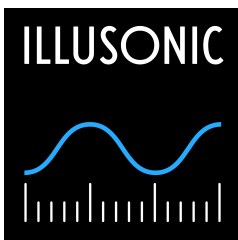
## Indice de directivité

L'indice de directivité (D ou DI) est le facteur de directivité (DF) en dB :  $(DI = 10 * \log DF)$ .

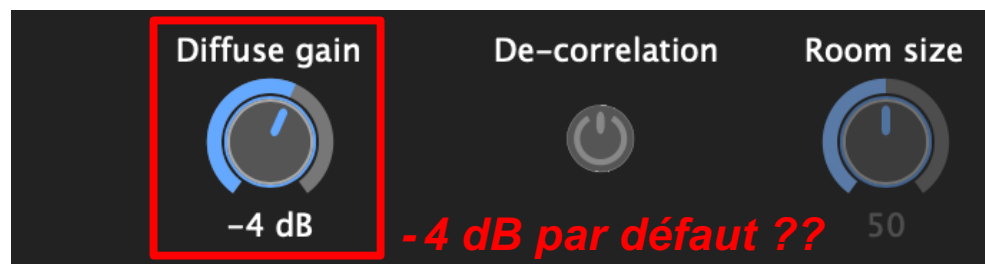


# RAPPORT DE CAPTURE en fonction de l'Indice de Directivité DI





# A/B – Format Decoder v5.1.0



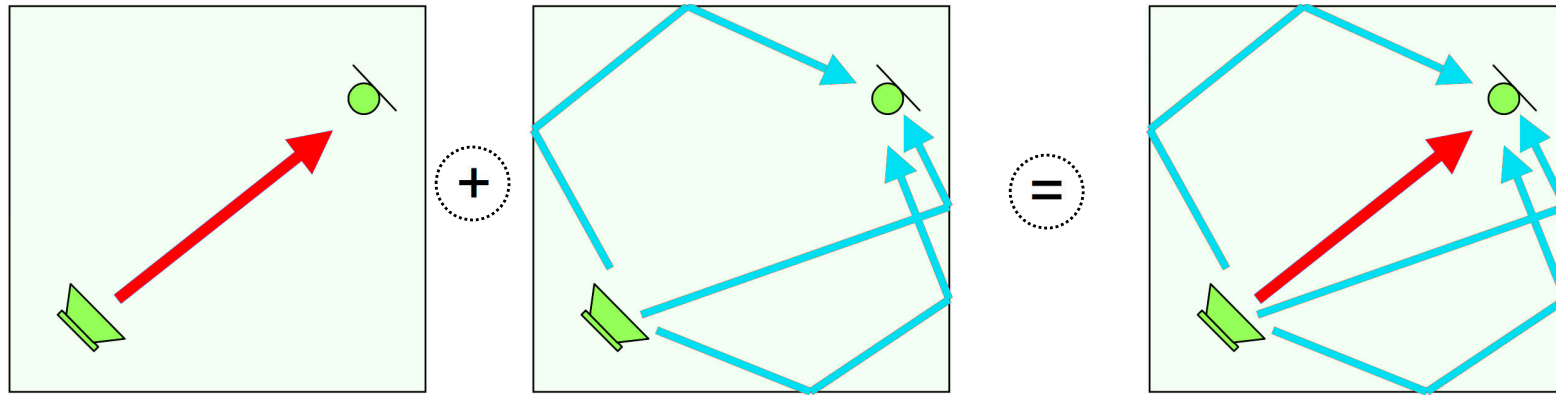
Diffuse gain = **DI** Directivité Index

↘ Diffuse gain dB = ↗ **DI** dB

Diffuse gain de **-6 dB** = **+6 dB DI**

- **Diffuse gain** : ajuste la quantité d'énergie sonore diffuse contenue dans l'enregistrement de **-18 dB** à **+6 dB**. Le gain d'ambiance affecte directement les signaux d'entrée et modifie donc tous les canaux simultanément. Ce curseur de contrôle peut être utilisé pour ajouter plus de son de salle à un enregistrement sans ajouter de réverbération artificielle. Dans cet algorithme, le son diffus contenu dans les signaux d'entrée est extrait puis augmenté ou diminué.

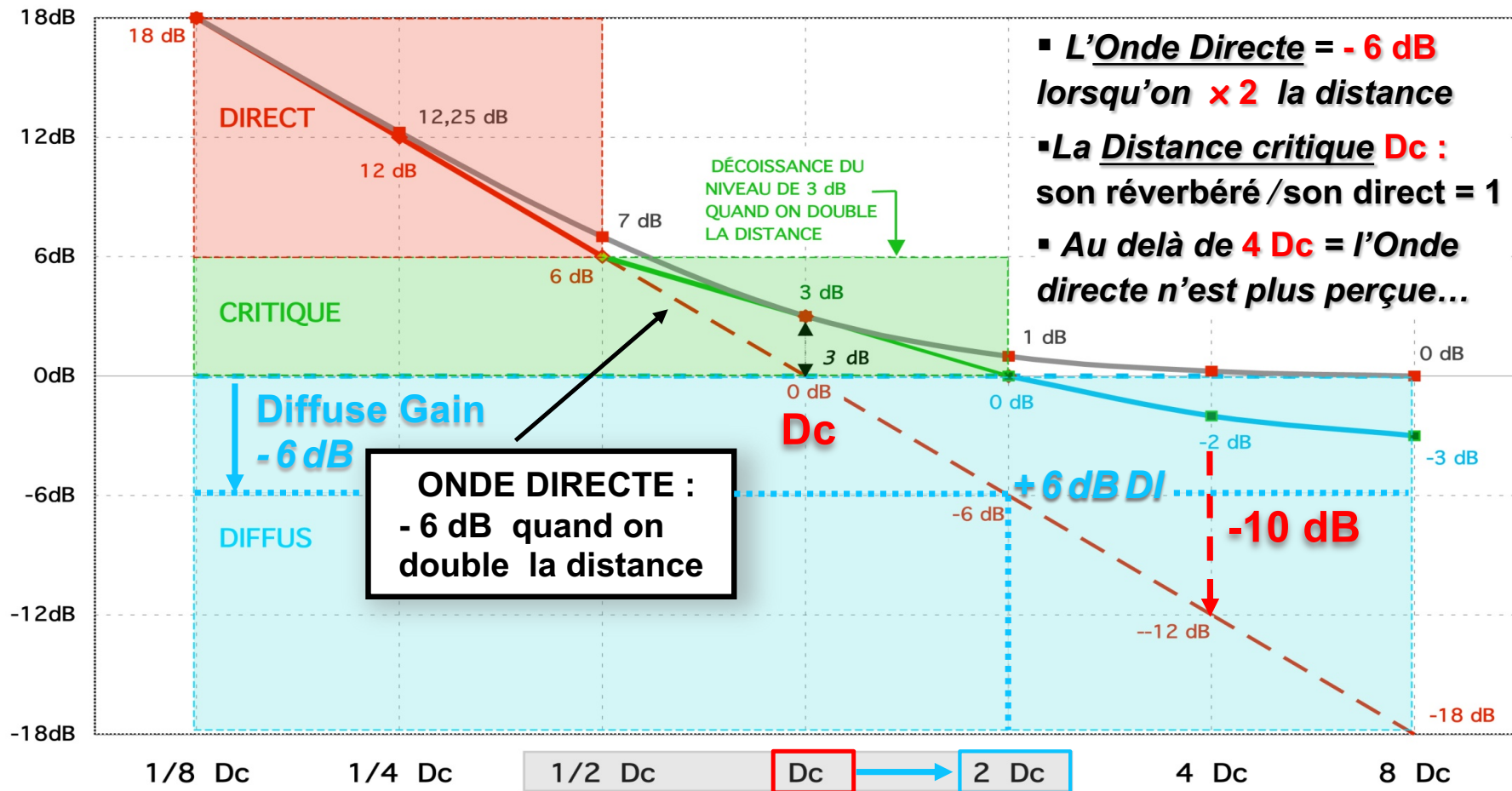
# CHAMP DIRECT - CHAMP DIFFUS - DISTANCE CRITIQUE $D_c$



Champ direct

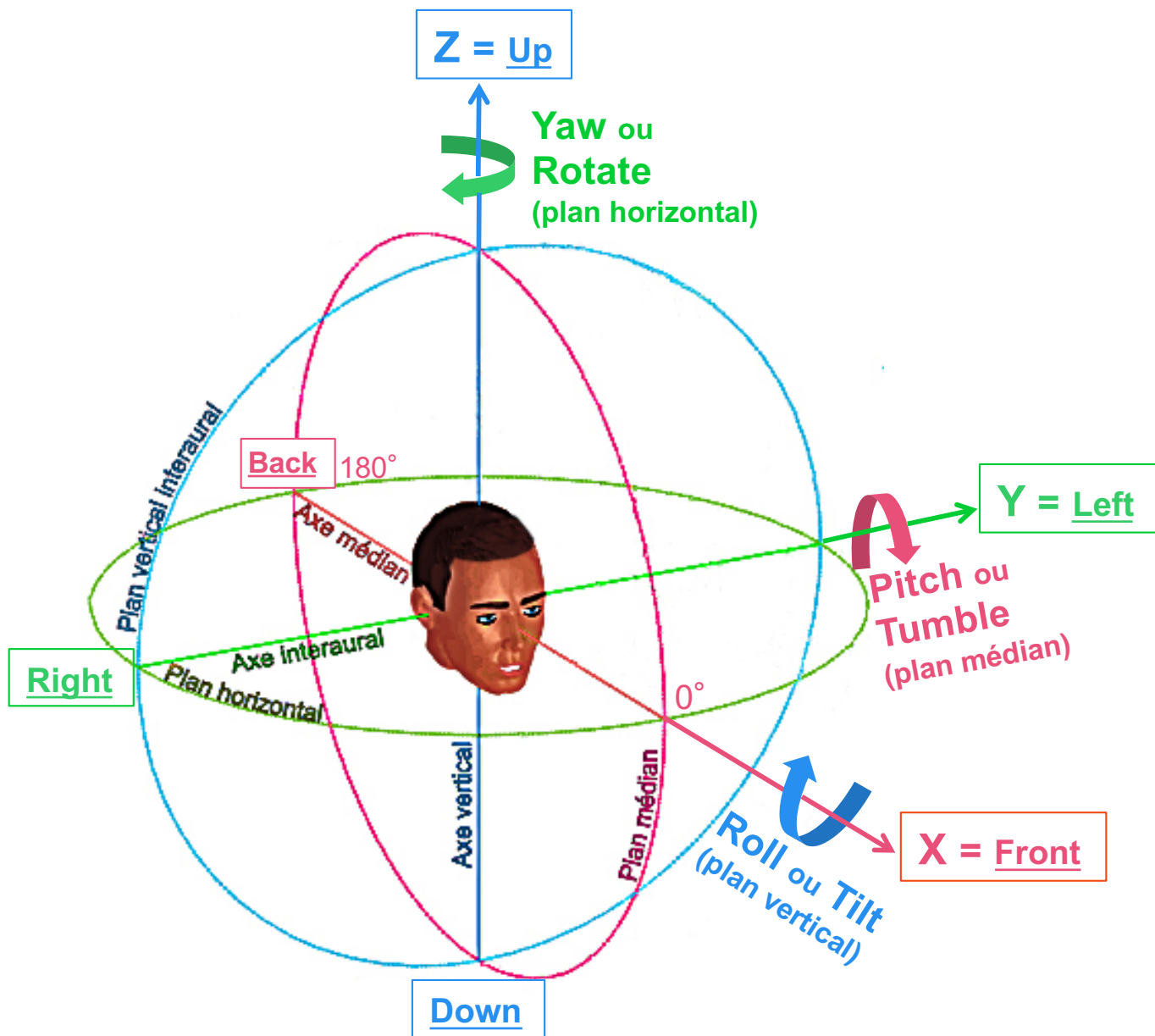
Champ diffus

Propagation dans un local



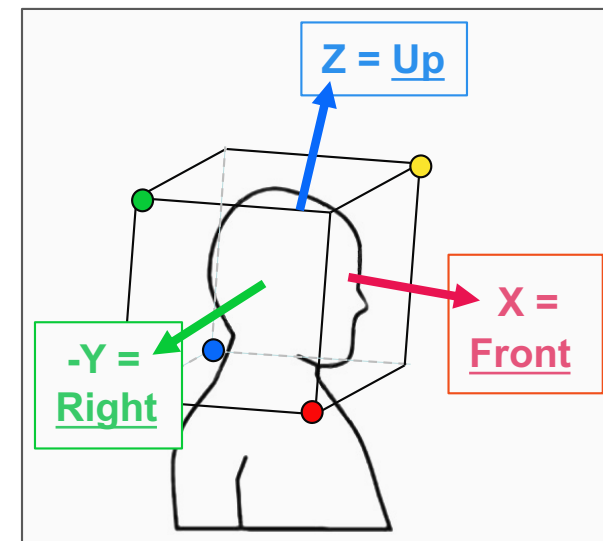
- **L'Onde Directe = - 6 dB** lorsqu'on **x 2** la distance
- **La Distance critique  $D_c$  :** son réverbéré / son direct = 1
- **Au delà de 4  $D_c$  = l'Onde directe n'est plus perçue...**

# Rotation Ambisonic **3D**



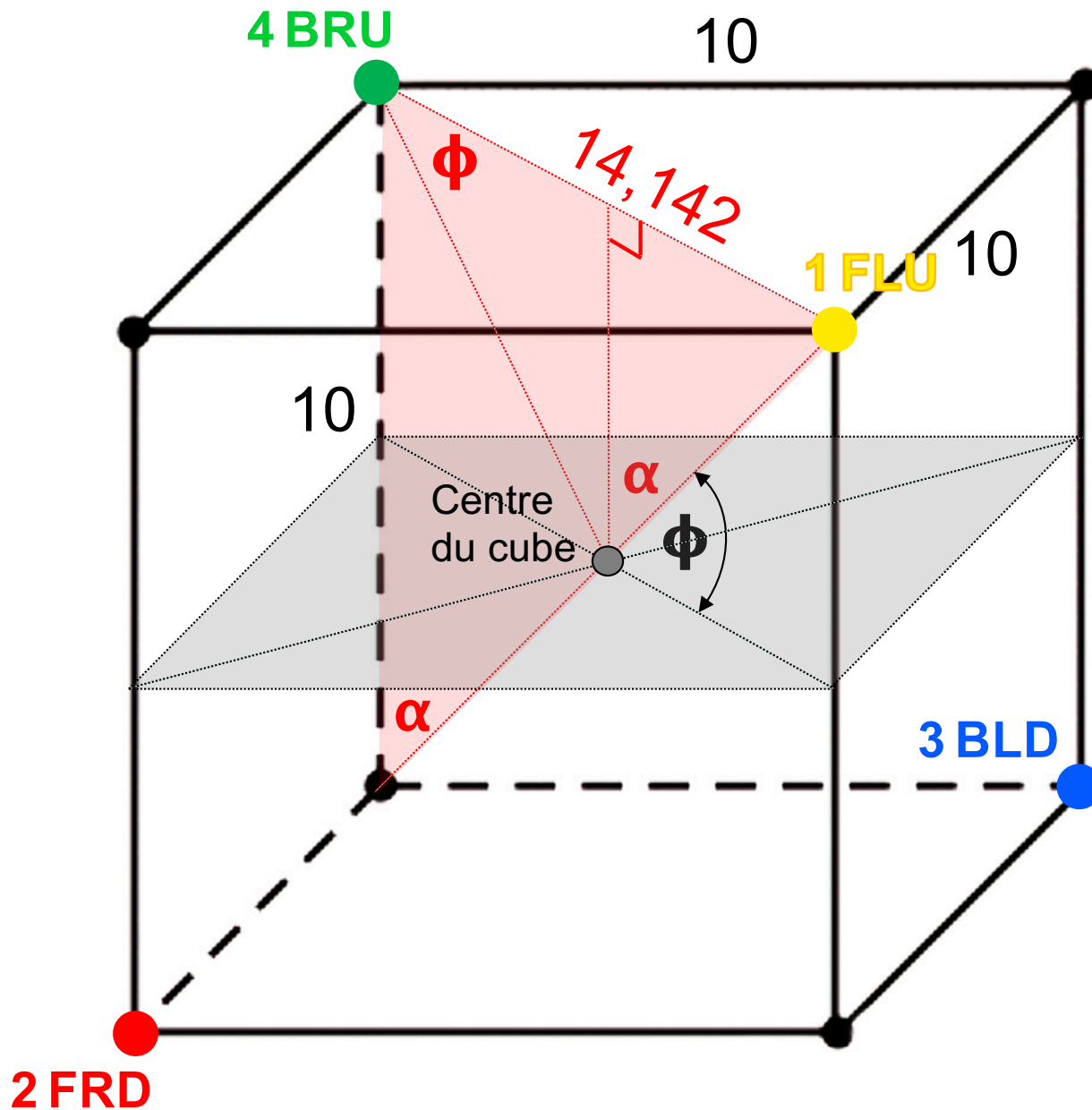
## Les 3 Plans :

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



Représentation des capsules par rapport aux axes XYZ...

# Principes du CUBE Ambisonique



$$\text{Tang } \alpha = \frac{14,142}{10}$$

$$\alpha = 54,73^\circ$$

Angle entre 1 FLU et 4 BRU :

$$\alpha \times 2 = 109,46^\circ$$

Élévation de 1 FLU :

$$\phi = 35,264^\circ$$

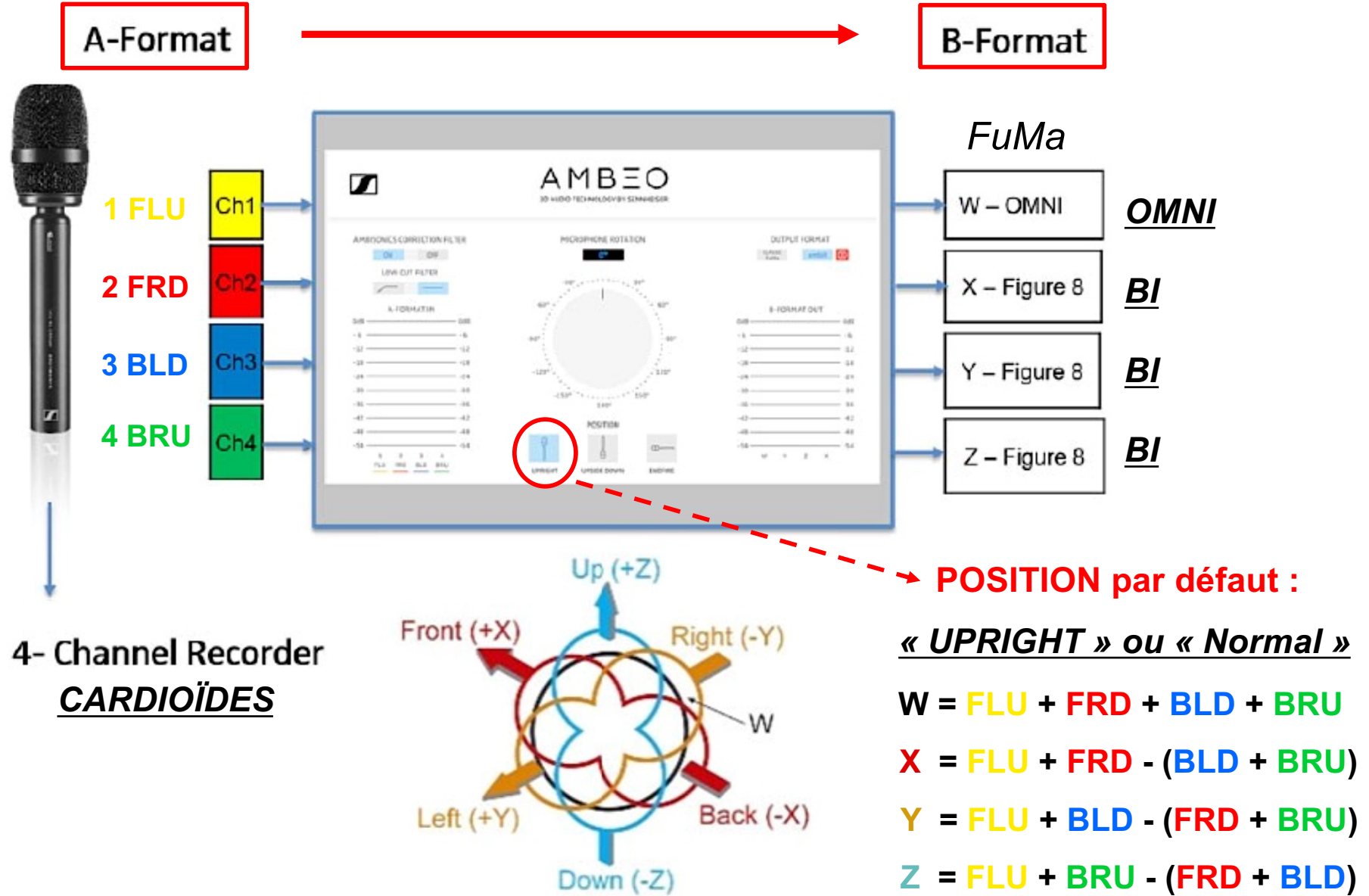


SENNHEISER

# AMBEO

3D AUDIO TECHNOLOGY BY SENNHEISER

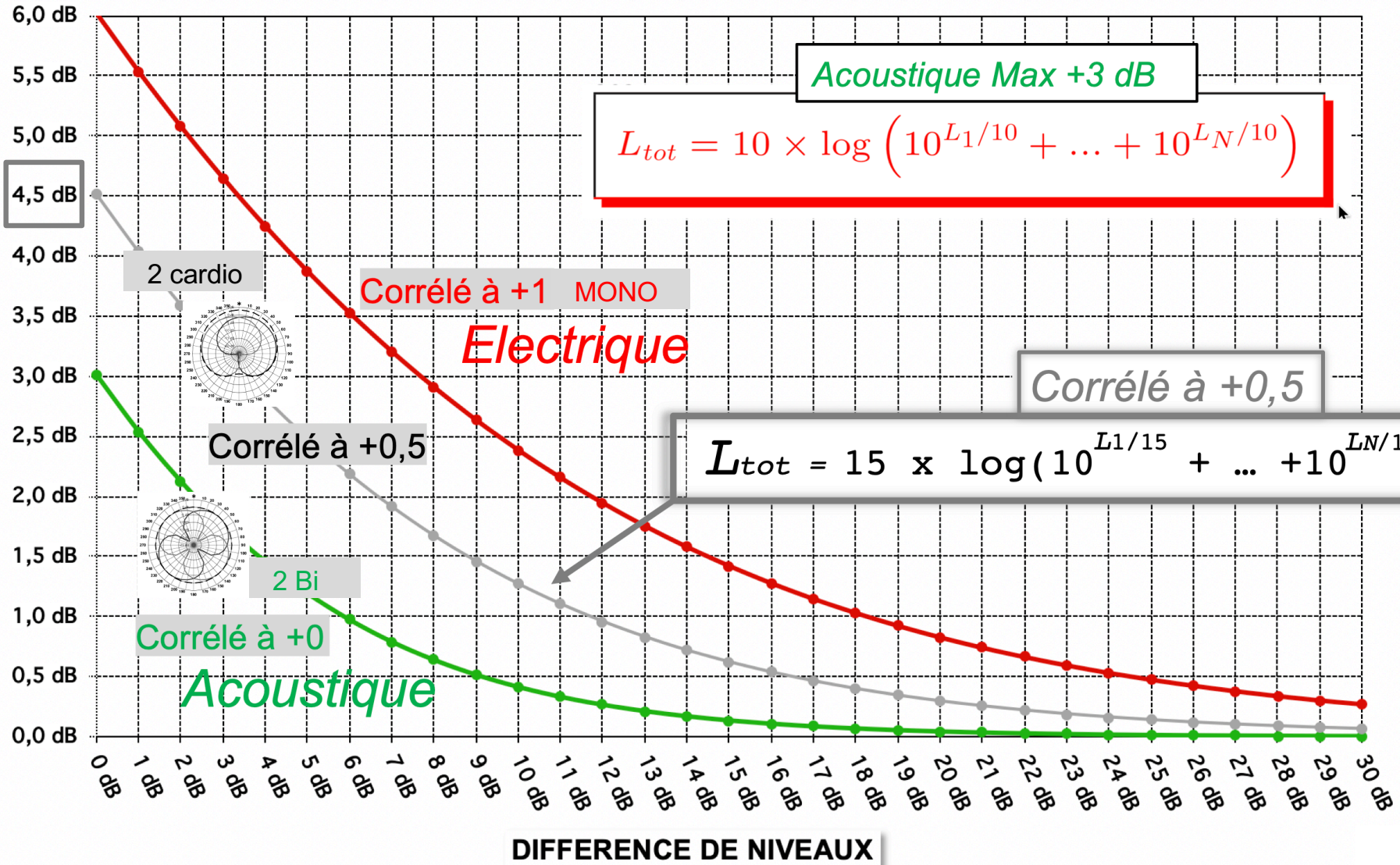
Plug-in convertisseur de format A vers le Format B spécialement conçu par Sennheiser, téléchargeable **gratuitement** en format VST, AU ou AAX.



4- Channel Recorder  
CARDIOÏDES

# Addition des niveaux

Correction à ajouter au niveau le plus élevé



## Caractéristiques du couple stéréophonique :

\* Directivité  
des micros **L** et **R**

**0,500**

Angle entre  
les micros **L** et **R**

**109 °**

Distance entre  
les micros **L** et **R**

**2,8 cm**

\* Directivité après la  
SOMMATION de **L** et **R**  
( signaux en phases )

**0,633**

Distance de la  
source sonore

**10,0 m**

Pourcentage en niveau  $\Delta L$  et en temps  $\Delta T$   
( entre les micros **L** et **R** )

$\Delta L$  dB

$\Delta T$  ms

**94 %**

**6 %**

Angle total de  
prise de son utile  
du couple

**120 °**

Affaiblissements  
à l' avant **0°**  
du couple

**-2,0 dB**

Affaiblissements  
à l' arrière **180°**  
du couple

**-13,6 dB**

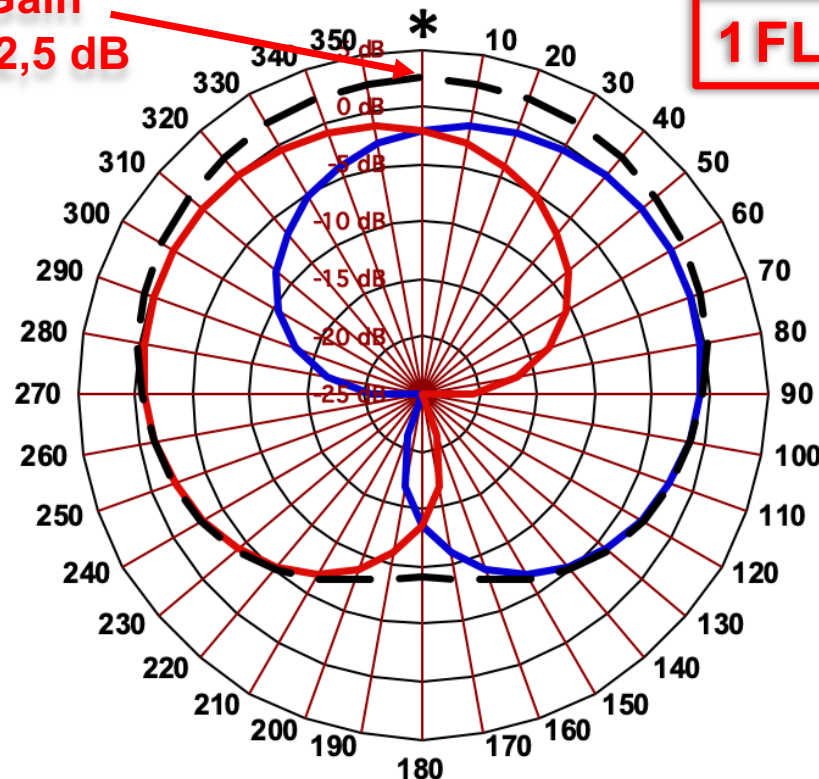
Après SOMMATION :  
coefficient de directivité  
du couple **Q**  
( réf du Cardio :  $Q = 3$  )

**1,9**

Rapport de capture  
ou Facteur de Distance =  $\sqrt{Q}$

**1,4**

Gain  
**+2,5 dB**



**1 FLU + 2 FRD**

## Matriçage Axe X

\* NOTE :

Micro OMNI = 1

Micro INFRA  $\approx 0,66$  ( -10 dB arrière)

Micro CARDIO = 0,5

Micro SUPER  $\approx 0,375$  (-12 dB arrière)

Micro BI = 0

## LES LIENS :

[https://www.lesonbinaural.fr/EDIT/EXCEL/Angle\\_de\\_prise\\_de\\_son\\_pour\\_un\\_couple\\_stereo.xls](https://www.lesonbinaural.fr/EDIT/EXCEL/Angle_de_prise_de_son_pour_un_couple_stereo.xls)

<https://www.lesonbinaural.fr>



* Caractéristique du micro <b>FRONTAL</b>	<b>0,633</b>
* Caractéristique du micro <b>DORSAL</b>	<b>0,633</b>

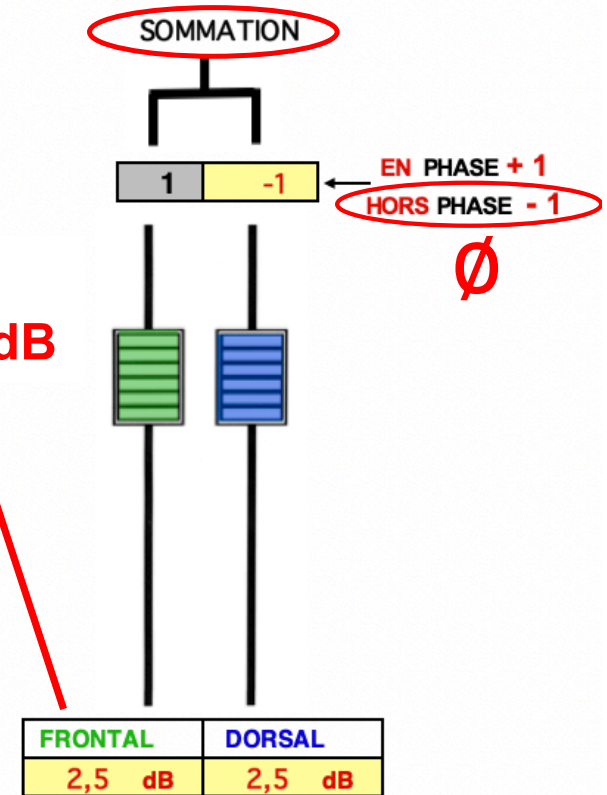
Différence de sensibilité entre le micro <b>FRONTAL</b> et le micro <b>DORSAL</b>	<b>0,0 dB</b>
---	---------------

RÉSULTATS DE LA SOMMATION DES 2 MICROS VISANT DANS DES DIRECTIONS OPPOSÉES	
* Caractéristique de directivité du micro après sommation	Niveau maximum du micro après sommation
<b>0,000</b>	<b>-0,2 dB</b>

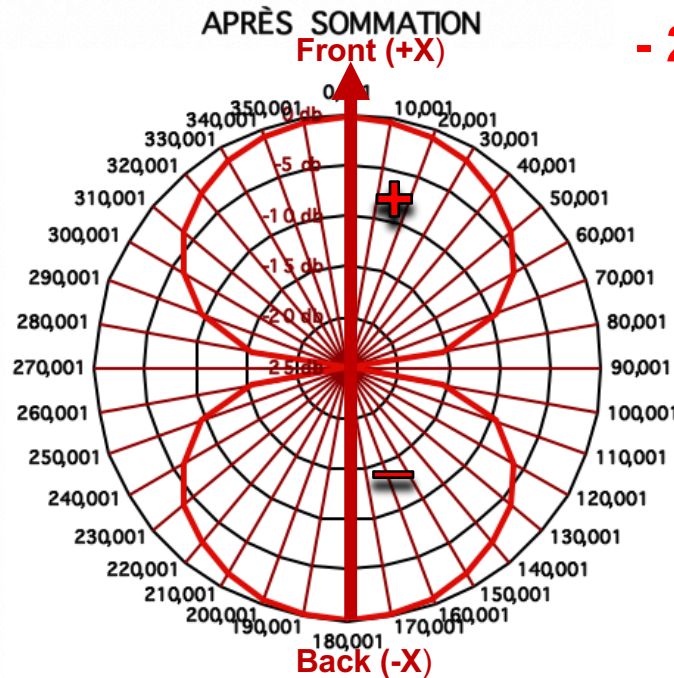
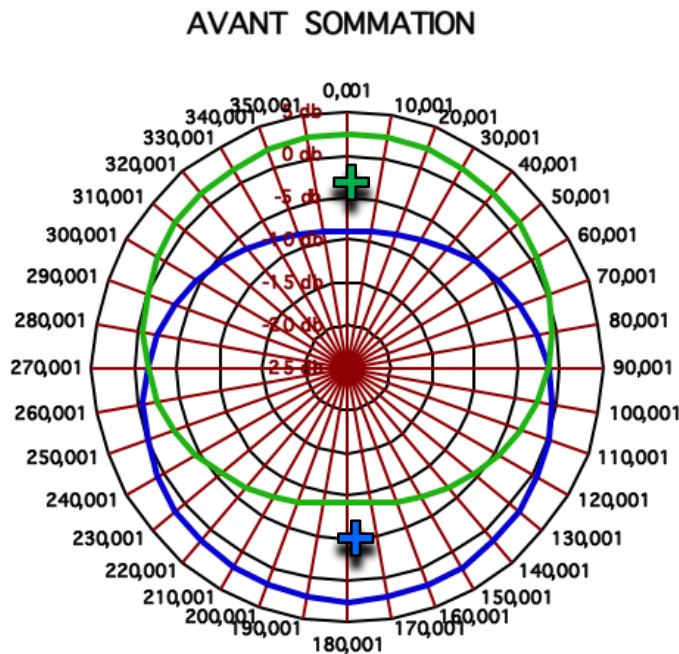
**\* NOTE :**

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

# Matriçage Axe X



Perte - 2,7 dB



Copyright © 2009 Bernard Lagnel

$$X = 1FLU + 2FRD - (3BLD + 4BRU)$$

# Matriçage Ambisonic

* Caractéristique du micro <b>FRONTAL</b>	<b>0,633</b>
* Caractéristique du micro <b>DORSAL</b>	<b>0,633</b>

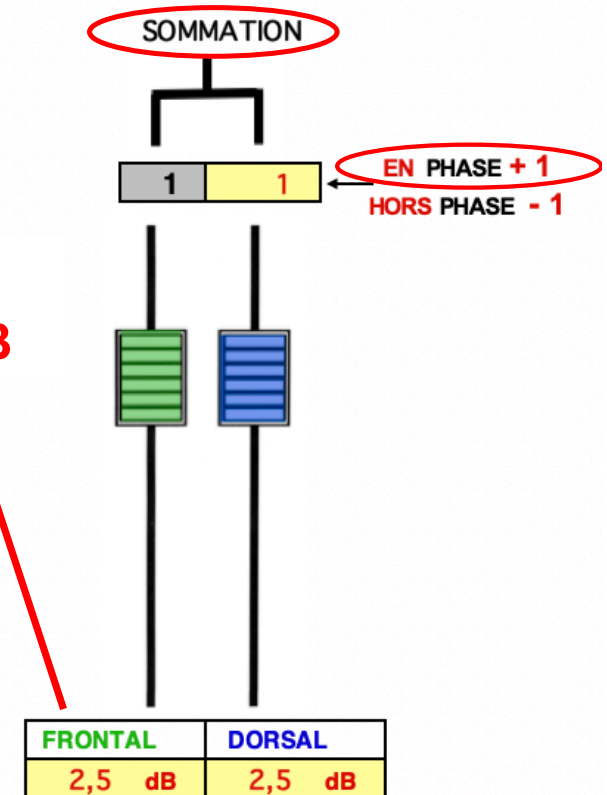
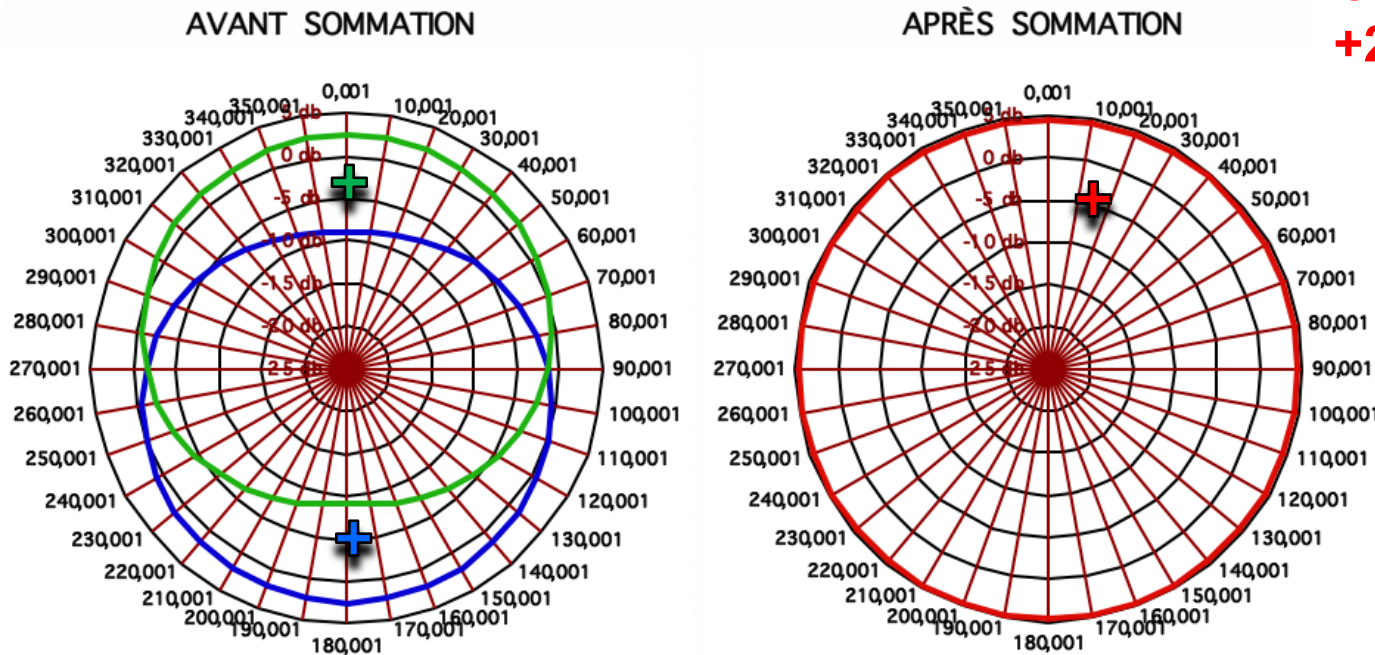
Différence de sensibilité entre le micro <b>FRONTAL</b> et le micro <b>DORSAL</b>	<b>0,0 dB</b>
---	---------------

RÉSULTATS DE LA SOMMATION DES 2 MICROS VISANT DANS DES DIRECTIONS OPPOSÉES	
* Caractéristique de directivité du micro après sommation	Niveau maximum du micro après sommation
<b>1,000</b>	<b>4,5 dB</b>

\* 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

# Matriçage Omni W

Gain +2 dB



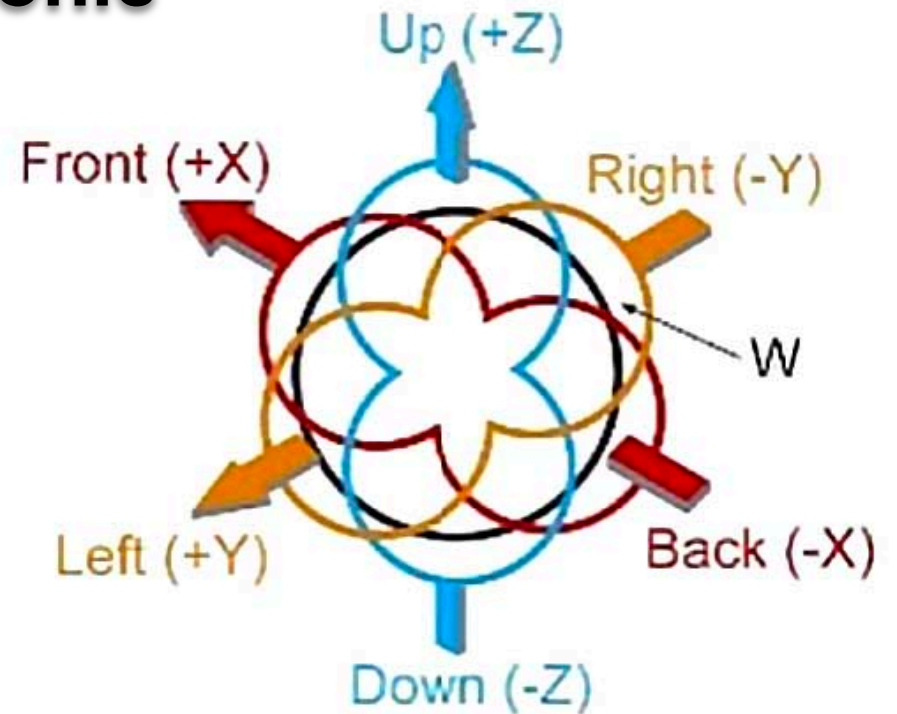
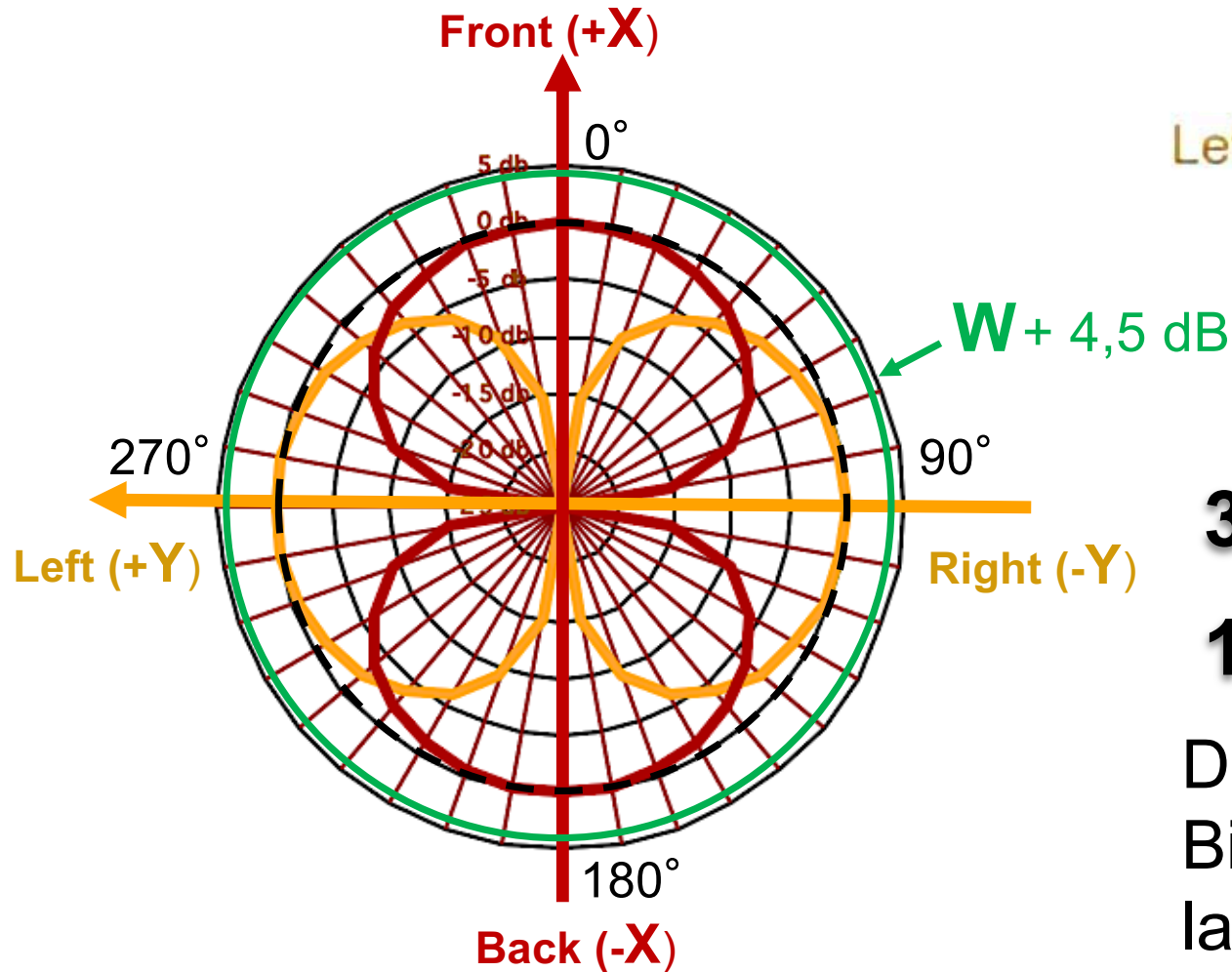
Copyright © 2009 Bernard Lagnel

$$W = 1FLU + 2FRD + (3BLD + 4BRU)$$

# Matriçage Ambisonic

# Sphère Ambisonic

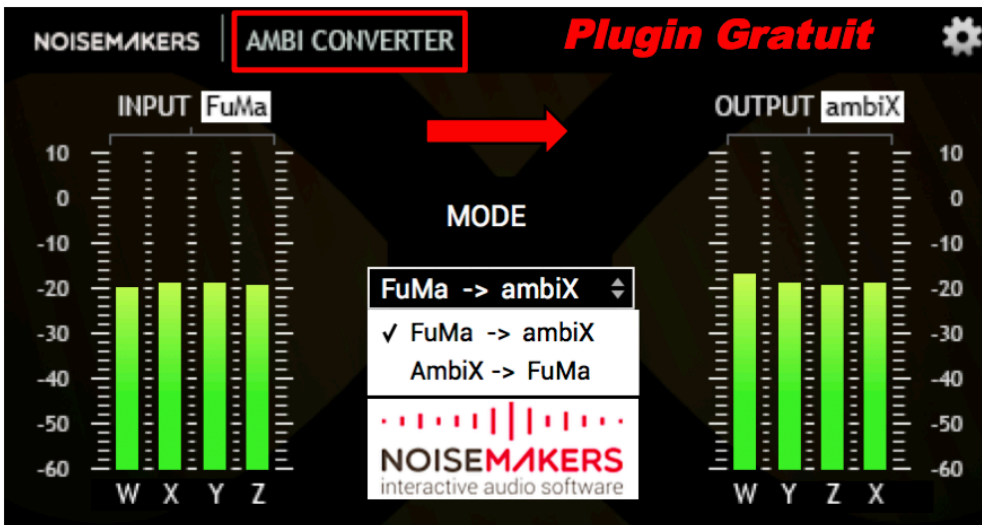
## B-Format



**3 Bi XYZ = -0,2 dB**

**1 Omni W = +4,5 dB**

Différence de 4,7 dB entre Bi et Omni quelle que soit la corrélation...



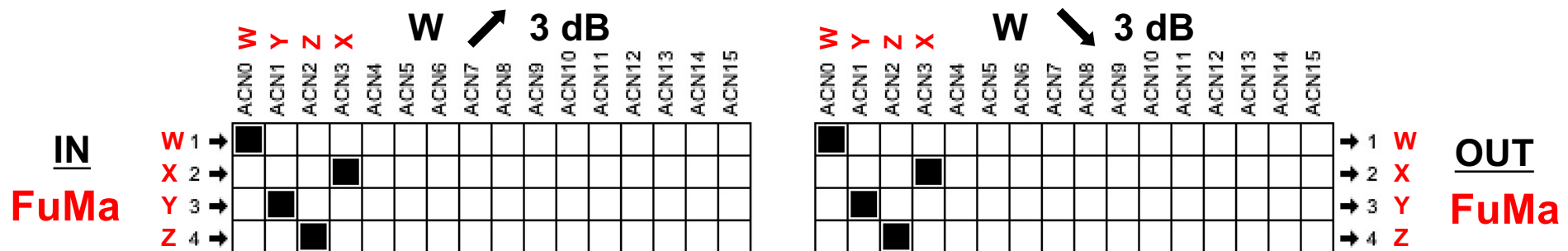
<https://www.noisemakers.fr/ambi-converter/>

## DAW REAPER VST

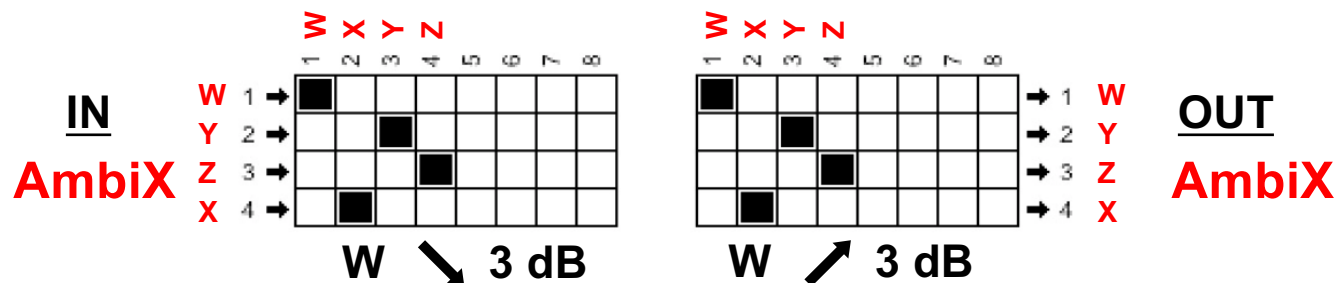
### Conversion de FuMa vers AmbiX

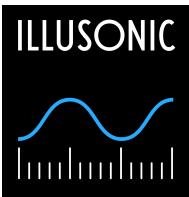
- « FuMa » signifie «Furse-Malham», c'est à dire que l'ordre des canaux est **W X Y Z** avec le canal W normalisé :  $1/\sqrt{2} = -3 \text{ dB}$ .
- « AmbiX » signifie l'ordre des canaux ACN avec la normalisation SN3D, c'est à dire que l'ordre des canaux est **W Y Z X** sans mise à l'échelle des canaux (W > +3 dB à Y Z X).

### Insertion d'un plugin AmbiX (ou SN3D) dans une chaine FuMa :



### Insertion d'un plugin FuMa dans une chaine AmbiX (ou SN3D) :

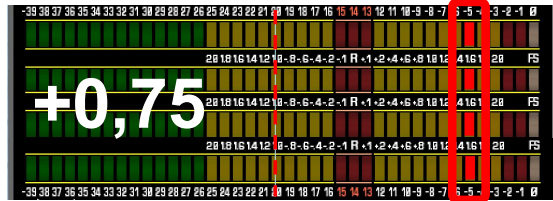




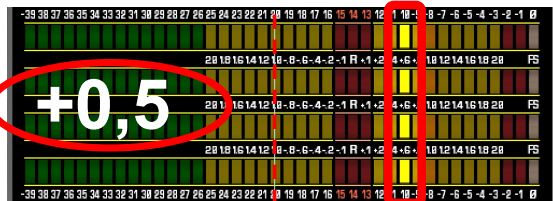
# Corrélation correspondant au B-Format **AmbiX**

## **B - Format (AmbiX)**

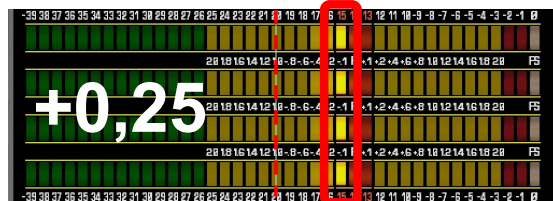
Bruit Rose Corrélé à :



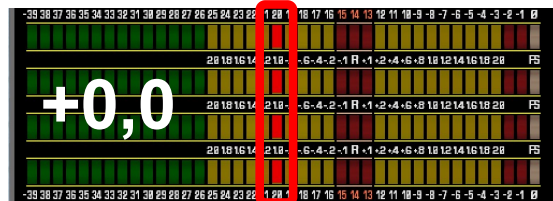
-1 +0 +1



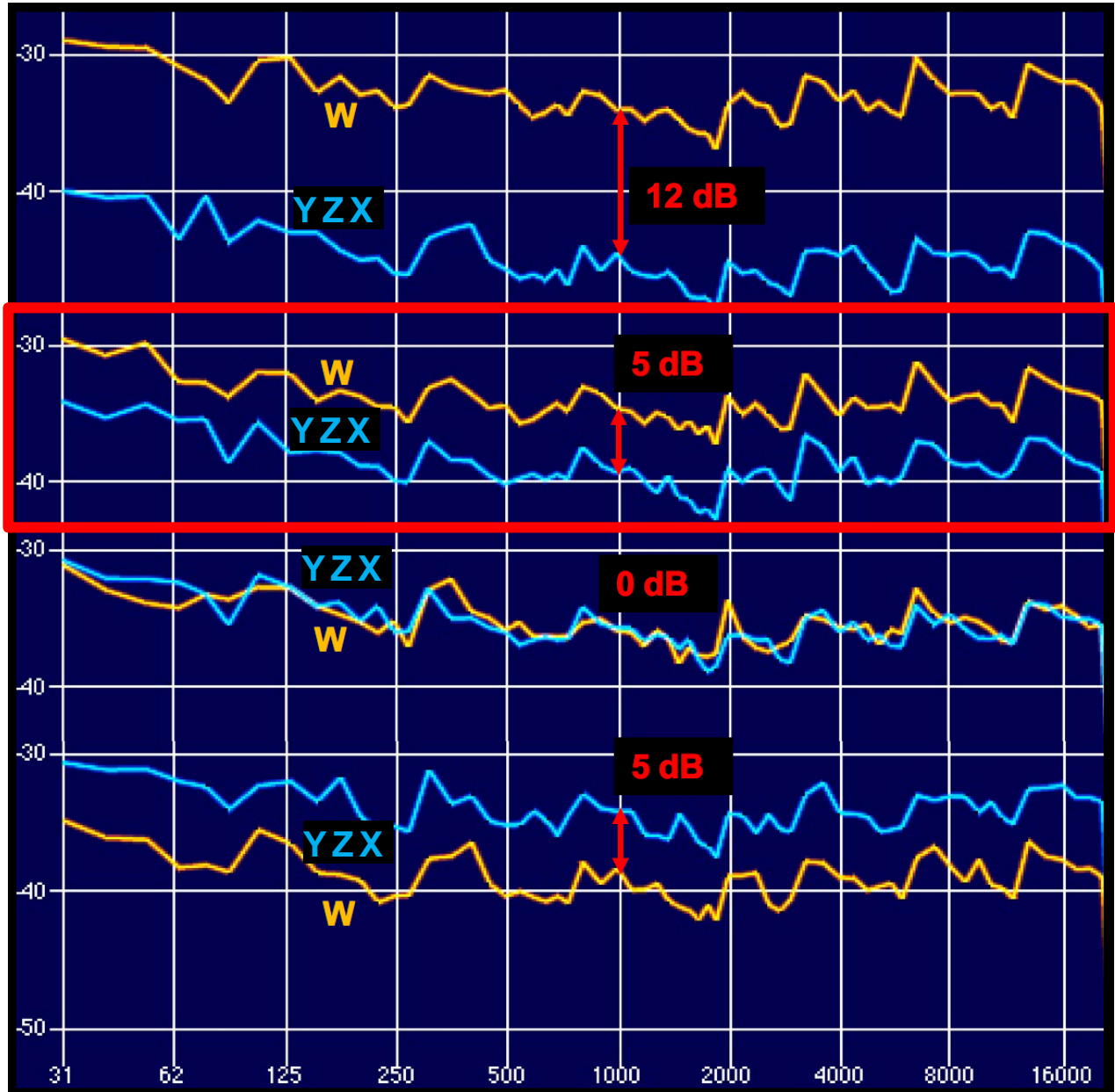
-1 +0 +1



-1 +0 +1



-1 +0 +1



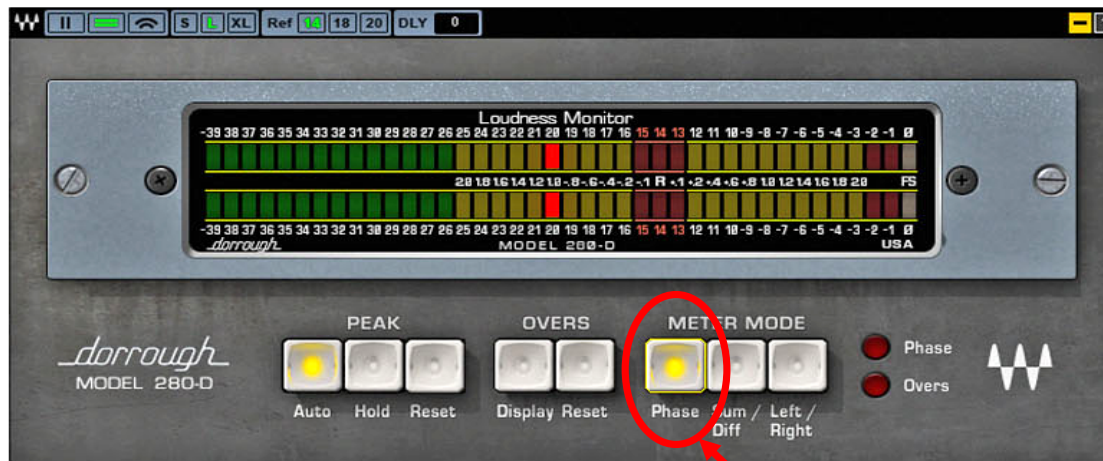


# Le Plug-in WAVES **Dorrough** Stéréo utilisé comme Phasemètre :

## CORRÉLATEUR DE PHASE "ANALOGIQUE"

29\$

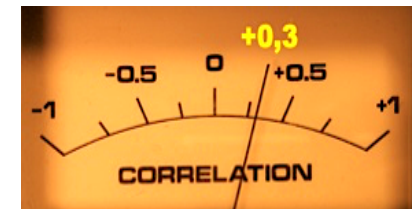
<https://www.waves.com/plugins/dorrough-stereo>



Mode Phase

### Caractéristiques Techniques :

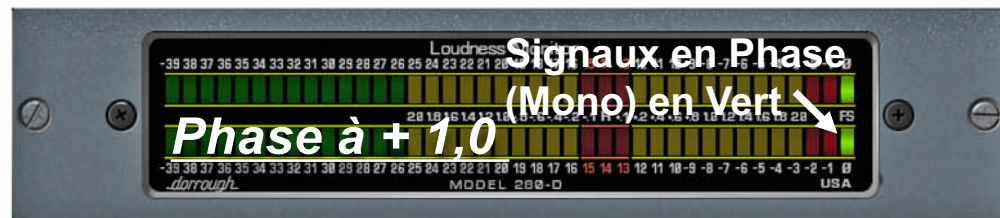
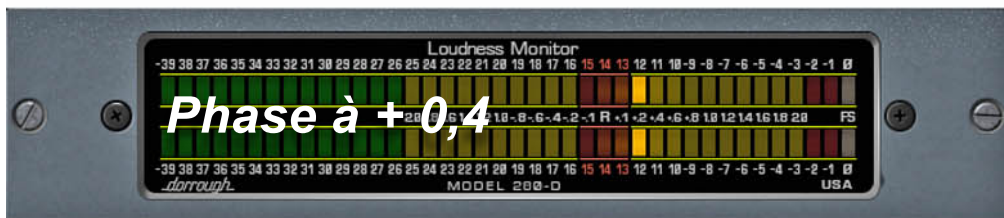
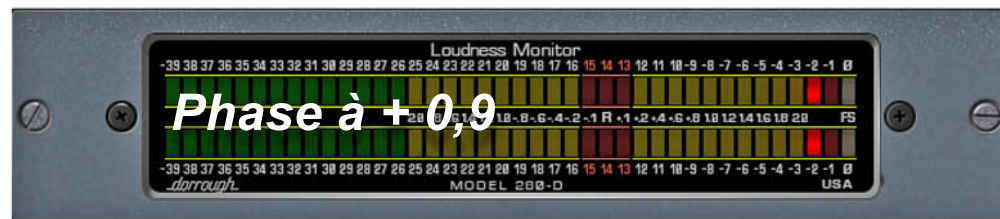
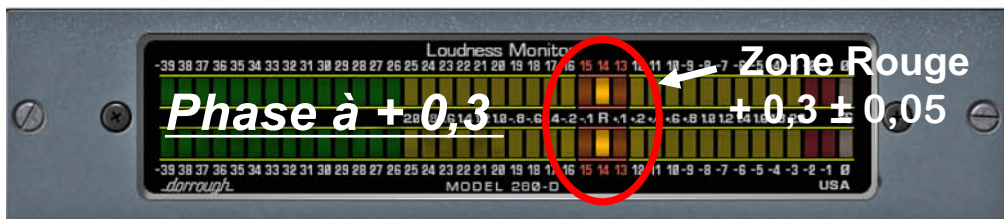
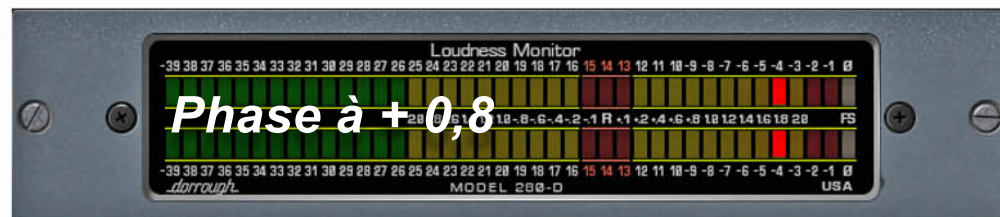
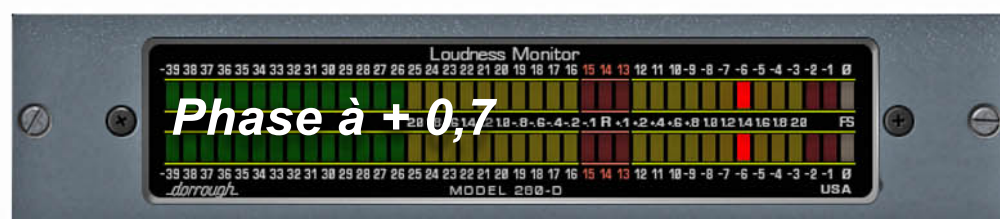
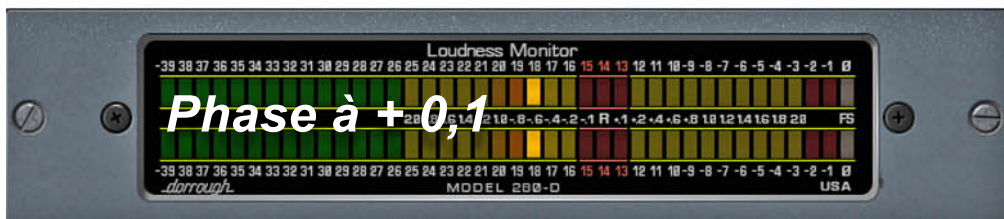
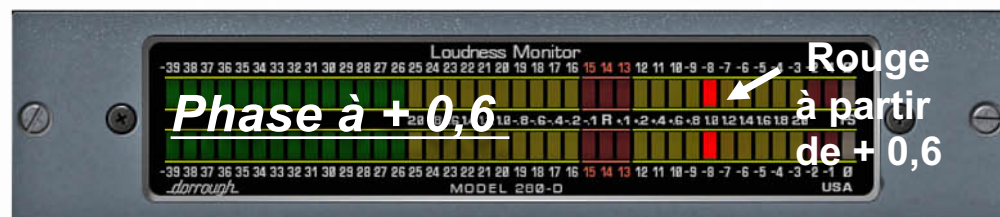
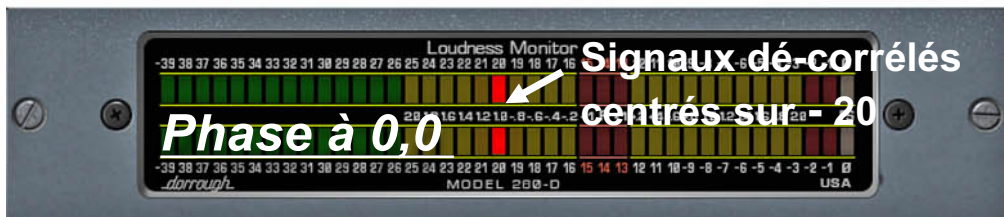
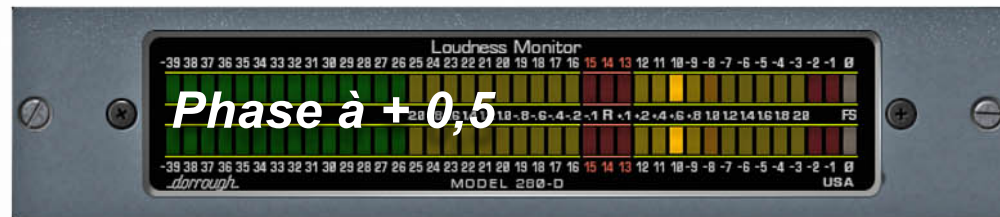
- Temps d'intégration  $\approx$  500 ms
- Seuil de sensibilité pour une réponse exacte  $\approx$  - 32 dBFS (Affichage de la même valeur pour des écarts max de 32 dBFS d'IDL)
- Réponse linéaire de la phase et non logarithmique comme sur la plupart des *Phasemètres Plug-ins*...
- **Phasemètre Plug-in comparable aux phasemètres "Analogiques" du siècle dernier...**



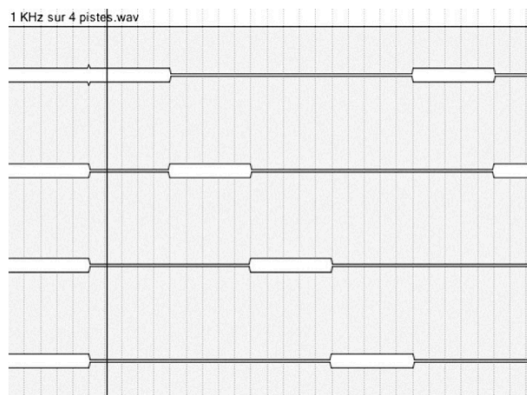
**+0,3 = répartition Stéréo homogène pour une corrélation "Analogique"**

Étude psycho-acoustique faite à Radio France sur du **bruit rose**  
( valable pour la musique classique et les ambiances )

# Indications linéaires de la phase sur le Plug-in Waves **Dorrough** Stéréo :



# Sons Techniques Ambisonics



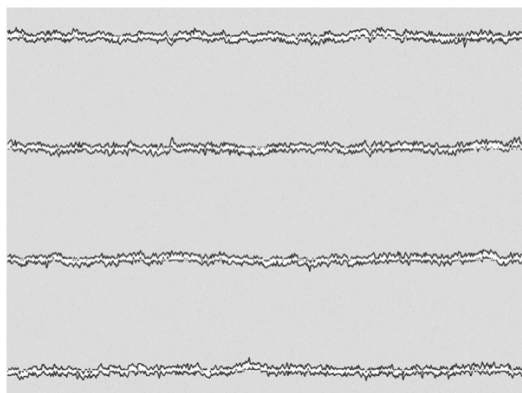
## 1 KHz sur 4 Pistes ©

1 KHz à -18 dBFS sur 4 pistes destiné au Multicanal en Quad et à l'Ambisonic (calibration, équilibre, diaphonie...)  
Cinq cycles de 40 secondes (10 s de modulation par piste)

[Télécharger](#)

3 min 30 sec

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



## 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.

[Télécharger](#)

4 min 40 sec

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

Pour le **A-Format** prendre une Corrélation = +0,5

Pour le **B-Format** prendre uniquement une Dé-corrélation = +0,0



# AMBISONIC = Système Coïncident en ILD

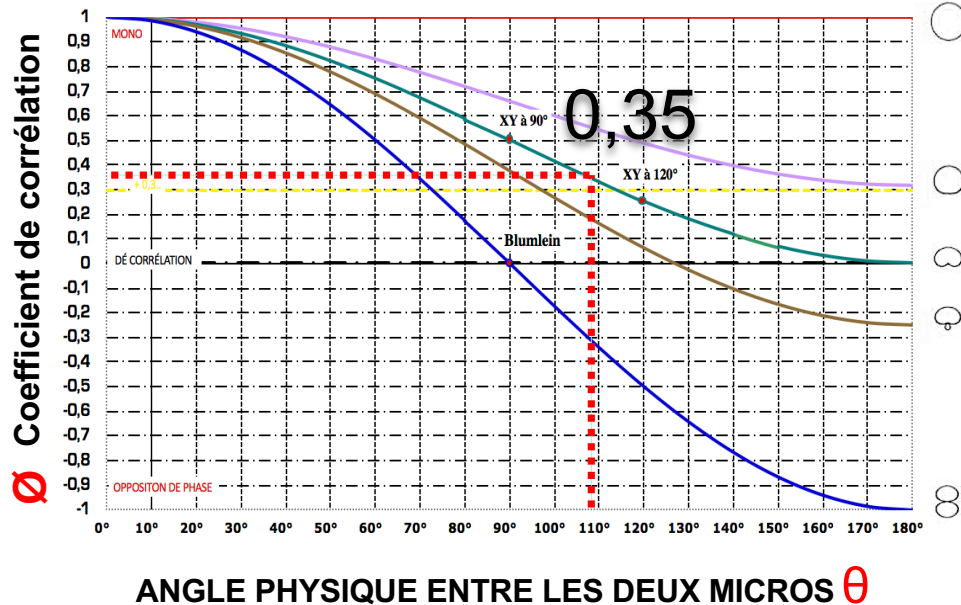
## A-Format : **FLU FRD BLD BRU**

### La Théorie :

$\emptyset$  : coef de corrélation théorique en Champ proche...

$$\emptyset = a + (1 - a) \cdot \text{Cos } \theta$$

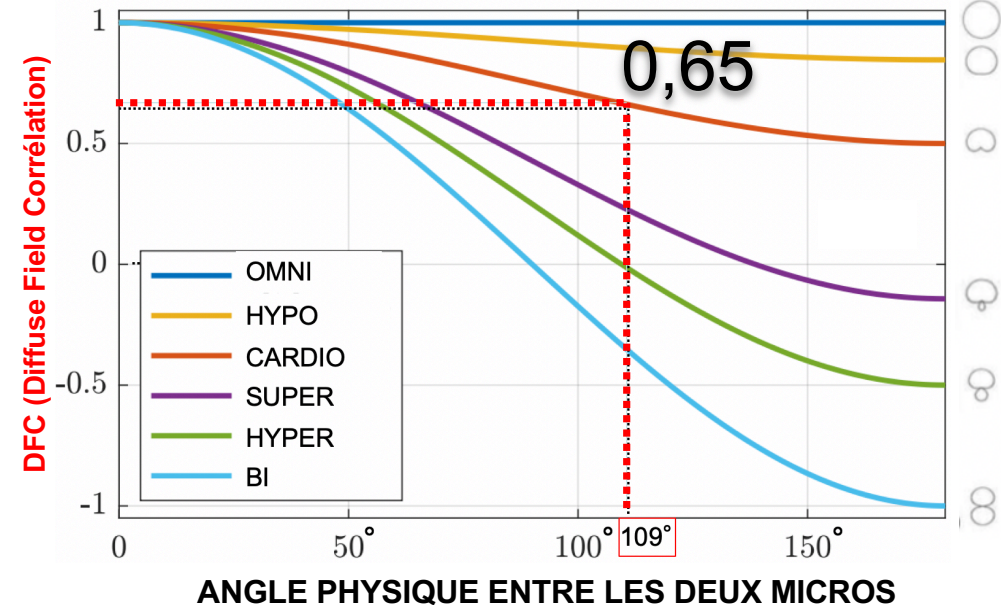
Omni	a = 1
Cardio	a = 0,5
Bi	a = 0



### Dans le Champ diffus...

Fonctions de cohérence spatiale de paires de microphones coïncidents de même types :

Document ILLUSONIC



**+ 0,3 = répartition Stéréo homogène pour une corrélation "Analogique"**

Étude psycho-acoustique faite à Radio France sur du **bruit rose**  
( valable pour la musique classique et les ambiances )

# AMBISONIC = Système Coïncident en ILD

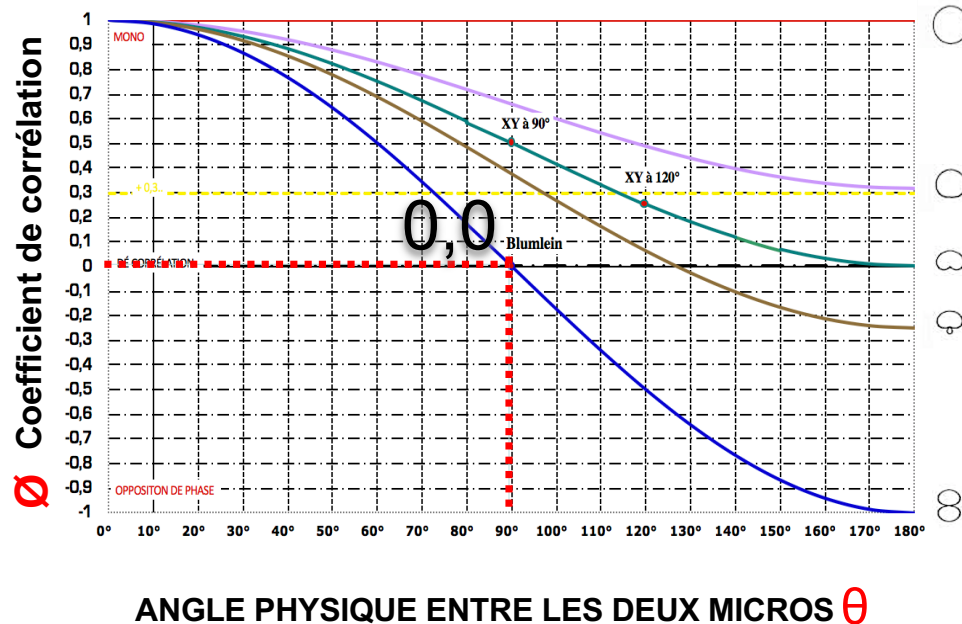
## B-Format : **W** **X** **Y** **Z**

### La Théorie :

$\emptyset$  : coef de corrélation théorique en Champ proche...

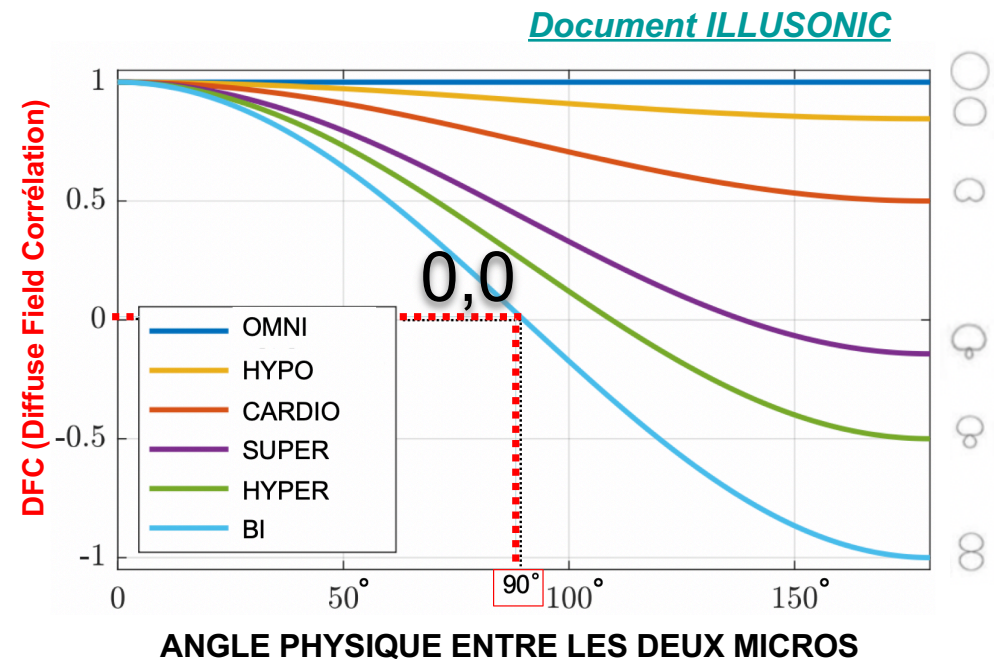
$$\emptyset = a + (1 - a) \cdot \text{Cos } \theta$$

Omni	a = 1
Cardio	a = 0,5
Bi	a = 0

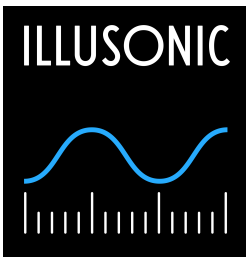


### Dans le Champ diffus...

Fonctions de cohérence spatiale de paires de microphones coïncidents de même types :



**En B-Format : **W** **X** **Y** **Z** sont dé-corrélées...**



# A/B – Format Decoder v5.1.0

**A-Format** (bruit rose corrélé à +0,5) vers **Stereo**

Corrélation en fonction de l'angle Stéréo  
pour des variations du « Diffuse gain »

**A/B-Format Decoder**

**Decoding**

Rotation: 0°    Elevation: 0°

**FOCUS**

Center: 50%    Front: 50%    Wide: 50%    Surround: 50%

Rear: 50%    Front Height: 50%    Surround Height: 50%    Rear Height: 50%

**ANGLE**

Azimuth: Front: 50°    Wide: 60°    Surround: 135°

Rear: 150°    Front Height: 45°    Surround Height: 135°    Rear Height: 150°

Diffuse gain: 0 dB    De-correlation:    Room size: 50

**W Signal Bass**

Cross-over:    Gain: 0 dB    Frequency: 50 Hz

Order: Butterworth 3rd

Invert bass:    Invert bass

**Outputs**

**GAIN**

Center: 0 dB    Front: 0 dB    Wide: 0 dB    Surround: 0 dB

Rear: 0 dB    Front Height: 0 dB    Surround Height: 0 dB    Rear Height: 0 dB

Delay / Shelving: Surround    Delay: 20 ms    Frequency: 6 kHz    Gain: -3 dB

**Formats**

Input format: A-Format

Microphone distance: Coincident    Microphone position: normal

Output format: Stereo

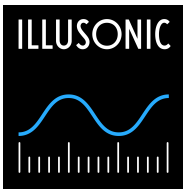
Binaural output:    Binaural output

**Channel ordering**

Input: LF RF LB RB    Output channel test:    Output: L R

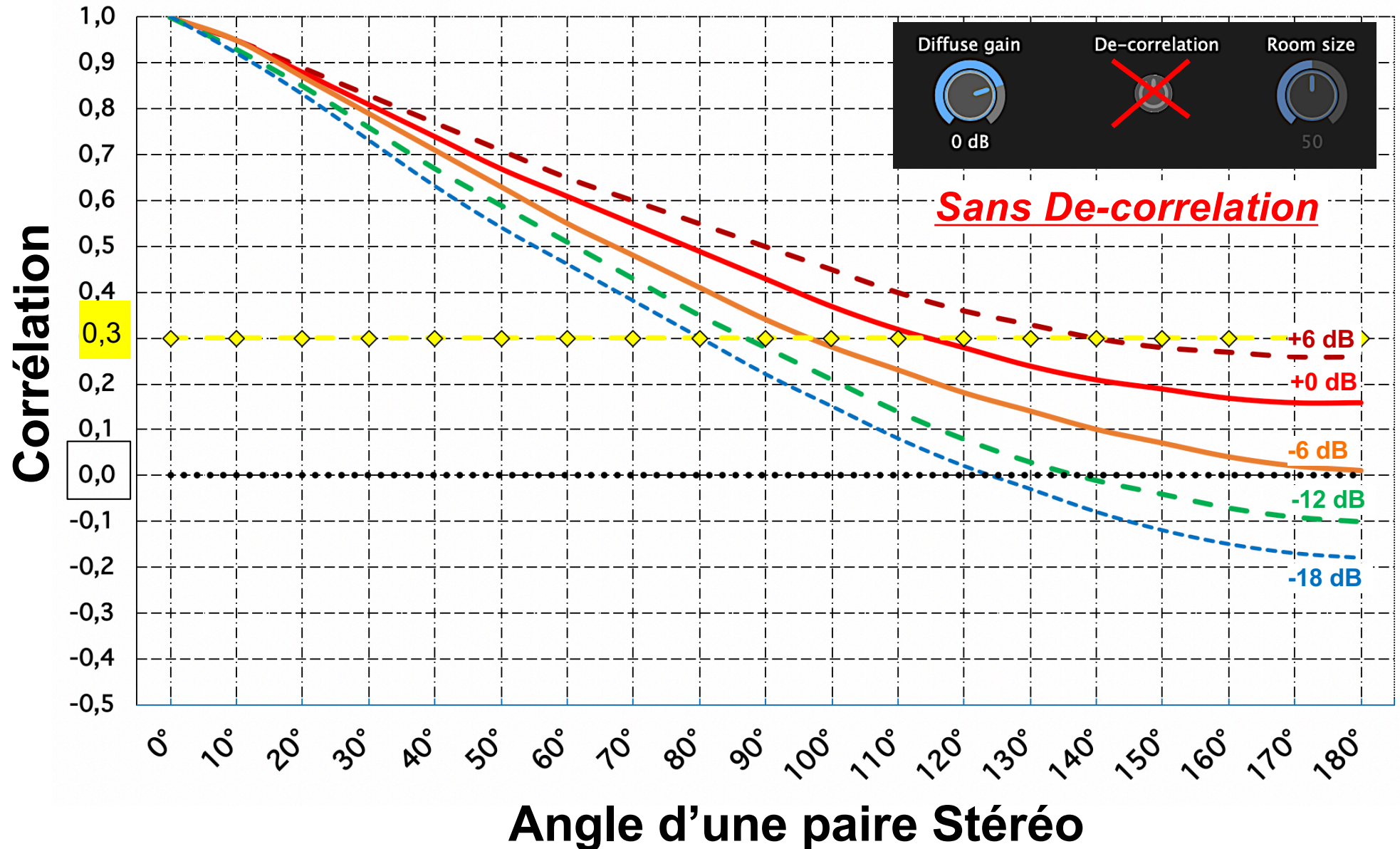
**ILLUSONIC**

© Copyright Illusonic GmbH, Greifensee, Switzerland, 2024. All rights reserved. v5.1.0 - Software expiration: Dec 31, 2025



# Corrélation en fonction de l'angle Stéréo pour des variations du « Diffuse gain »

**Corrélation identique quelque soit la valeur du Focus !**



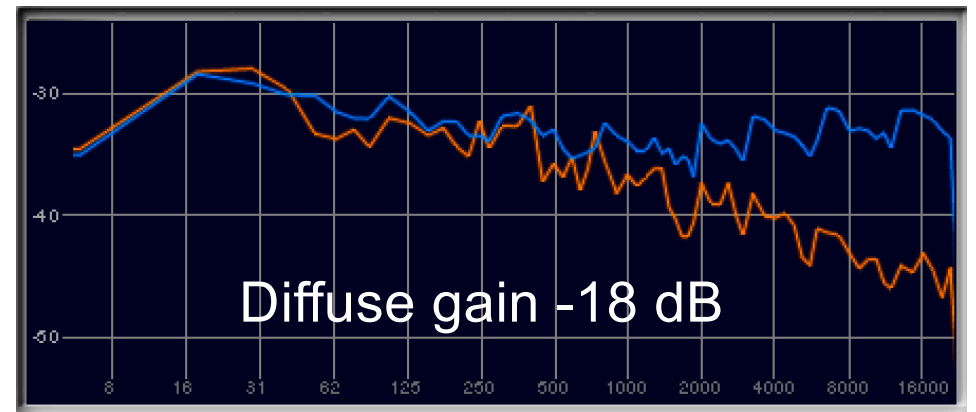
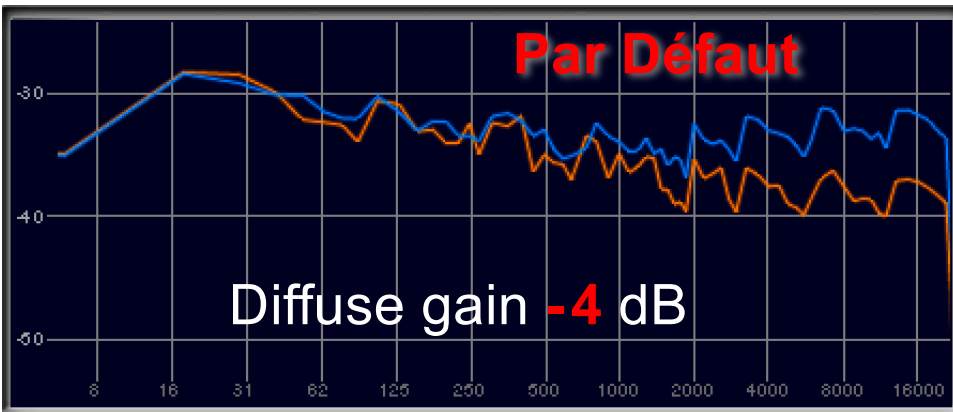
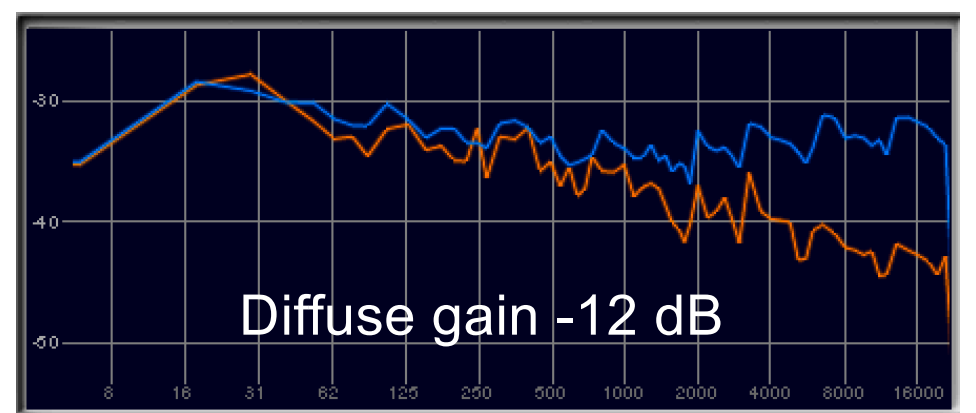
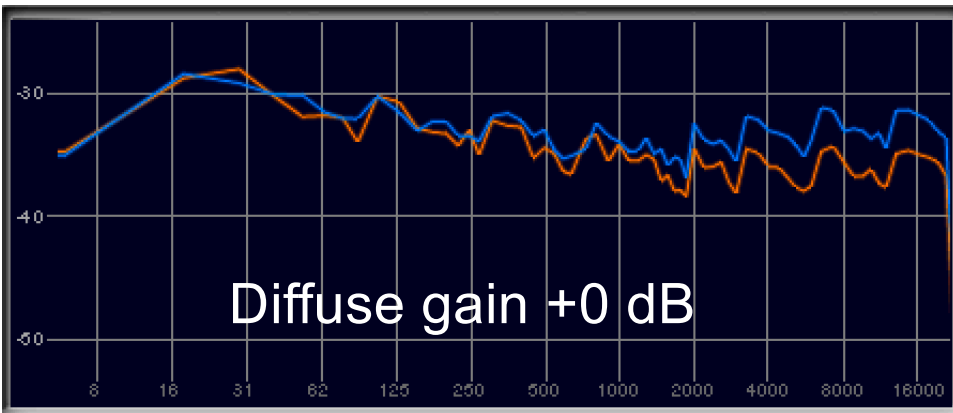
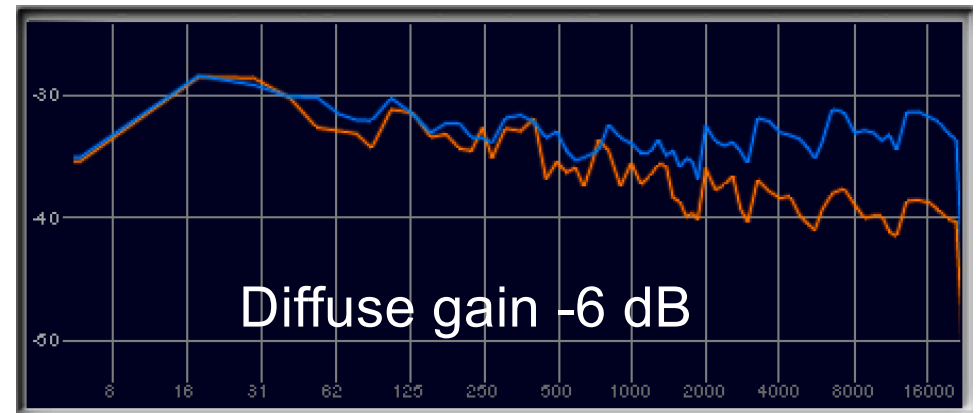
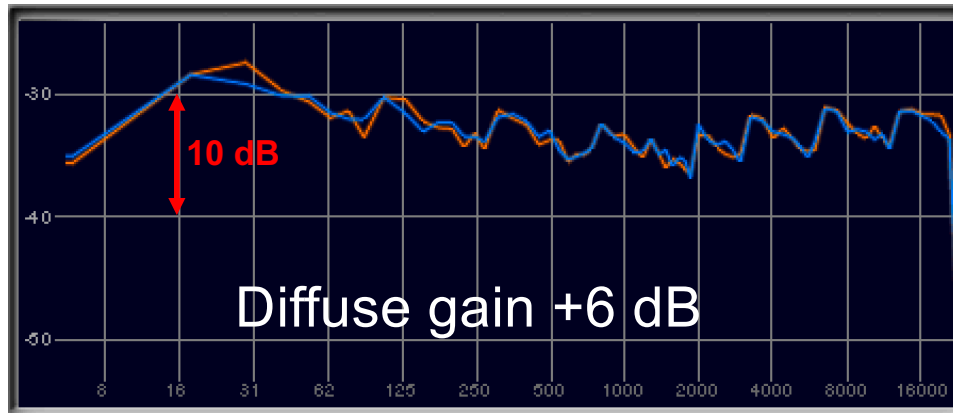


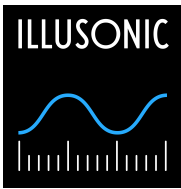
— MESURE (sans De-correlation)  
— BRUIT ROSE (Référence)

Diffuse gain    De-correlation    Room size

0 dB    50

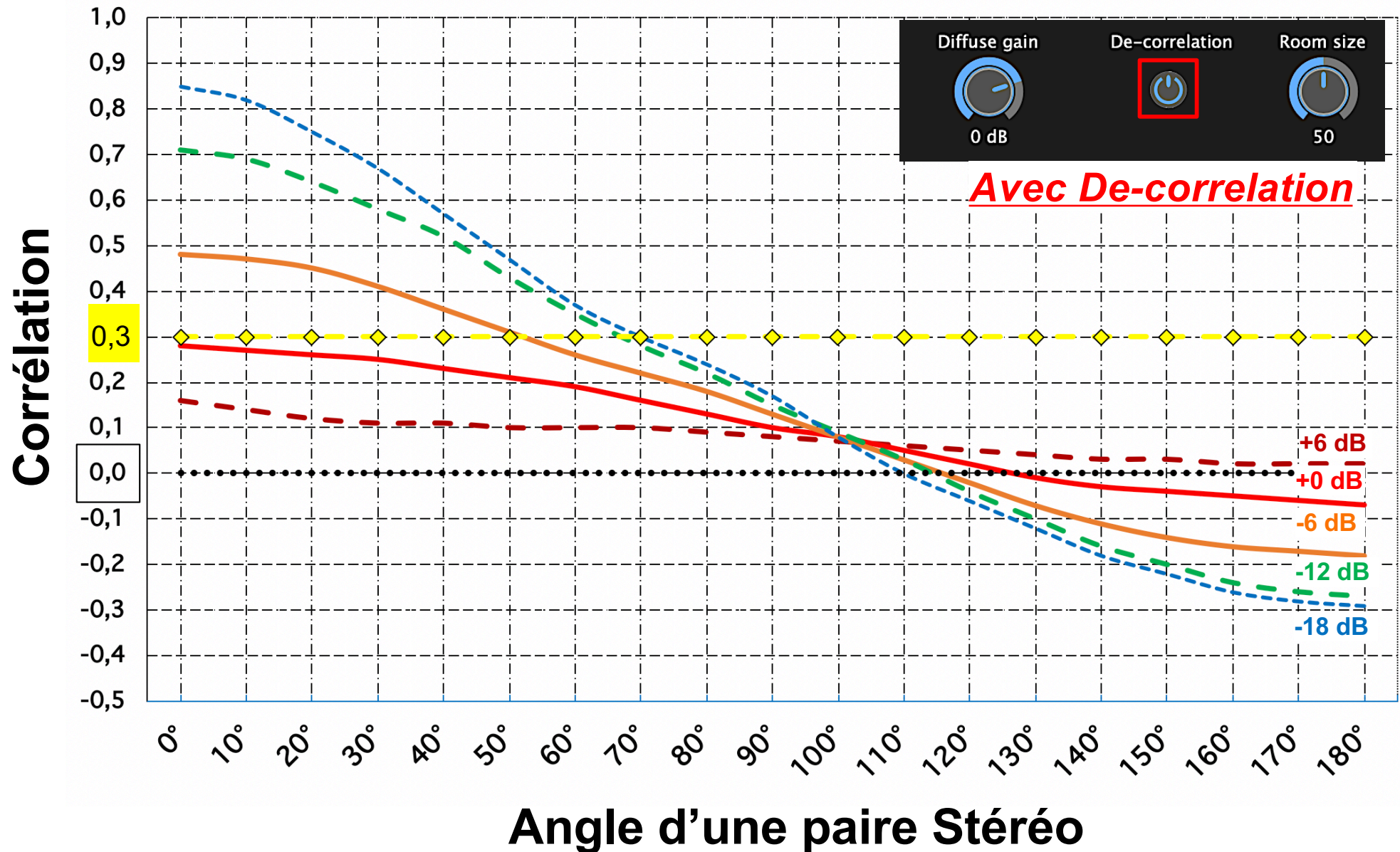
### Sans De-correlation





# Corrélation en fonction de l'angle Stéréo pour des variations du « Diffuse gain »

**Corrélation identique quelque soit la valeur du Focus !**



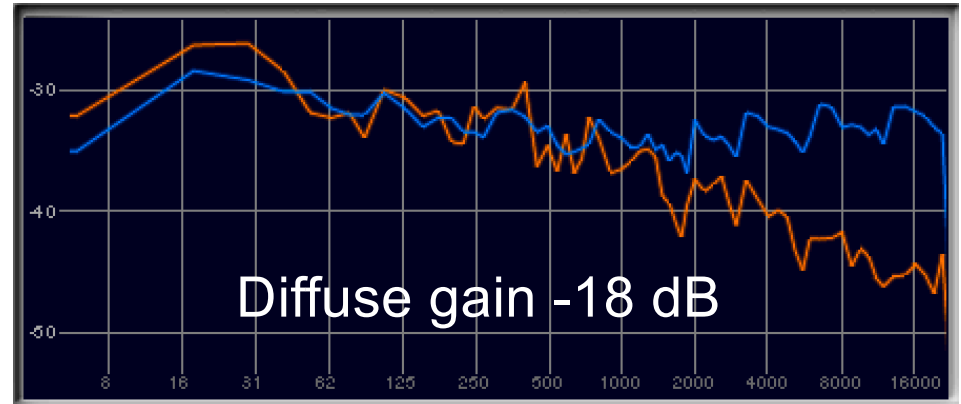
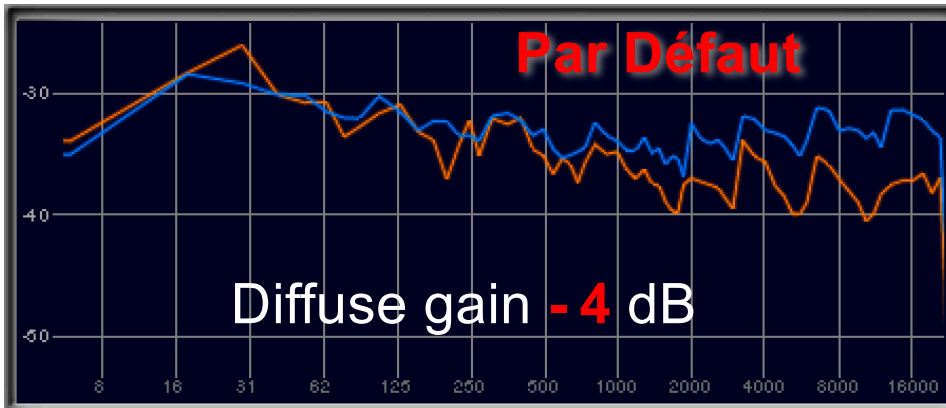
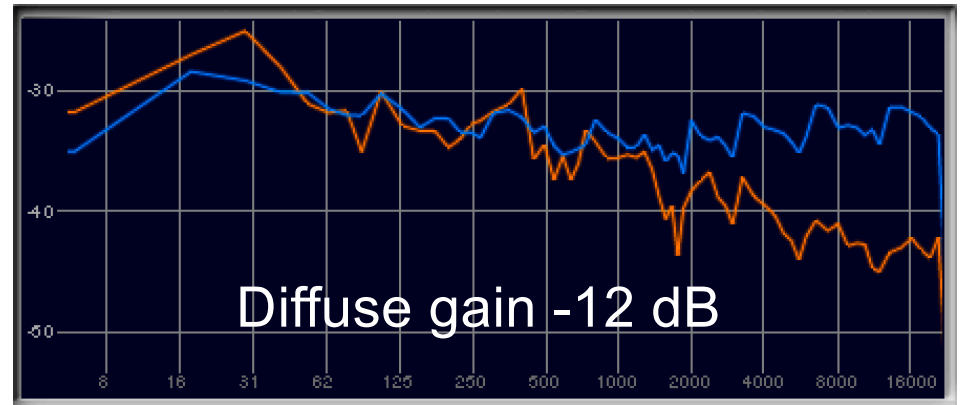
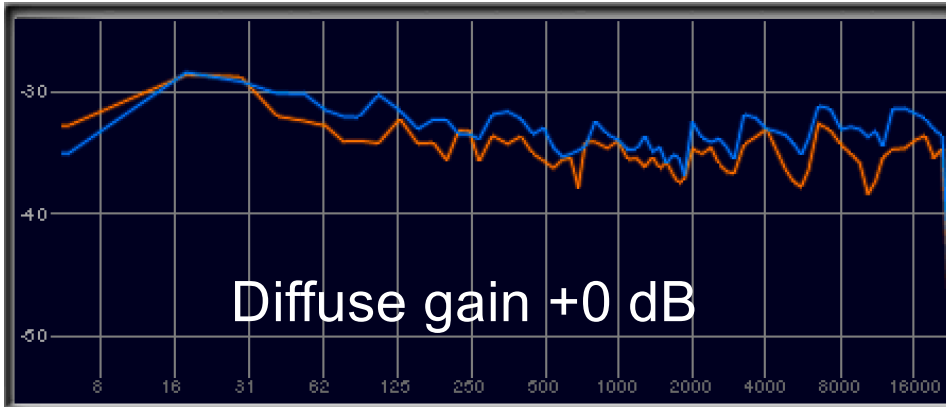
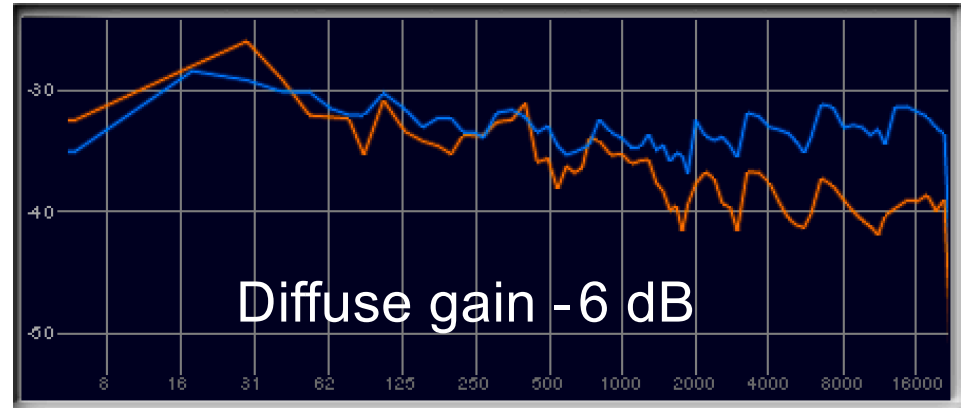
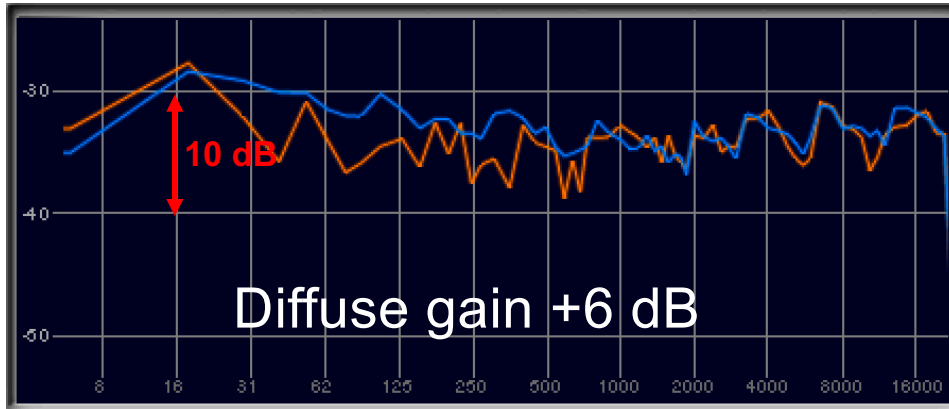


— **MESURE (sans De-correlation)**  
— **BRUIT ROSE (Référence)**

**Avec De-correlation**

Diffuse gain      De-correlation      Room size

0 dB      50





# Manipulation de la composante W en dB sur REAPER

Baisser W de 4 ou 5 dB n'a que peu de conséquence sur la fréquence !!

**1** Formats

- Input format: A-Format
- Microphone distance: Coincident
- Microphone position: normal
- Output format: B-Format AmbiX
- Binaural output:
- A/B-Format Decoder:

**2** MChannelMatrix (1.5.02)

W = -4 dB

INPUT Volumes

1	4.00 dB	0.00 dB	0.00 dB	0.00 dB
2	0.00 dB	0.00 dB	0.00 dB	0.00 dB
3	0.00 dB	0.00 dB	0.00 dB	0.00 dB
4	0.00 dB	0.00 dB	0.00 dB	0.00 dB

SIDE-CHAIN Volumes

1	↓
2	↓
3	↓
4	↓

**3** Decoding

Rotation: 0°

Elevation: 0°

W Signal Bass

Cross-over:

Gain: 0 dB

Frequency: 50 Hz

Order: Linkwitz-Riley 2nd

Invert bass:

Outputs

Center: mute

Front: 0 dB

Wide: 0 dB

Surround: mute

Rear: -10 dB

Front Height: 0 dB

Surround Height: 0 dB

Rear Height: 0 dB

Delay / Shelving: Surround

Delay: off

Frequency: 8.4 kHz

Gain: -2.5 dB

Formats

- Input format: B-Format AmbiX
- Microphone distance: Coincident
- Microphone position: normal
- Output format: Stereo
- Binaural output:

Channel ordering

- Input: W Y Z X
- Output: L R
- Output channel test:

PAZ - Analyser Stereo (Waves)

0.00dB

Center

ILLUSONIC

BRUIT ROSE

6.00

0.00

11.3

6

18

30

42

54

25.1

25.5

1

2

30

40

50

8

16

31

62

125

250

500

1000

2000

4000

8000

16000

© Copyright Illusonic GmbH, Greifensee, Switzerland, 2024. All rights reserved. v5.1.0 - Software expiration: Dec 31, 2025



F8  $=1/(((0,017*(10^{(B2/20)})^2)+(0,1294*(10^{(B3/20)})^2)+(0,2241*(10^{(B4/20)})^2)+(0,2588*(10^{(B5/20)})^2)+(0,2241*(10^{(B6/20)})^2)+(0,1294*(10^{(B7/20)})^2)+(0,017*(10^{(B8/20)})^2))/2)+(((0,017*(10^{(B8/20)})^2)+(0,1294*(10^{(B9/20)})^2)+(0,2241*(10^{(B10/20)})^2)+(0,2588*(10^{(B11/20)})^2)+(0,2241*(10^{(B12/20)})^2)+(0,1294*(10^{(B13/20)})^2)+(0,017*(10^{(B2/20)})^2))/2)$

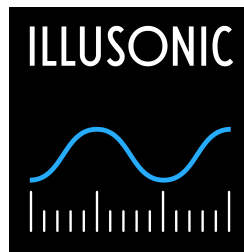
Source à :	Données	Directivité
0° >>	0,0 dB	Index de directivité
30° >>	-0,6 dB	Di = 4,7 dB
60° >>	-2,5 dB	Coef de directivité
90° >>	-6,0 dB	Q = 3,0
120° >>	-12,0 dB	Rapport de capture
150° >>	-18,0 dB	Q^(1/2) = 1,7
180° >>	-25,0 dB	
210° >>	-18,0 dB	
240° >>	-12,0 dB	
270° >>	-6,0 dB	
300° >>	-2,5 dB	
330° >>	-0,6 dB	

**Pour un Cardio :**  
**Q = 3    DI = 4,7**

recueil de normes françaises « acoustique »  
tome 1 NF S31-009 Annexe B 1982 AFNOR

$W = 0$  et  $X-Y-Z = 0$  dB

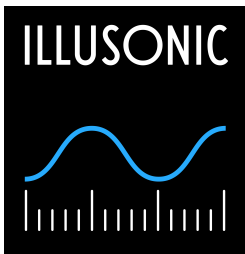
Source à :	Données	Directivité
0° >>	0,0 dB	Index de directivité
30° >>	-1,4 dB	Di = 8,7 dB
60° >>	-10,7 dB	Coef de directivité
90° >>	-27,2 dB	Q = 7,4
120° >>	-23,1 dB	Rapport de capture
150° >>	-15,2 dB	Q^(1/2) = 2,7
180° >>	-14,4 dB	
210° >>	-15,2 dB	
240° >>	-23,1 dB	
270° >>	-27,2 dB	
300° >>	-10,7 dB	
330° >>	-1,4 dB	



**ILLUSONIC**

A/B-Format Decoder v 4.6.0

**Input : 1KHz    FOCUS 100%**



# ILLUSONIC

A/B-Format Decoder v 4.6.0

**Input : 1KHz FOCUS 100%**

*W = -3 dB et X-Y-Z = 0 dB*

Source à :	Données	Directivité
0° >>	0,0 dB	Index de directivité Di = <b>7,3</b> dB
30° >>	-0,9 dB	
60° >>	-6,0 dB	
90° >>	-28,6 dB	Coef de directivité Q = <b>5,3</b>
120° >>	-19,7 dB	
150° >>	-13,9 dB	
180° >>	-12,7 dB	Rapport de capture Q^(1/2) = <b>2,3</b>
210° >>	-13,9 dB	
240° >>	-19,7 dB	
270° >>	-28,6 dB	
300° >>	-6,0 dB	
330° >>	-0,9 dB	

*W = 0 dB et X-Y-Z = 0 dB*

Source à :	Données	Directivité
0° >>	0,0 dB	Index de directivité Di = <b>8,7</b> dB
30° >>	-1,4 dB	
60° >>	-10,7 dB	
90° >>	-27,2 dB	Coef de directivité Q = <b>7,4</b>
120° >>	-23,1 dB	
150° >>	-15,2 dB	
180° >>	-14,4 dB	Rapport de capture Q^(1/2) = <b>2,7</b>
210° >>	-15,2 dB	
240° >>	-23,1 dB	
270° >>	-27,2 dB	
300° >>	-10,7 dB	
330° >>	-1,4 dB	

*W = 0 dB et X-Y-Z = -3 dB*

Source à :	Données	Directivité
0° >>	0,0 dB	Index de directivité Di = <b>9,6</b> dB
30° >>	-1,9 dB	
60° >>	-19,0 dB	
90° >>	-23,4 dB	Coef de directivité Q = <b>9,2</b>
120° >>	-25,3 dB	
150° >>	-16,0 dB	
180° >>	-13,7 dB	Rapport de capture Q^(1/2) = <b>3,0</b>
210° >>	-16,0 dB	
240° >>	-25,3 dB	
270° >>	-23,4 dB	
300° >>	-19,0 dB	
330° >>	-1,9 dB	

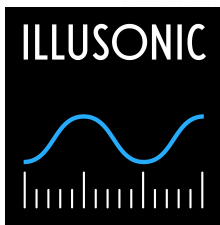
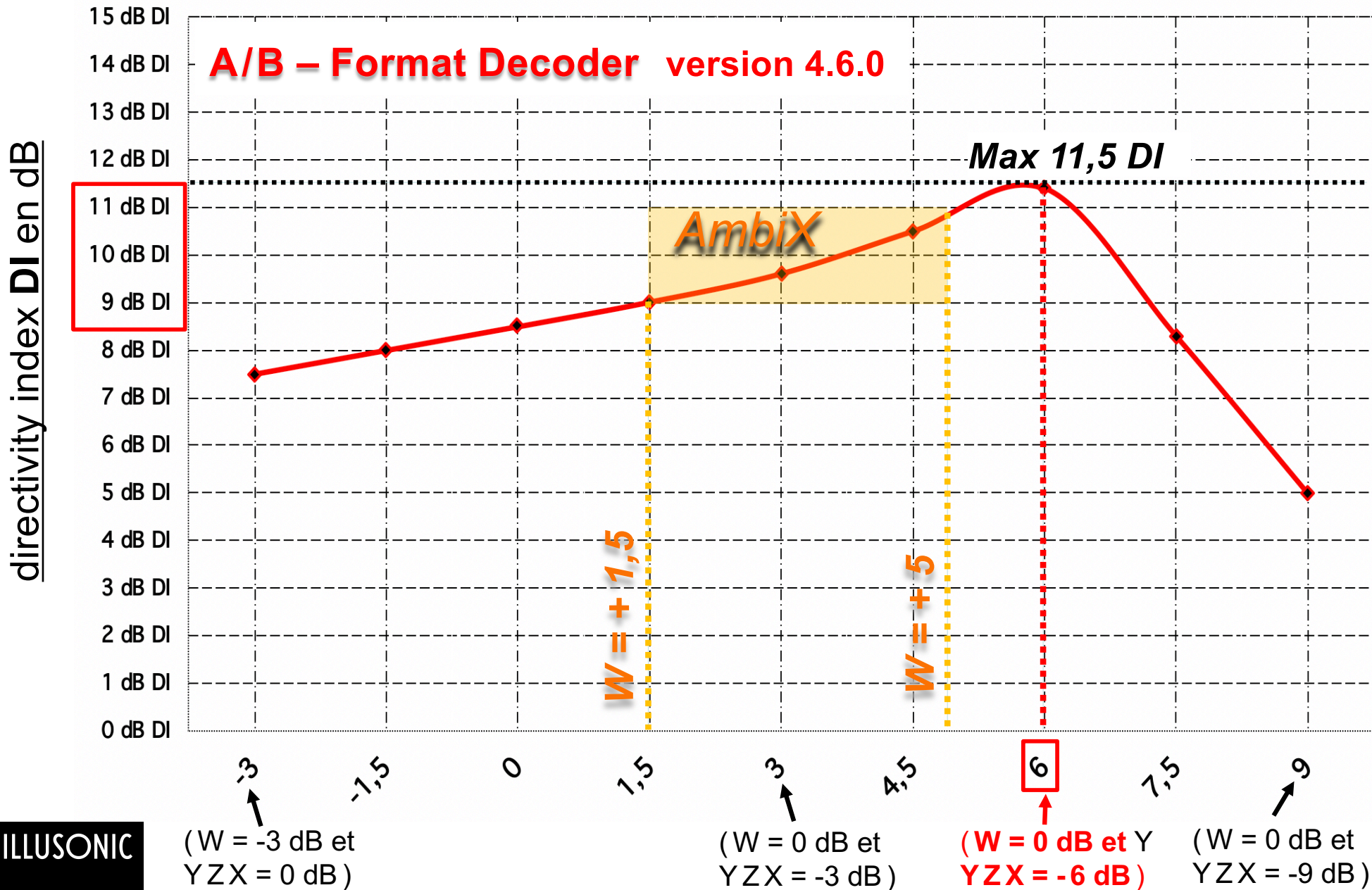
*W = 0 dB et X-Y-Z = -6 dB*

Source à :	Données	Directivité
0° >>	0,0 dB	Index de directivité Di = <b>11,4</b> dB <b>MAX</b>
30° >>	-5,5 dB	
60° >>	-13,9 dB	
90° >>	-16,9 dB	Coef de directivité Q = <b>13,7</b>
120° >>	-30,0 dB	
150° >>	-15,4 dB	
180° >>	-12,5 dB	Rapport de capture Q^(1/2) = <b>3,7</b>
210° >>	-15,4 dB	
240° >>	-30,0 dB	
270° >>	-16,9 dB	
300° >>	-13,9 dB	
330° >>	-5,5 dB	

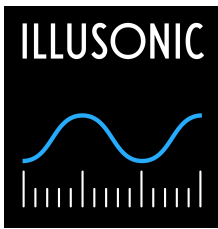
*W = 0 dB et X-Y-Z = -9 dB*

Source à :	Données	Directivité
0° >>	0,0 dB	Index de directivité Di = <b>5,0</b> dB
30° >>	-1,2 dB	
60° >>	-1,8 dB	
90° >>	-10,0 dB	Coef de directivité Q = <b>3,2</b>
120° >>	-18,4 dB	
150° >>	-7,7 dB	
180° >>	-8,9 dB	Rapport de capture Q^(1/2) = <b>1,8</b>
210° >>	-7,7 dB	
240° >>	-18,4 dB	
270° >>	-10,0 dB	
300° >>	-1,8 dB	
330° >>	-1,2 dB	

# POUR UN FOCUS de 100%



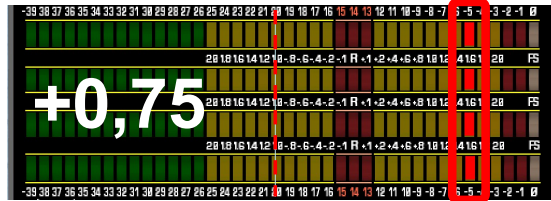
**Manipulation de la composante W en dB**



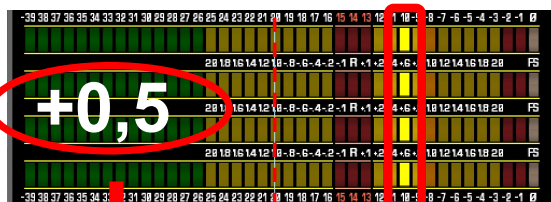
# Corrélation correspondant au B-Format *AmbiX*

**Baisser W de 4 ou 5 dB en B-Format (*AmbiX*) !**

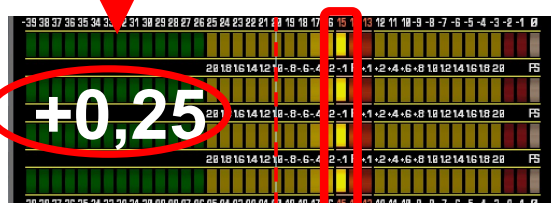
Bruit Rose Corrélé à :



-1 +0 +1



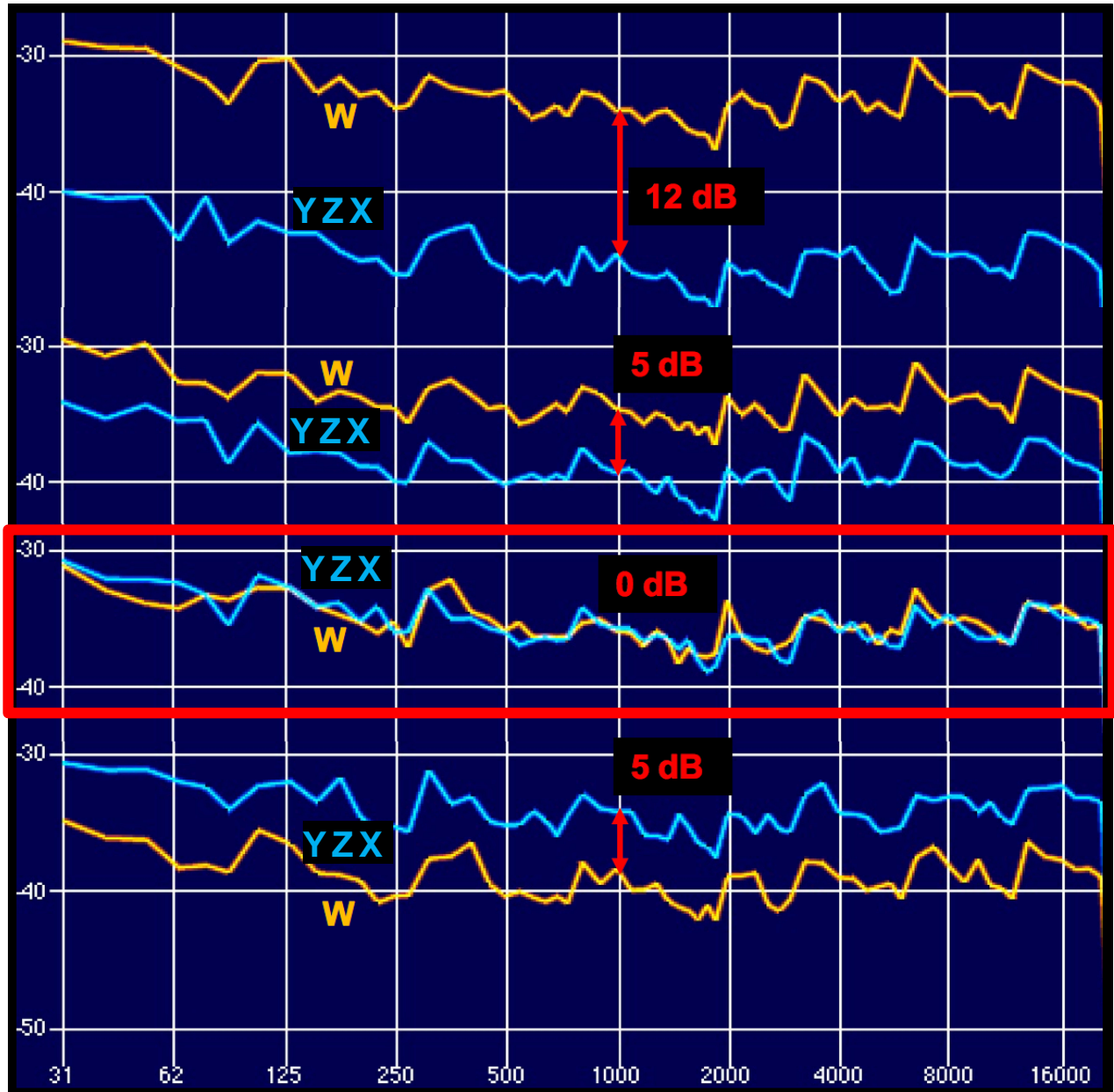
-1 +0 +1



-1 +0 +1



-1 +0 +1



# ILLUSONIC

ILabdecoder

## **Illusonic AB decoder plugin**

**Illusonic GmbH**  
Greifensee, Switzerland

Contact:  
[support@illusonic.com](mailto:support@illusonic.com)

[www.illusonic.com](http://www.illusonic.com)

Information in this document is subject to change without notice.  
All rights reserved.

## Introduction

Our licensable algorithms are used in millions of third party consumer and professional audio products. Examples are audio plug-ins, upmix processors, webcams, tele-conferencing systems, active loudspeakers, soundbars, and TVs.

Over the years sound engineers have frequently shown interest in testing and using some of our algorithms. But we realized that our stand-alone test softwares were not very suitable for this task. So we decided to do free-of-charge *Test Balloon Plug-Ins* for those technologies which generated the most interest from sound engineers. For now, the main purpose is to get feedback and recognition.

The plug-ins are provided as-is, without any warranty or liability from Illusonic GmbH.

**Important!** When using multi-channel, channel ordering is a complex topic. Typically on ProTools (AAX) channel ordering is *Film* and on other workstations (VST) it is *SMPTE/ITU*. Nevertheless, we provide both *Film* and *SMPTE/ITU* on both platforms, due to requests from several power users.

To make verification easier that channel ordering is as intended we added the function *Channel Test*. This is how it works:

1. Select an input or output format.
2. Press *Channel Test* button.
3. Observe whether the test sound goes linearly through the displayed channel list. If yes, all is OK.

## A/B-Format Decoder (ILabdecoder)

The ILabdecoder plug-in is a parametric A-Format and B-Format decoder. Decoding spatial resolution is about as high as third order Ambisonics, without the disadvantages of higher order microphones (low frequency sensitivity, spatial aliasing). Output options are many channel-based formats and binaural.



The beam display shows the effective directivity pattern of the output channels. Directivity can be adjusted with the **Focus** parameters. The **Angle** parameters allow to adjust look-direction of the different channels. Via **Rotation** and **Elevation** the whole sound image look-direction is modified. The **Diffuse** parameters allow to modify direct-to-reverb ratio and degree of inter-channel de-correlation of diffuse sound.

# 1 ILabdecoder - Plugin

The plugin is available in the following plugin formats:

- Mac OS
  - AAX  
64 bit architecture
  - VST3  
64 bit architecture
  - AU  
64 bit architecture

Installation: use the provided installer package.

- Windows
  - VST3  
32 and 64 bit architecture

Installation: unzip the delivery package and copy the plugin file (.vst3) into the folder:

C:\Program Files\Common Files\VST3

- Linux
  - LV2  
64 bit architecture

Installation: unzip the delivery package and copy the plugin into the plugin folder.



The parameters are described in *Illusonic\_Plugins.pdf*.



## 2 ILabdecoder - Changelog

### Version 1.10.0: December 04, 2019

- add Windows VST3 versions

### Version 1.11.0: December 19, 2019

- add 'cube' and 'cube+center' formats
- improvements for LFE (add 10 dB overhead / add 'mute' state / fix slider)
- fixes and improvements for HRTF handling

### Version 1.12.0: February 21, 2020

- redesign plugin interface
- add AU version
- sundry gui and usability updates

### Version 1.13.0: June 02, 2020

- new Test Channels function to conveniently check input and output multi-channel channel ordering
- changed channel ordering of Cube format to 5.1 (C and LFE are silent)
- added new Cube & Center & Sides format
- many other small changes, additions and improvements

### Version 1.14.0: August 27, 2020

- improve beam display
- improve output mixing processing
- many other small changes, additions and improvements
- add "notarization" for all Mac versions

### Version 2.0.0: December 14, 2020

- add new input format "A.1-Format", which is A-Format plus an omni mic signal
- variable crossover for use of A.1-Format's .1 signal
- remove hard limiter at 0dB output

### Version 2.1.0: March 05, 2021

- fix issue where Reaper could use only 8 channels

### Version 2.2.0: March 15, 2021

- fix issue where signals were not properly muted

### Version 2.3.0: March 31, 2021

- improve diffuse sound handling, this improves overall quality significantly
- change default parameters to be more optimal for all formats
- improve binaural rendering and add headphones equalization

### Version 2.4.0: July 28, 2021

- add A-Format as output option
- new output format: Cube + Center + Side + Rear

### Version 2.4.1: July 28, 2021

- improve signal processing performance

#### **Version 2.5.0: October 18, 2021**

- bug fixes in algorithm core
- change "Cube+Center+Side+Rear"
  - edit handling of surround channels "..+Back"
  - put rear center to LFE position
- macOS: add support for 'arm64' (based on beta version of AAX SDK)

#### **Version 2.6.0: December 09, 2021**

- fix issue where 2nd and 3rd order B-Format output formats were disabled
- fix issue with the new cube+center+side+back format

#### **Version 2.6.1: December 10, 2021**

- fix for v2.6.0

#### **Version 2.7.0: April 13, 2022**

- enable bass panel for all input formats (= bass management with W signal as bass)
- cube formats: fixed bug with channel type assignments
- now show expiration date in plug-in
- set expiration date to end of 2023

#### **Version 2.8.0: May 25, 2022**

- add level meters for input and output audio
- add 'alt' versions for 7.1.x audio formats

#### **Version 2.8.1: June 01, 2022**

- enable downgrading in all plugin installers
- Mac: add support for systems down to MacOS 10.10

#### **Version 3.0.0: September 30, 2022**

- major update
- new decorrelator
- improved binaural rendering

#### **Version 3.0.1: November 10, 2022**

- bug fix for beam display

#### **Version 3.0.2: April 05, 2023**

- bug fixes (cross-over range mapping / memory leak / parameter initialisation / numerical issue)
- code improvements (algorithm release / responses plot / header sync)

#### **Version 4.0.0: August 16, 2023**

- improved audio quality (more reactive)
- replaced de-correlators for diffuse sound
- fixed channel ordering bugs in a couple of output formats
- move to latest version of JUCE framework, including support for new Pro Tools formats

- now all plugins can connect input and output to any bus sizes
    - if input bus has more channels than input format, additional channels are ignored
    - if input bus has fewer channels than input format, missing channels are set to zero
    - if output bus has more channels than output format, additional channels are set to zero
    - if output bus has fewer channels than output format, then only the number of available channels are given out
- ⇒ as a consequence of this, plugins now always show all input and output formats, disregarding bus choice

#### **Version 4.0.1: August 18, 2023**

- fix parameter attachment in core

#### **Version 4.0.2: September 04, 2023**

- fix channel ordering for AAX and VST3 after changes in dev environment core

#### **Version 4.1.0: October 30, 2023**

- add 'microphone position' parameter
- add look-ahead in analysis
- implement automation shortcut for Pro Tools ("ctrl-alt-command click" to open "add to automation" dialogue)
- improve automation string listing

#### **Version 4.3.0: December 21, 2023**

- add 9.1.x loudspeaker formats
- add new control 'Wide' for angle, focus and gain
- add new control 'Rear Height' for angle, focus and gain
- add separate 'azimuth' and 'elevation' controls
- fix for VST3 compatibility with JRiver Media Center

#### **Version 4.4.0: February 16, 2024**

- fix 9.1.x channel ordering
- fix bug in initialisation (which lead to problems with channel order)

#### **Version 4.5.0: March 08, 2024**

- now all channels (except L, R, C) feature separate delay and high shelving filters
- fixed output channel gains for binaural (now also applied on diffuse sound)
- fixed 'mute' on new channels (front height, front wide, rear height) (before 'mute' was -20 dB)
- fixed initialization of delay/shelving, could sometimes cause issues

#### **Version 4.6.0: March 27, 2024**

- fix numerical issue that caused audio artefacts in certain low-level signal conditions

#### **Version 5.1.0: December 06, 2024**

- microphone choice (mm between capsules) dialogue is now also enabled for B-Format
- new formats B.1 FuMa und B.1 AmbiX, to use an omni mic for bass
- decorrelator is now disabled by default

Merci de votre attention

Site : <https://www.lesonbinaural.fr>

Mail : **b.lagnel@gmail.com**