助聽器進階信號處理

李沛群 (Li Pei-Chun)
馬偕醫學院 聽力暨語言治療學系
2015/11/15

報告大綱

- 數位助聽器的信號處理
- 問題描述與解決方案
  - 方向性麥克風
  - 噪音消除
  - 回饋音消除
- 信號處理的效益與副作用
  - 如何展示
- 還有哪些問題 (需求)

數位助聽器的信號處理

- 甚麼是信號處理
- 為什麼要做信號處理
- 助聽器的信號處理分為兩大類
  - 聽損補償
    - WDRC、ADRO、移頻處理
  - 環境控制
    - 方向性麥克風、噪音消除、風切聲消除、回饋音消除
- 用 Goldwave 模擬助聽器

是否能察覺語音共振峰特徵?

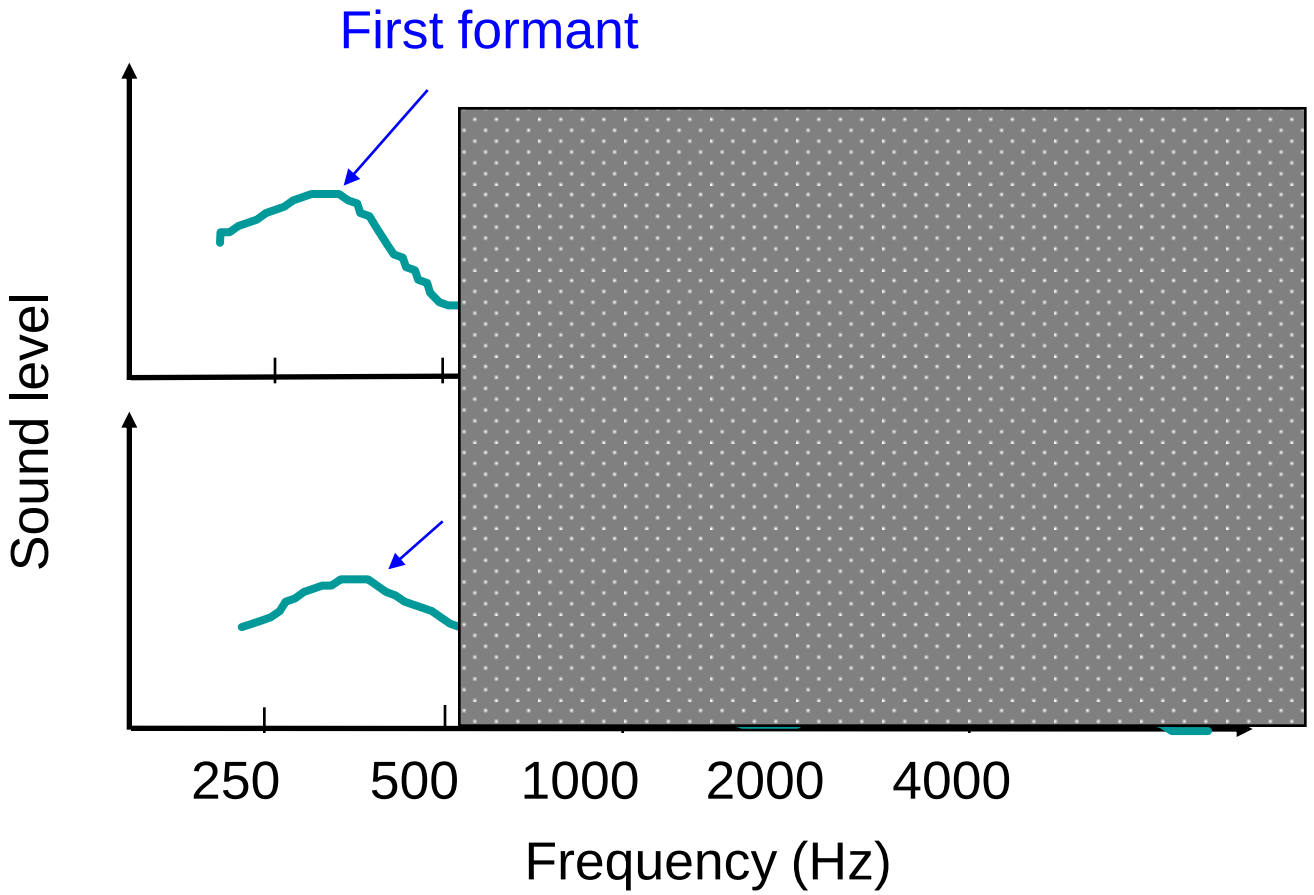
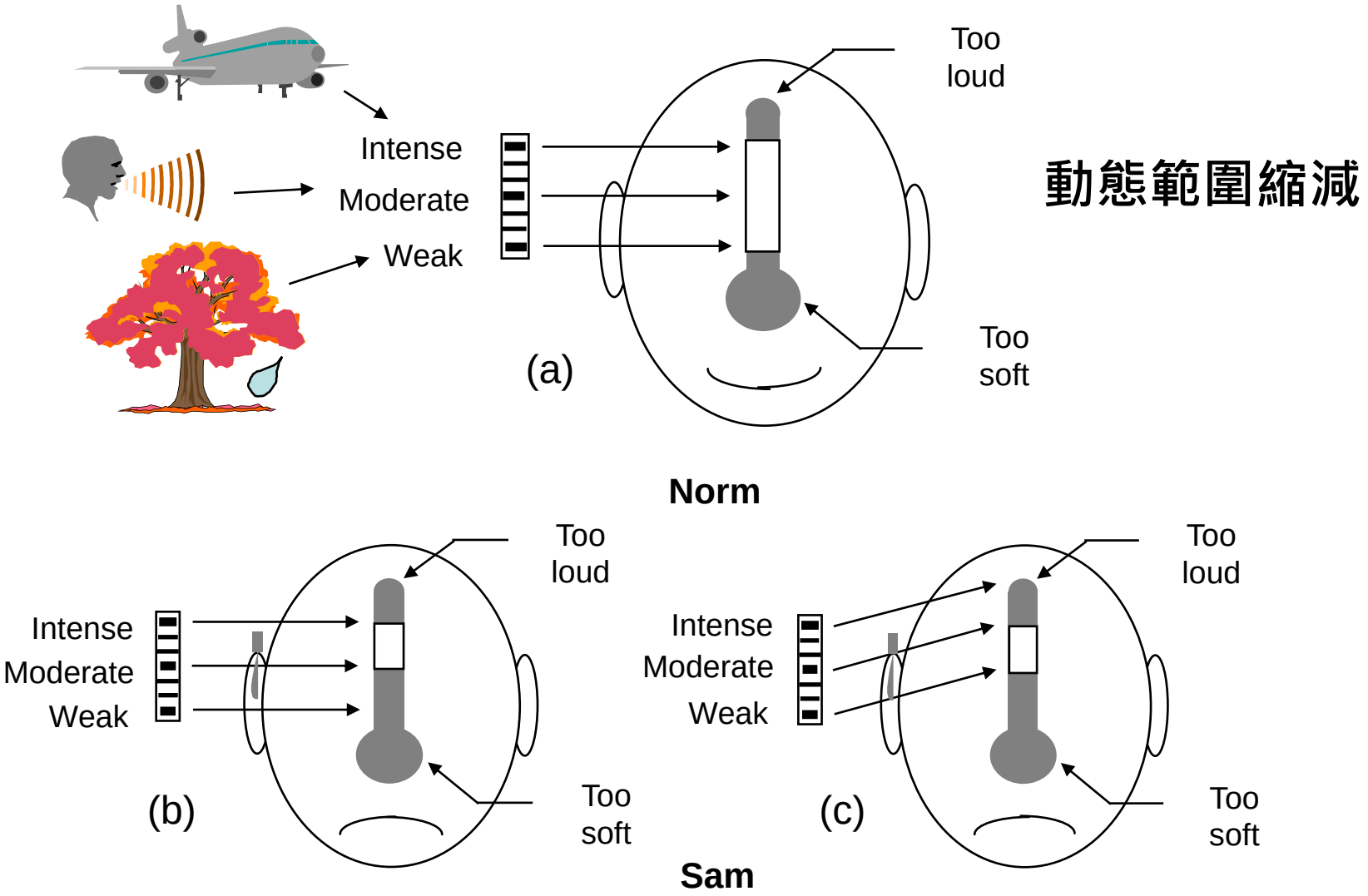


Figure 1.1 Similarity of the two vowels *oo* and *ee* when the second formant is inaudible because of hearing loss (grey area).

Source: Dillon (2001): *Hearing Aids*



動態範圍縮減

Figure 1.2 The relationship between the dynamic range of sounds in the environment and the dynamic range of hearing for: (a) normal hearing, (b) sensorineural hearing loss without amplification, and (c) sensorineural hearing loss with a constant amount of amplification for all input levels.

Source: Dillon (2001): *Hearing Aids*

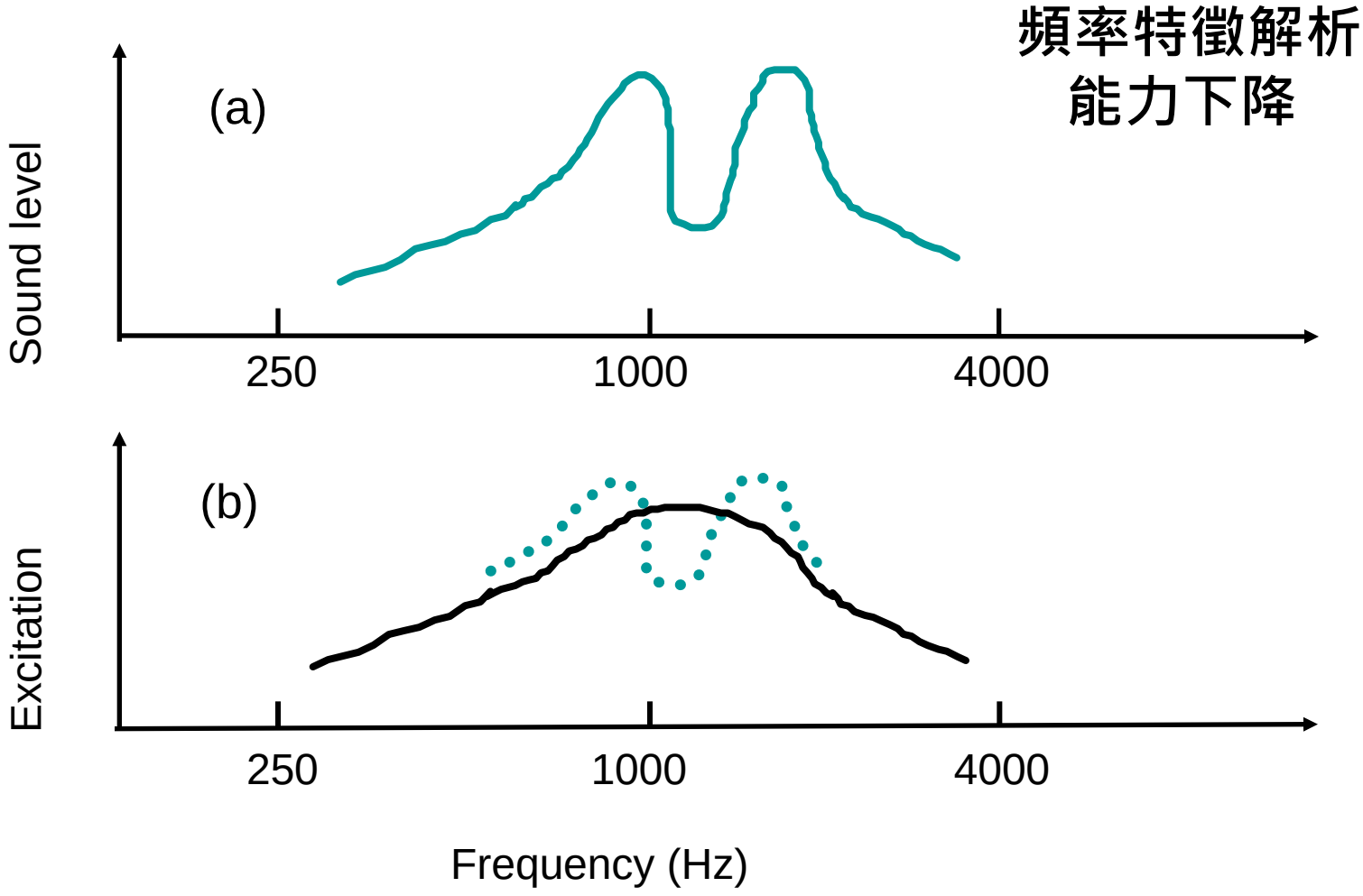
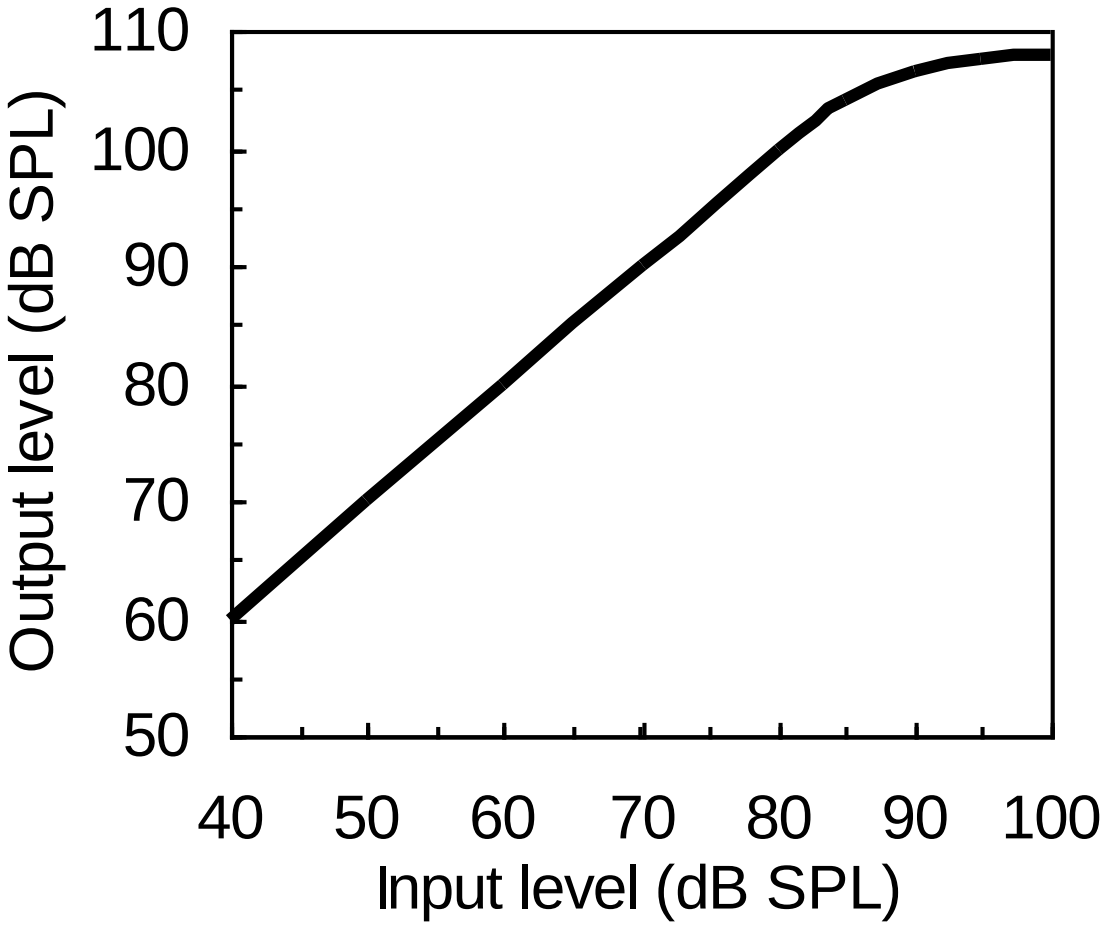


Figure 1.3 (a) Sound spectrum, and (b) representation in the auditory system for normal hearing (dotted line) and sensorineural hearing impairment (solid line).

Source: Dillon (2001): *Hearing Aids*



根據聽力損失
設計輸入輸出
轉換方式

Figure 1.4 Input-output diagram for a hearing aid with 20 dB gain, showing how the output SPL depends on the input SPL, for a particular signal or frequency.

Source: Dillon (2001): *Hearing Aids*

以上為設計聽損補償信號處理的指引概念範例

- 環境控制部分，主要以提高信噪比為目標
  - 方向性麥克風
  - 噪音消除
  - 回饋音消除: 因為回饋路徑的存在，造成有可能出現嘯叫聲 (嘯叫聲 即為噪音)
  - 關於聽損補償的需求描述，沒有太多著墨在有噪音的情況下該如何
  - 已知對噪音干擾的耐受性下降
    - 但是純粹提升信噪比，是否就足夠?
    - WDRC 是非線性的放大，不容易評估處理後的信噪比

方塊圖使用的符號

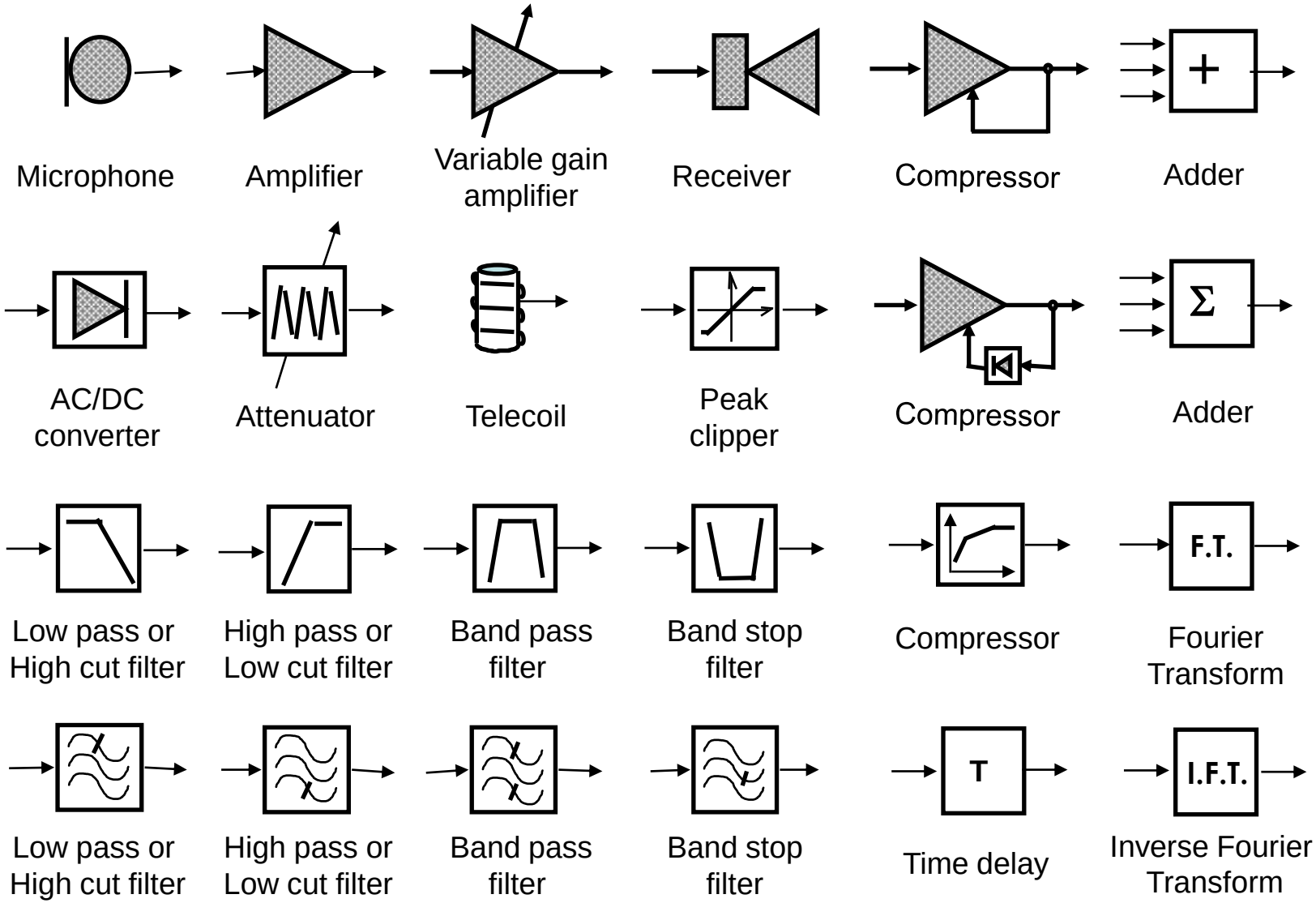


Figure 2.1 Symbols used in block diagrams.

Source: Dillon (2001): *Hearing Aids*

助聽器方塊圖範例

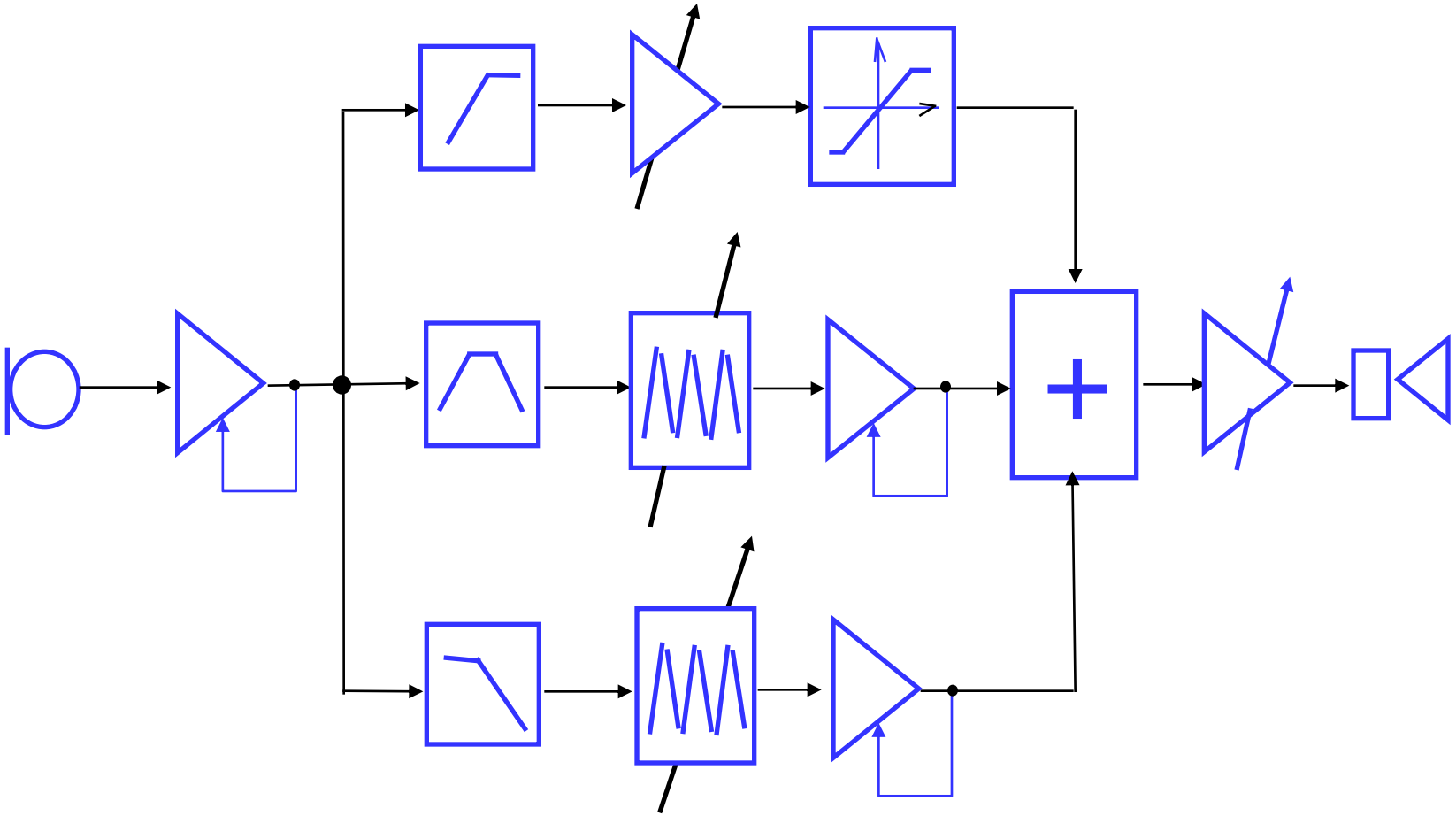


Figure 2.2 A three channel compression hearing aid.

Source: Dillon (2001): *Hearing Aids*

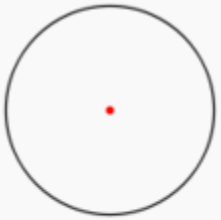
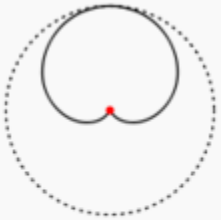
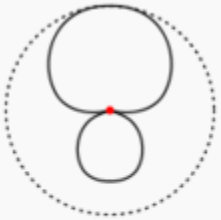
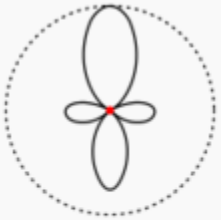
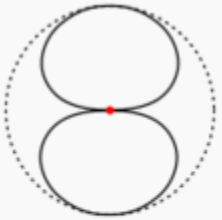
方向性麥克風

噪音的例子:

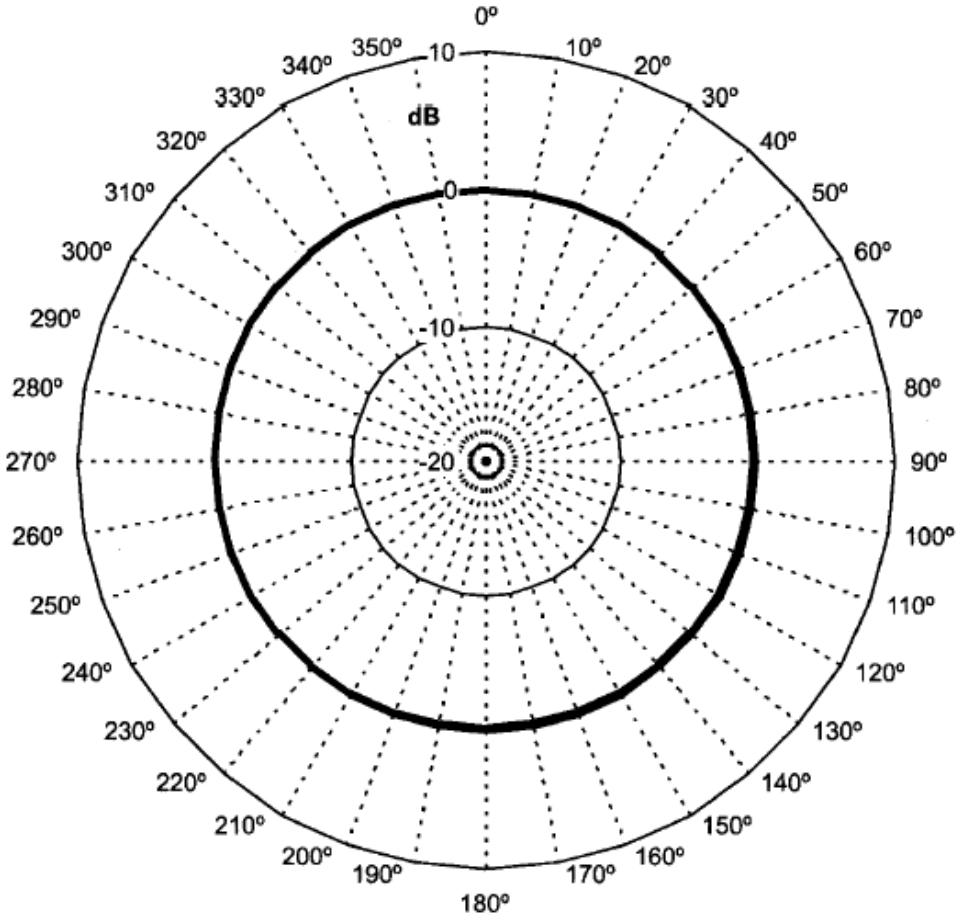
- 汽車噪音
- https://www.youtube.com/watch?v=X5D1ToZ6igs

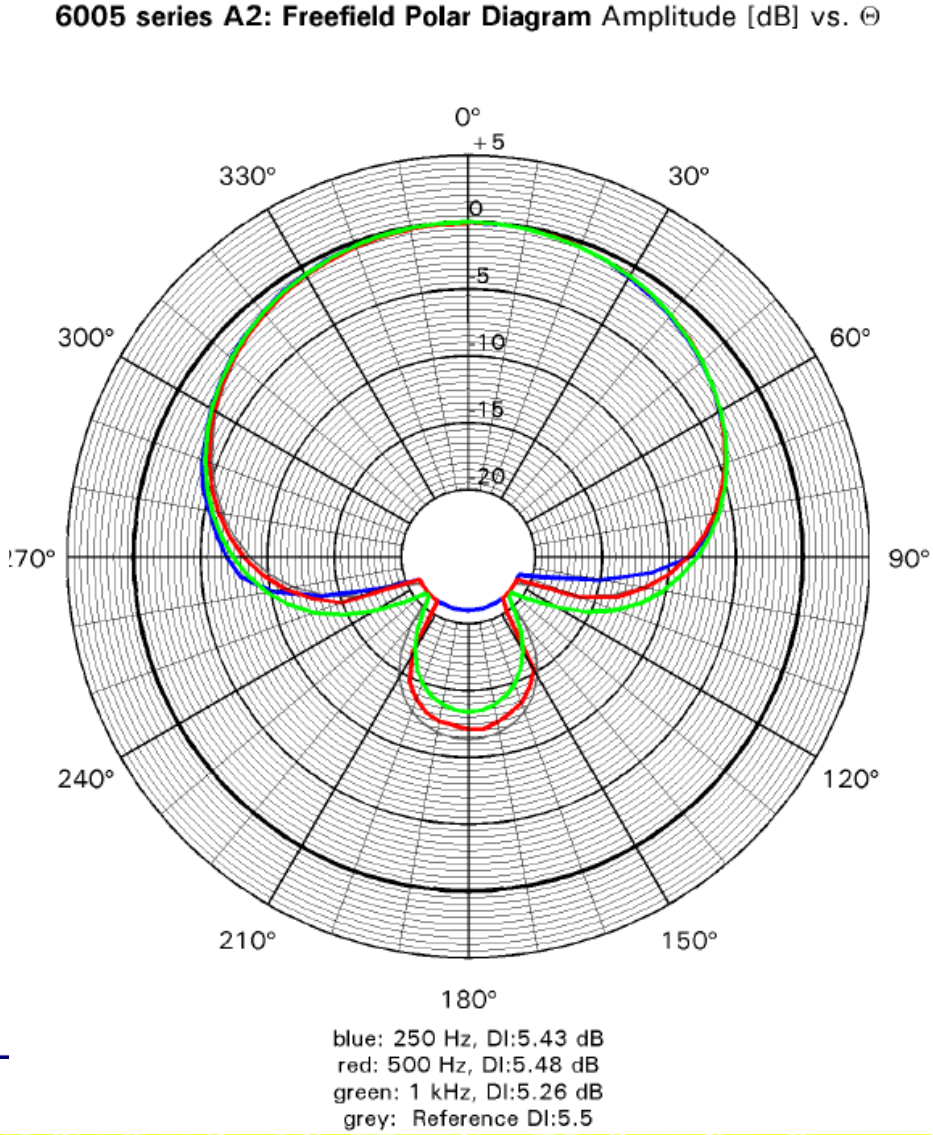
- 風扇噪音
- https://www.youtube.com/watch?v=qorkD6nPYQM

方向性麥克風的指向性

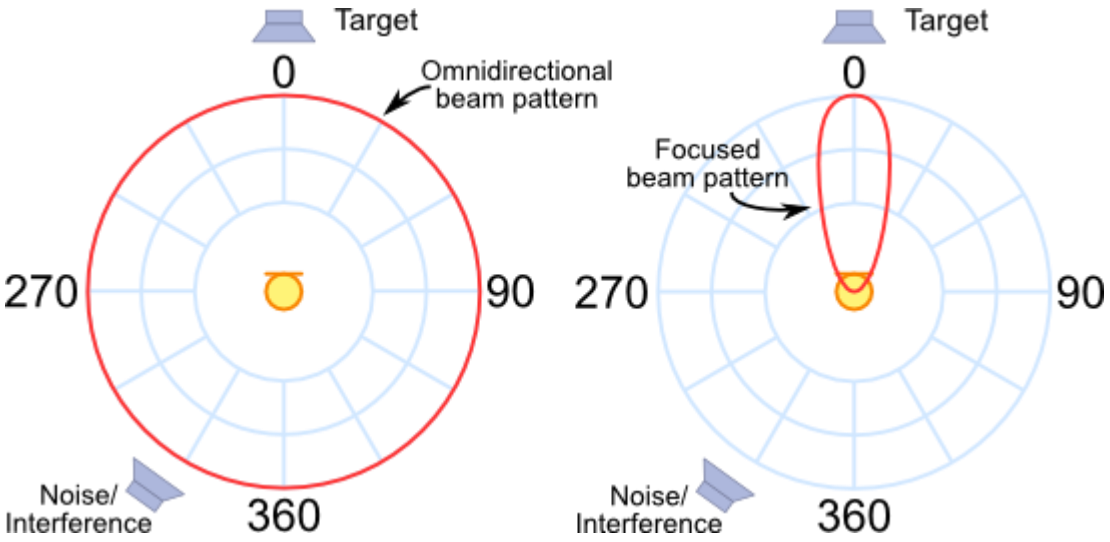
				
Omnidirectional 全指向式	**Cardioid** 心型指向	**Hypercardioid** 超心型指向	**Shotgun** 槍型指向	**Bi-directional** 雙指向式

麥克風的指向性

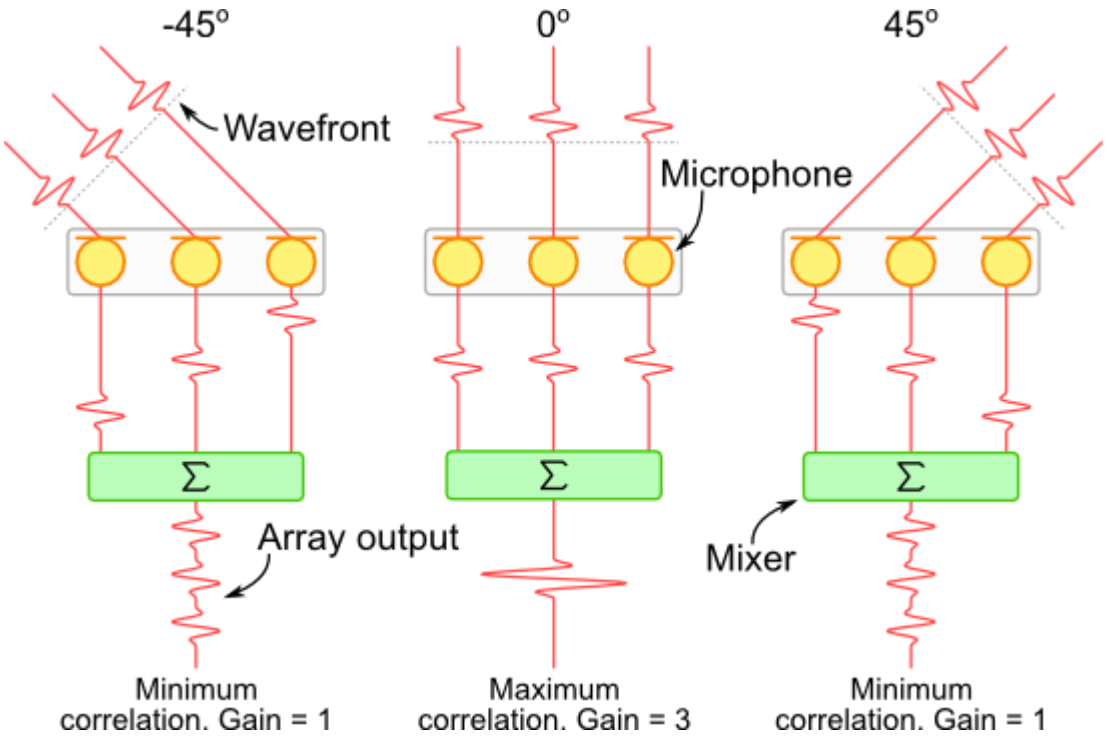

0°
10°
20°
30°
40°
50°
60°
70°
80°
90°
100°
110°
120°
130°
140°
150°
160°
170°
180°
190°
200°
210°
220°
230°
240°
250°
260°
270°
280°
290°
300°
310°
320°
330°
340°
350°
dB
-10
0
-10
-20


6005 series A2: Freefield Polar Diagram Amplitude [dB] vs. Θ
0°
+5
30°
60°
90°
120°
150°
180°
210°
240°
270°
300°
330°
0
5
10
15
20
blue: 250 Hz, DI:5.43 dB
red: 500 Hz, DI:5.48 dB
green: 1 kHz, DI:5.26 dB
grey: Reference DI:5.5

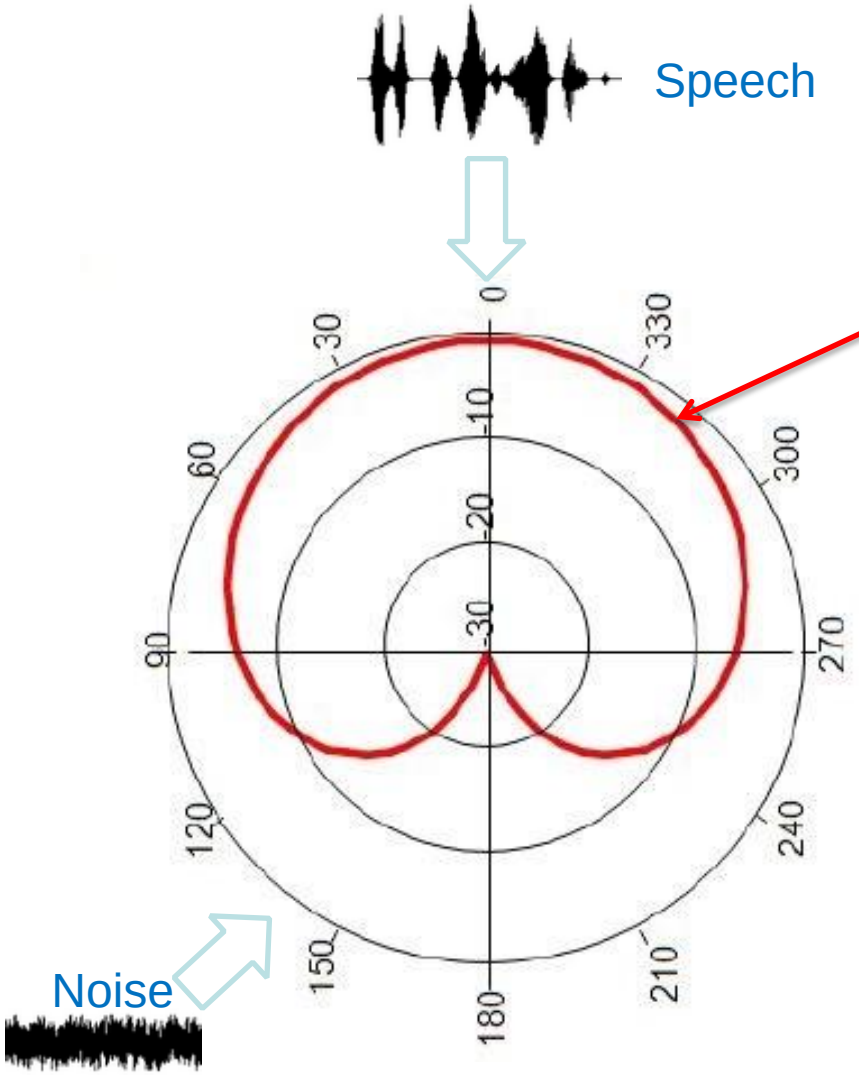
固定式方向性


Target
0
Omnidirectional
beam pattern
270
90
Noise/
Interference
360
Target
0
Focused
beam pattern
270
90
Noise/
Interference
360

工作原理

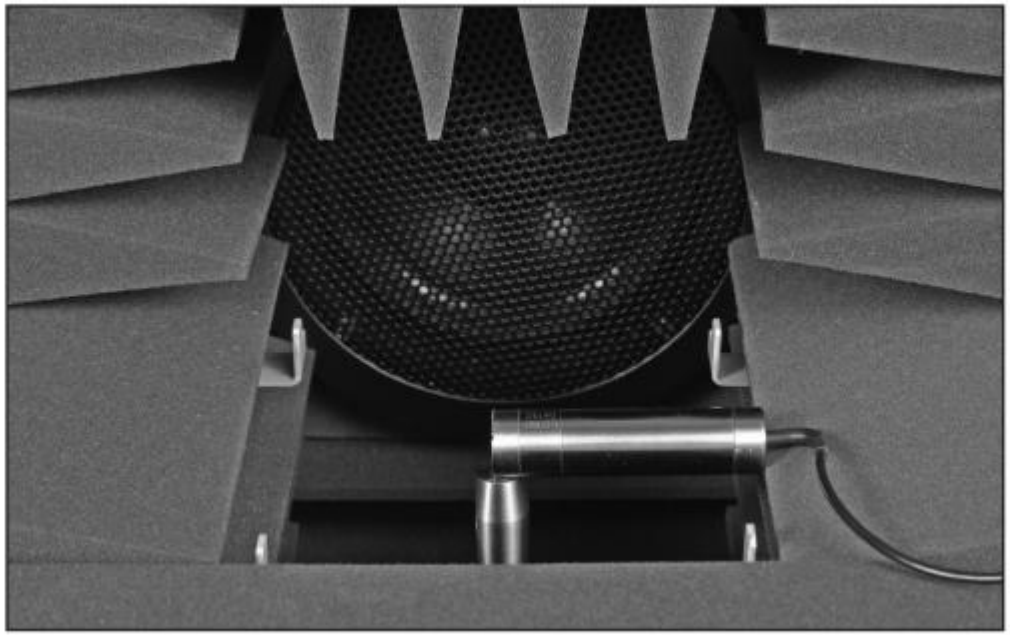

-45°
0°
45°
Wavefront
Microphone
Σ
Σ
Σ
Array output
Mixer
Minimum
correlation. Gain = 1
Maximum
correlation. Gain = 3
Minimum
correlation. Gain = 1

方向性麥克風 - 極相圖量測 (Fonix 8000)



Improve SNR performance.

方向性量測步驟一:



方向性量測步驟二：

Figure 6.3.1A: The BTE Rotating Shaft and Positioning Saddle

Figure 6.3.1B: BTE hearing aid positioned on Rotating Shaft and Positioning Saddle

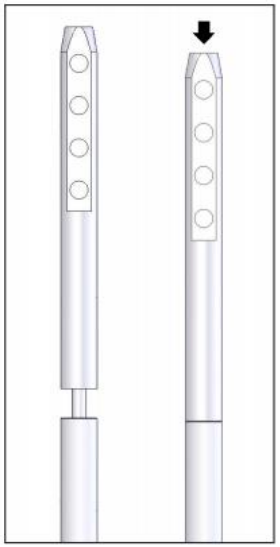
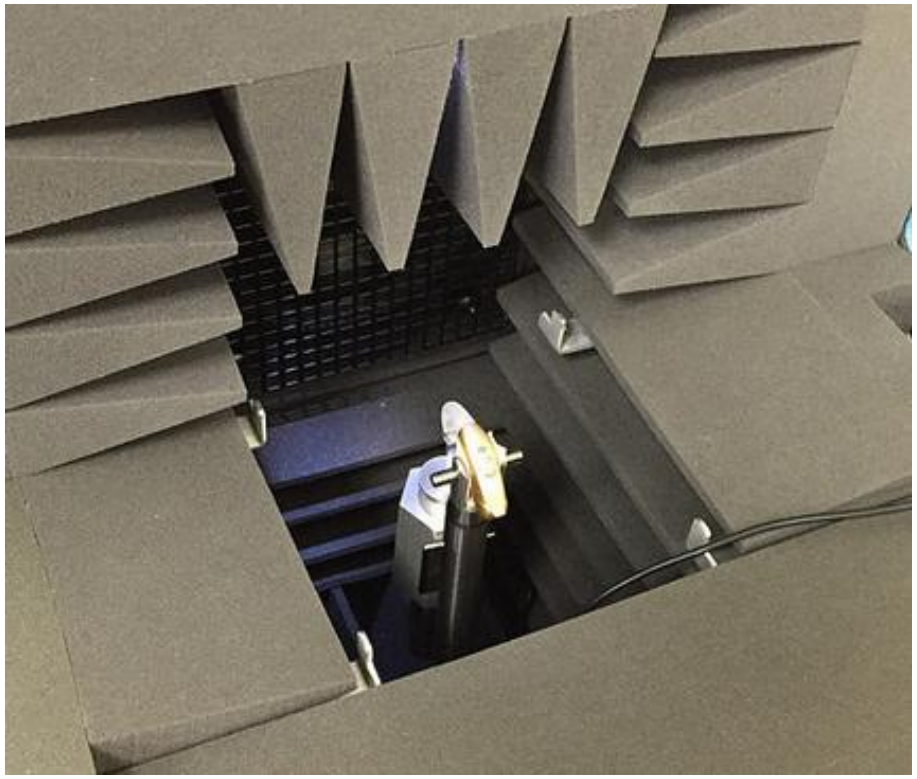
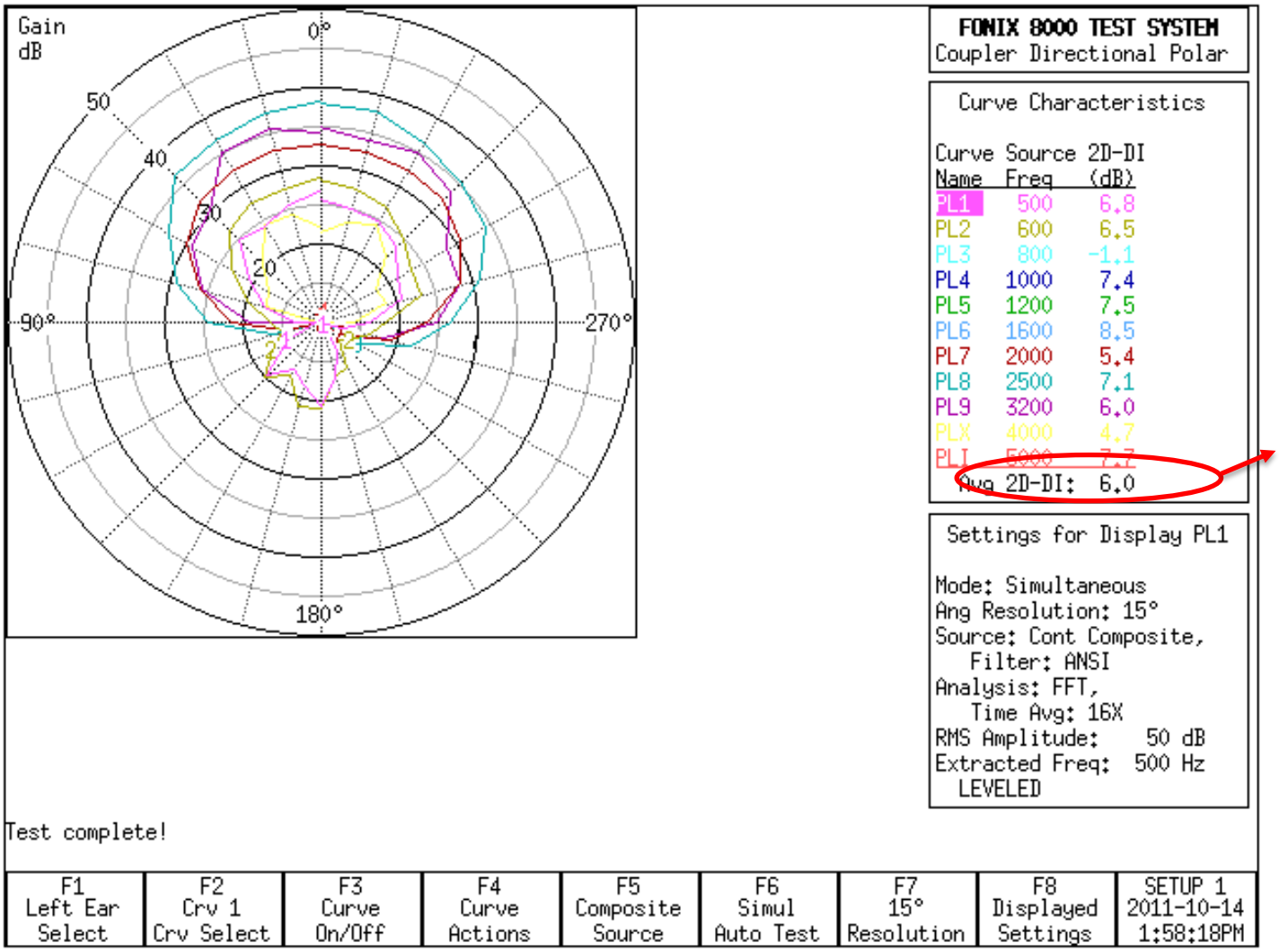


Figure 6.3.1C: Fully insert the shaft into the sound chamber



展示:

方向性量測步驟三:



Directivity Index (2D-DI)

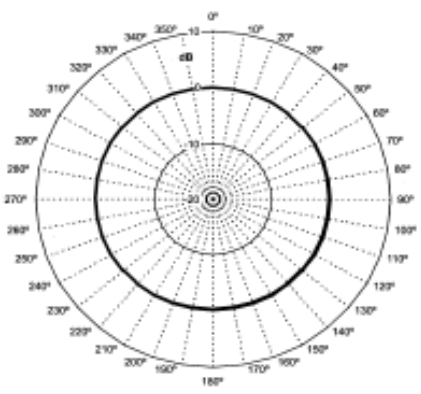
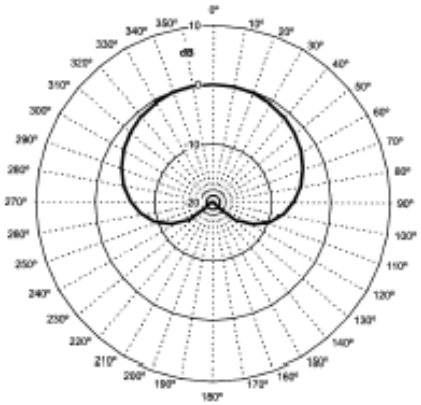
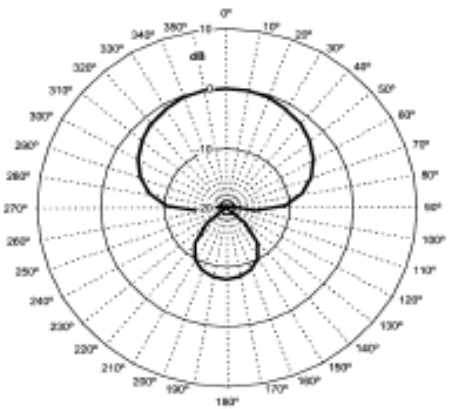
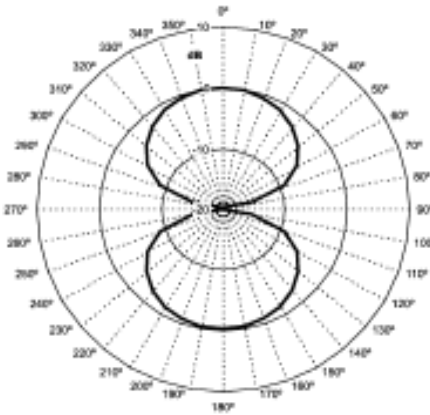
2D-DI: Two dimensional directivity

$$D = 10Log_{10}Q$$

$$Q = \frac{4\pi|M_{ax}|^2}{\int_0^{2\pi}\int_0^{\pi}|M(\alpha,\theta)|^2|\sin\theta|d\theta d\alpha}$$

where $|M_{ax}|^2$ is the magnitude of the mean-square sensitivity to sound waves that arrive at the device along the chosen reference axis, and $|M(\alpha,\theta)|^2$ is the mean- square sensitivity to sound waves that arrive at the device from an elevation angle α and an azimuth angle θ.

非固定(自動調整)方向性

- 自動由全向性轉為方向性，也可以追蹤聲源改變指向
- 判斷不正確，或是沒有考慮身體、頭部、耳廓的效應，有可能讓聲音品質更糟

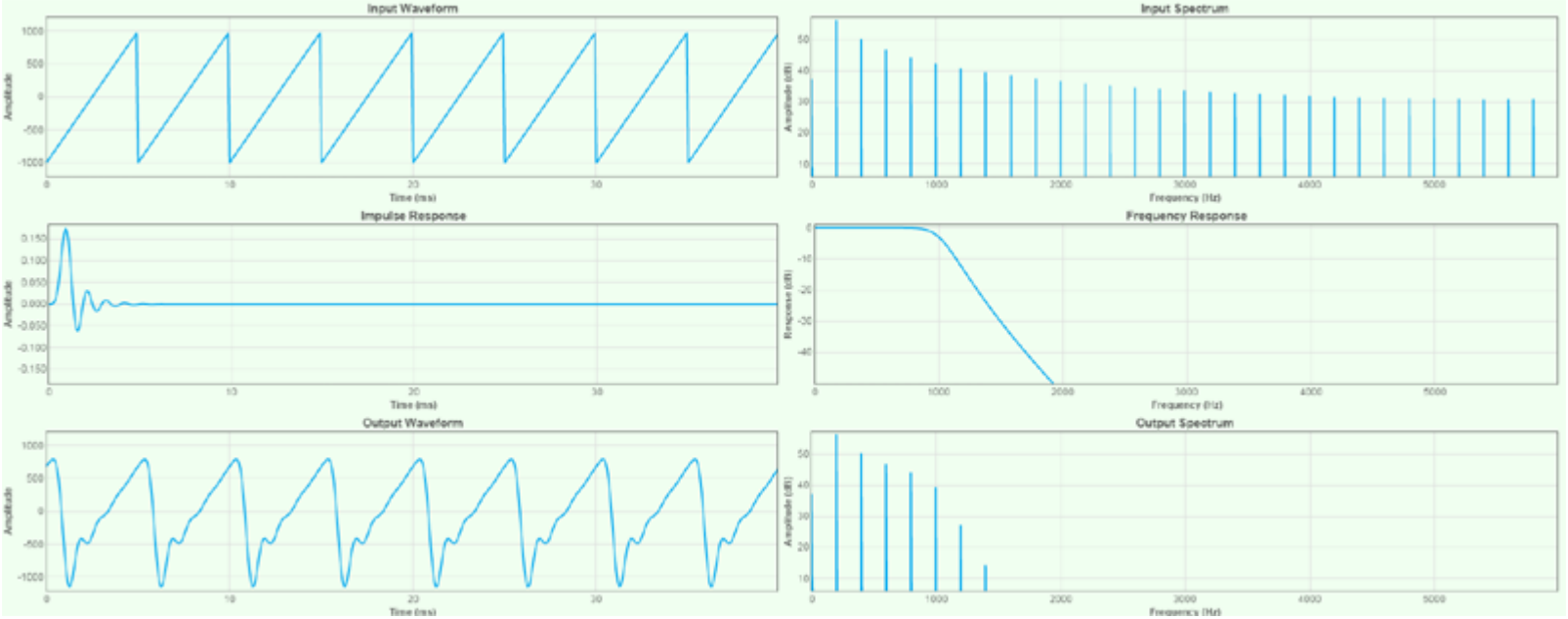
噪音消除

噪音消除問題的描述

- 助聽器收到的信號是 (想聽到的信號) 混合 (噪音) $y(n) = x(n)+v(n)$
  - 想聽到的信號: 說話的聲音、音樂 $x(n)$
  - 噪音: 不想聽到的信號。例如風扇聲音、馬達聲音等等 $v(n)$
    - 混合在這裡用的是 “相加” 來表示
  - 如果噪音的特性是
    - 平均值為 0 的隨機信號，像是 白噪音 (Goldwave)
    - 而且與想聽到的聲音信號是不具有相關性 (uncorrelated)
  - 那麼我們有沒有辦法從觀察到的$y(n)$ 估計出 $x(n)$

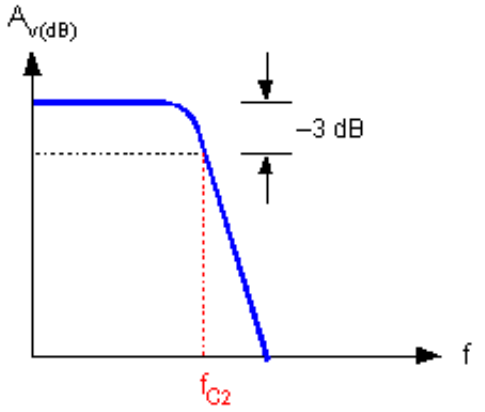
噪音消除 - 當想要聽到的聲音和噪音分布在不同頻帶

- 如果噪音的頻譜，和想要聽的信號的頻譜，是分開的
  - 直接使用濾波器，進行濾波，就可以把噪音消除

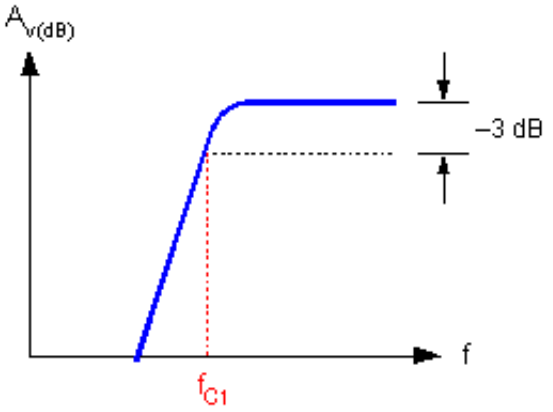


Internet institute for speech and hearing
http://www.speechandhearing.net/laboratory/esystem/

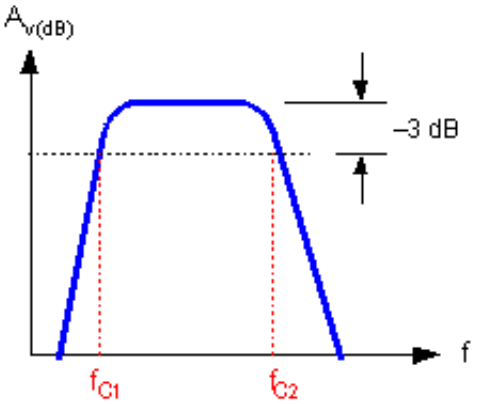
濾波器的頻率響應



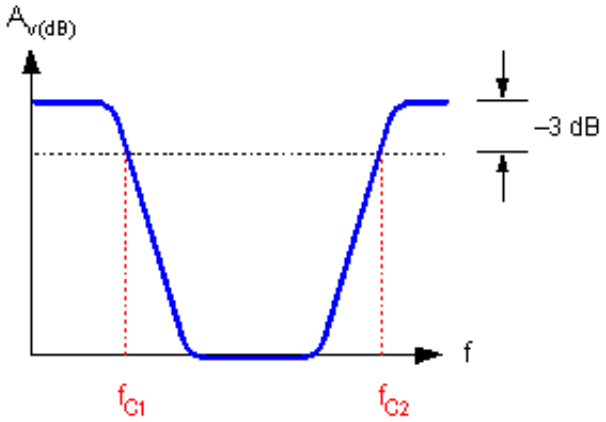
Low-pass filter



High-pass filter



Band-pass filter



Band-stop (notch) filter

帶通 = 高通 (串聯) 低通

http://www.allaboutcircuits.com/vol_2/chpt_8/4.html

帶拒 = 高通 (並聯) 低通

http://www.allaboutcircuits.com/vol_2/chpt_8/5.html

噪音消除示範

- 錄製一段語音
  - 語音的能量從頻譜來看，大約以 10kHz 以下的頻率為主
  - 用低通濾波器(10kHz) 對語音進行濾波，聽看看是否有差別

- 錄製一段噪音
  - 產生白噪音 (取樣率 44.1kHz => 白噪音頻寬約 0 – 22 kHz)
  - 用高通濾波器將 11 kHz 以下的噪音能量去除

- 將過濾後的語音和噪音混合，聽看看
- 把混合的聲音，再用低通濾波器(10kHz) 過濾，再聽看看

噪音消除 -當想要聽到的聲音和噪音有共同的頻率成分

- 如果噪音的特性是穩定的，例如平均值為 0，在頻譜上的能量分布也是穩定的 (例如白噪音) => 看頻譜
- 那麼就有機會估計出噪音的頻譜
  - 判斷目前是否有想聽到的聲音，如果沒有，目前收到的就都是噪音
- 把收到的信號的頻譜，扣掉噪音的頻譜，就是想聽到的聲音的頻譜。
- 把頻譜切成好幾段，在每一段裡面觀察其中的噪音和想聽到的聲音的信噪比，如果噪音比較大，信噪比低，就把那個頻段的增益調降。如果信噪比高，就維持目標增益。

噪音消除

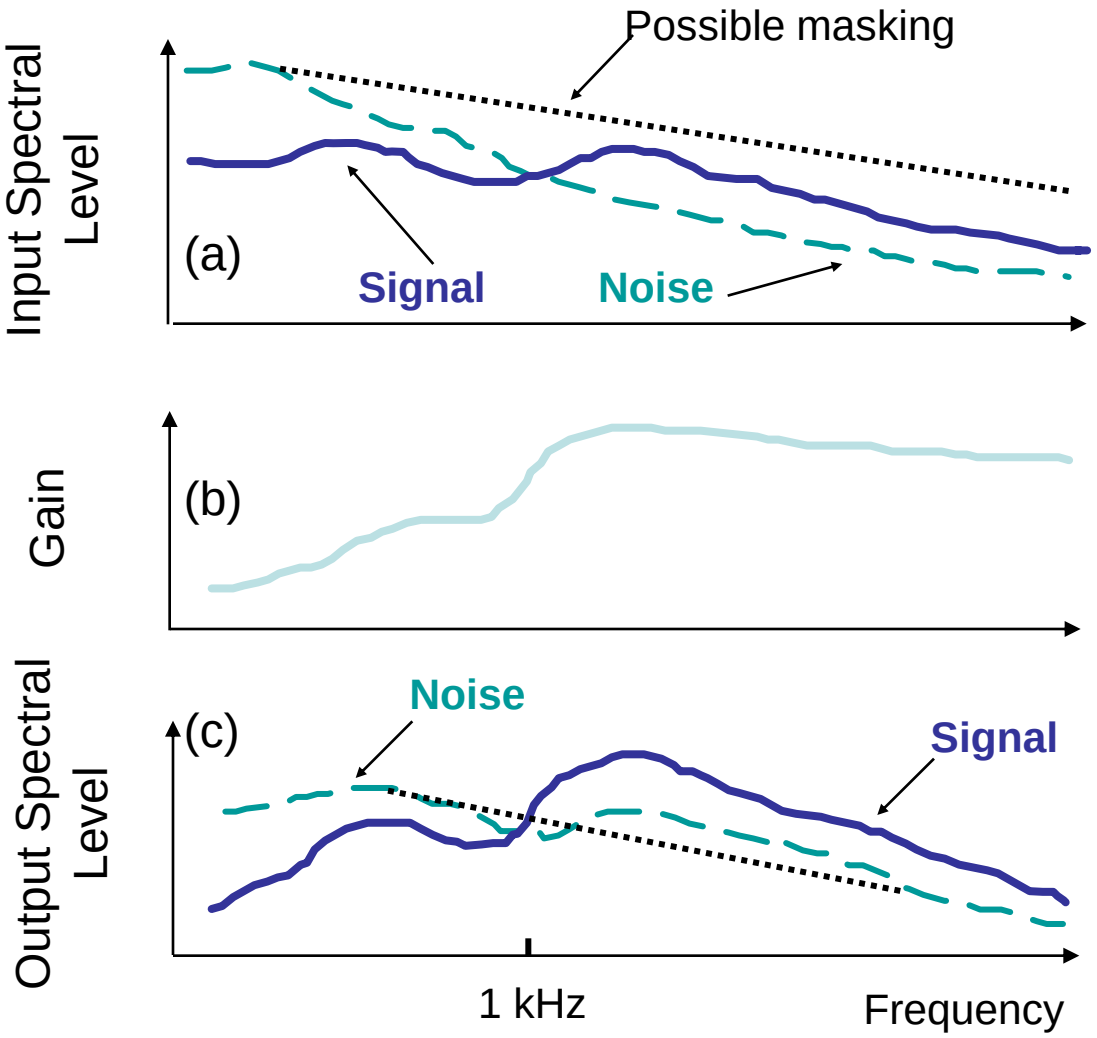


Figure 6.12 (a) Spectrum of the signal and noise input to a noise reduction hearing aid. (b) Gain applied to the signal and noise. (c) Spectrum of the signal and noise at the hearing aid output.

Source: Dillon (2001): *Hearing Aids*

噪音消除 - 副作用

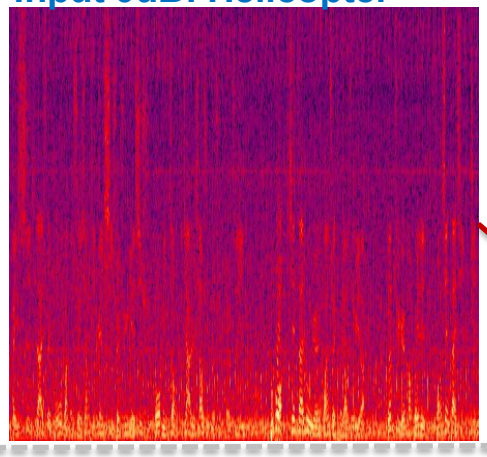
- 噪音消得越多，想聽的信號也可能跟著被消除得越多
  - Speech distortion index

噪音消除 - 其他的想法

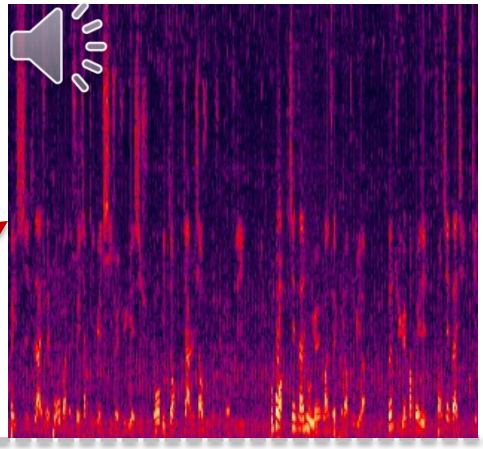
- 如果想聽的信號就是指語音
  - 用盡對產生語音 (speech production) 的知識來處理收到的信號
  - 也可以加上語言的模型 (language model) 來增強信號
- 只要假設是合理的，盡可能利用對該假設的已知，可以做出較好的估計
- 比較困難的情況
  - 嘈雜的人聲背景
  - 非穩態 / 短暫的噪音干擾

I. 穩態噪音:

Input 0dB: Helicopter



Output

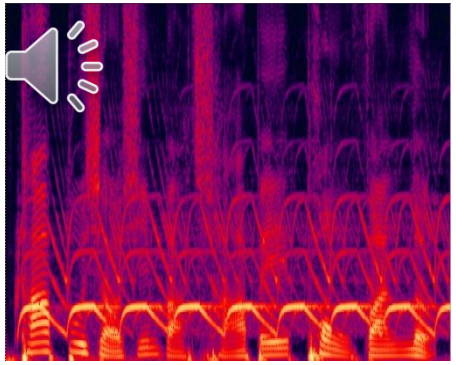


MMSE

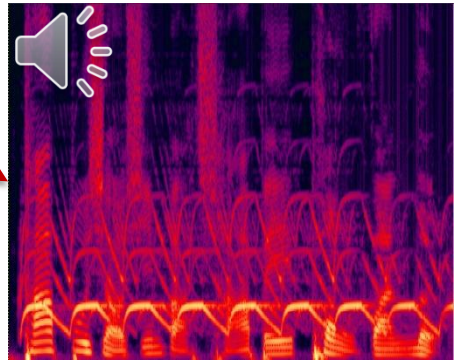
噪音
消除演算

II. 非穩態噪音:

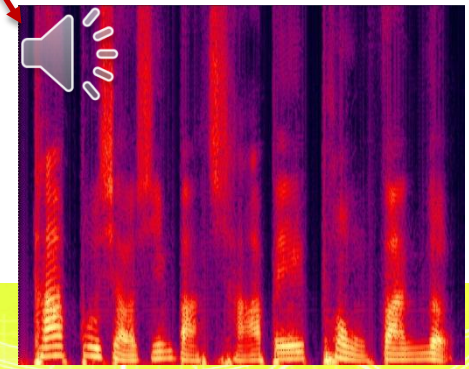
Input 0dB: siren-wail



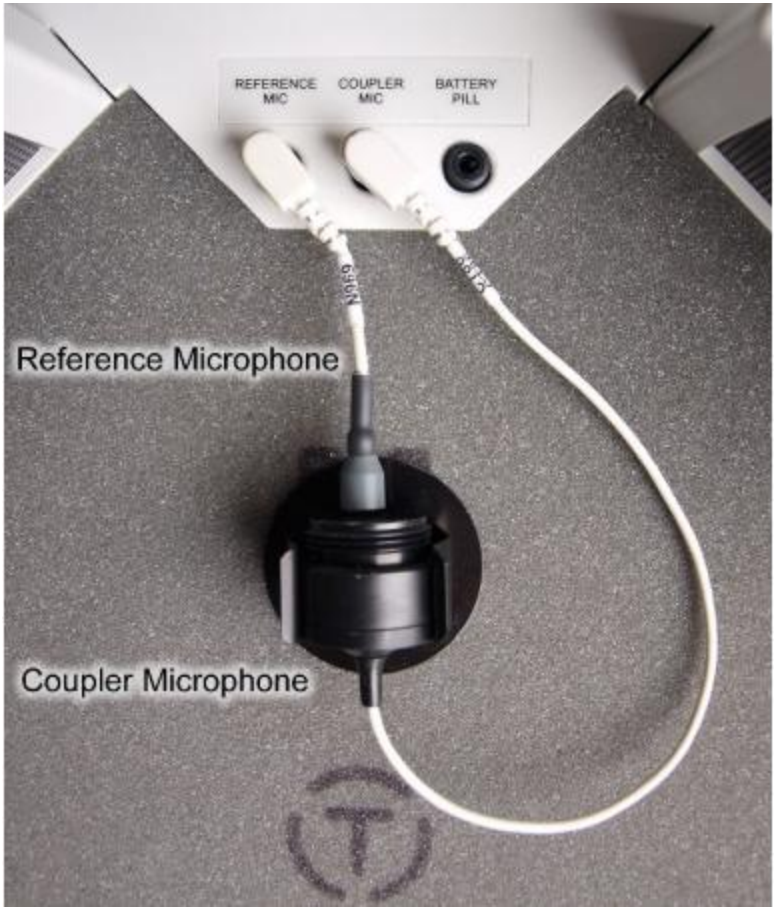
MMSE



DNN



噪音消除量測步驟一:

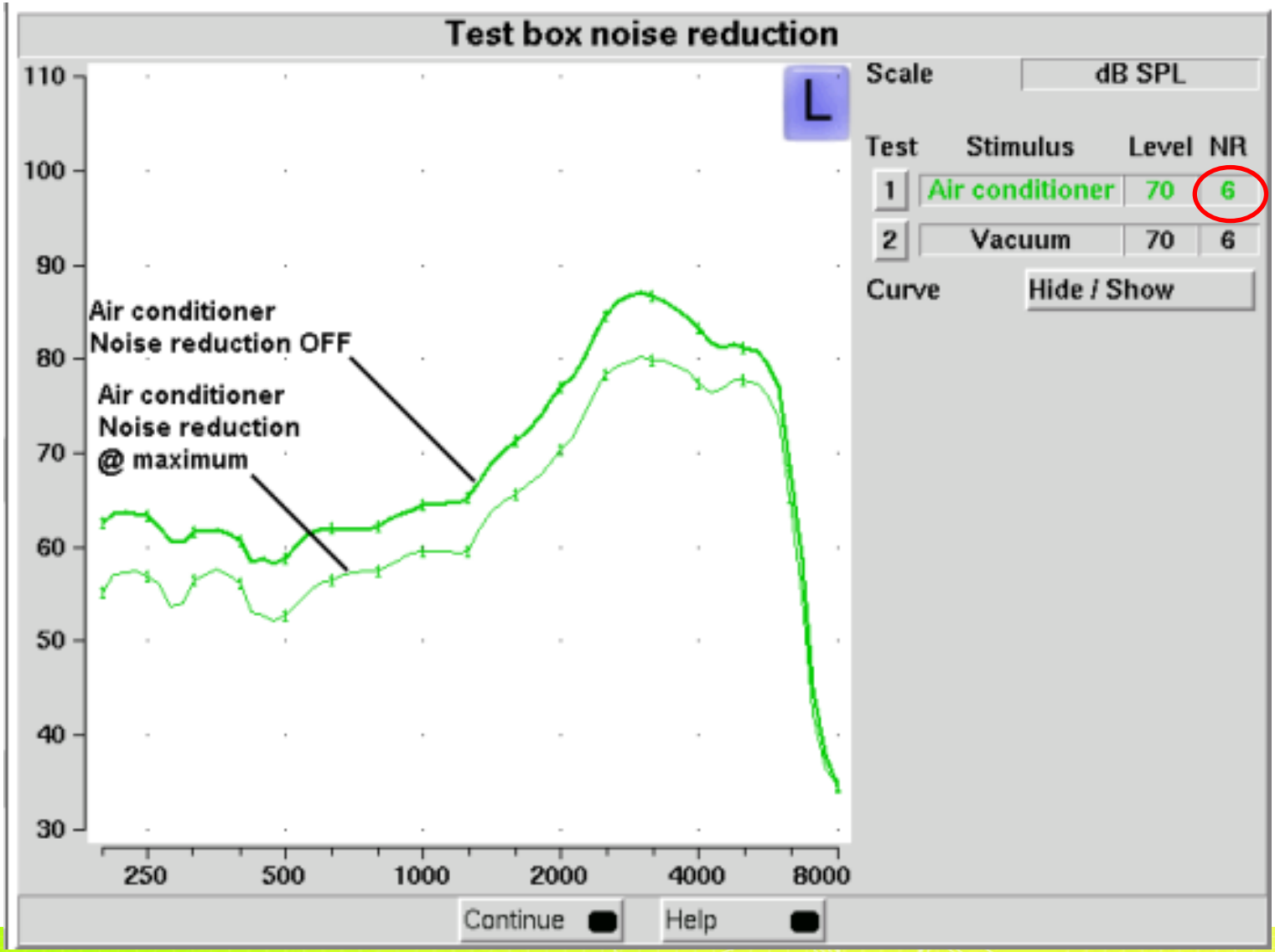

REFERENCE MIC
COUPLER MIC
BATTERY PILL
Reference Microphone
Coupler Microphone

噪音消除量測步驟二:

Sets of audioscan verifit:

1. Press “Test” -> “Test box measures” -> “Noise reduction.
2. Highlight and <PICK> Test 1, 2 or 3.
3. Highlight and <PICK> a stimulus type from the list box.
4. Highlight and <PICK> a stimulus level from the list box.
5. Highlight and <PICK> [Start test]
6. Highlight and <PICK> [Continue] to capture the comparison curve once the noise reduction function has responded.

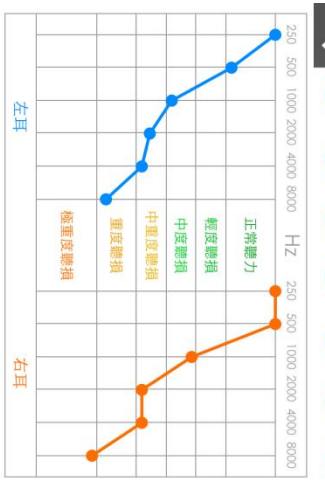
噪音消除量測結果


Test box noise reduction
110
100
90
80
70
60
50
40
30
L
Air conditioner
Noise reduction OFF
Air conditioner
Noise reduction
@ maximum
250
500
1000
2000
4000
8000
Scale
dB SPL
Test
Stimulus
Level
NR
1
Air conditioner
70
6
2
Vacuum
70
6
Curve
Hide / Show
Continue
Help

透過 app 展示噪音消除效果

Forced-Choice Word Recognition Test

Hearing threshold screening in ambient noise



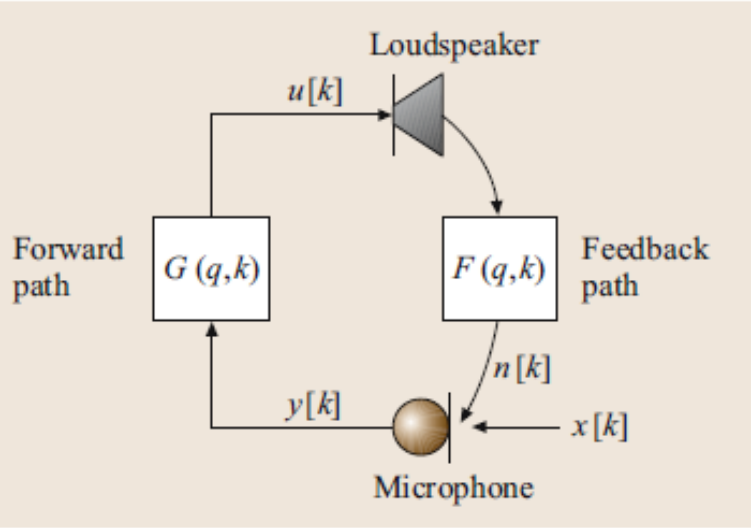
回饋音消除

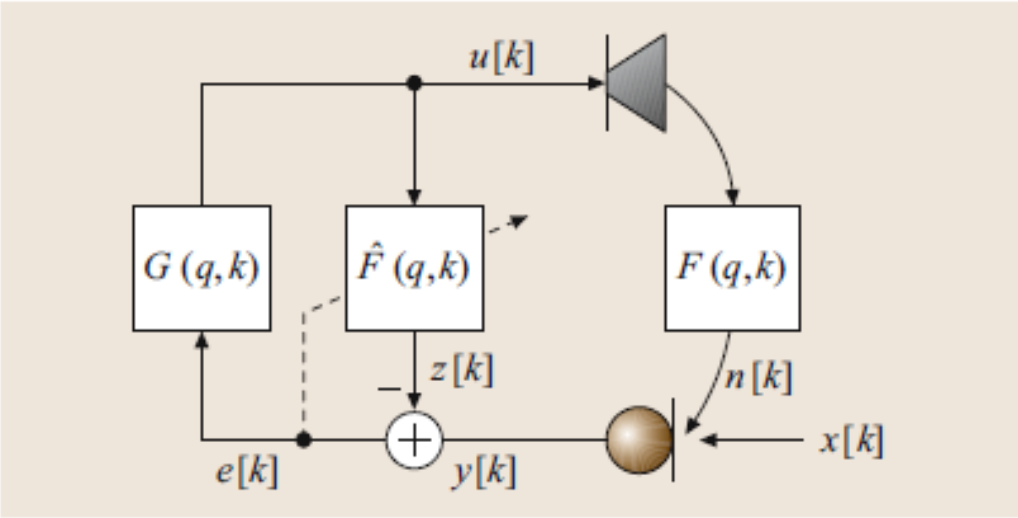
回饋音展示

如何在這裡產生回饋音?

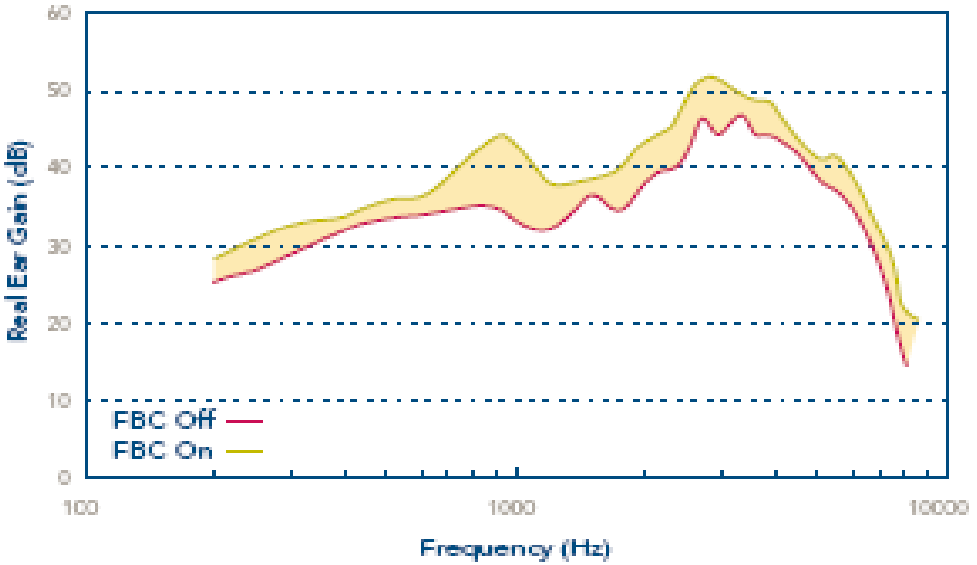
為什麼會有回饋音?

問題描述


Loudspeaker
u[k]
Forward path
G (q,k)
F (q,k)
Feedback path
n[k]
y[k]
x[k]
Microphone


u[k]
G (q,k)
F̂ (q,k)
F (q,k)
z[k]
n[k]
x[k]
e[k]
y[k]

最大穩定增益量 / 可增加的穩定增益量



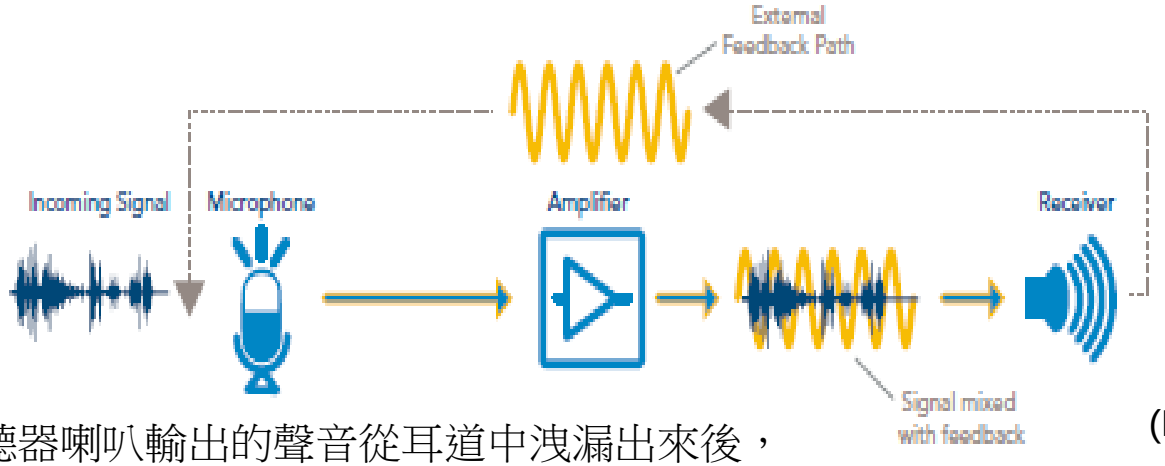
(Banerjee, 2006)

*紅線代表關閉回饋音抑制時的最大穩定增益量(MSG off)
*綠線代表開啟回饋音抑制時的最大穩定增益量(MSG on)
*粉紅色區域即為可增加的穩定增益量(added stable gain, ASG) 或是
回饋音發生前之增益差值(additional gain before feedback, AGBF)

(Ricketts et al., 2008)

回饋音之發生

- 使用者對於助聽器的滿意度大幅降低。(Kochkin, 2003)。
- 有28%的助聽器使用者因助聽器產生的嘯叫聲(whistling)而感到不滿意，而大部分的抱怨都與回饋音(feedback)有關。(Kochkin, 2005)。
- 助聽器的麥克風與接收器形成一個回饋音迴路，而此回饋音迴路的反應現象與其頻率、放大狀況、接收器與麥克風間的相位特性、甚至於個人耳道、耳廓等皆有關係。回饋音並非某單一頻率。(Hellgren et al., 1999)

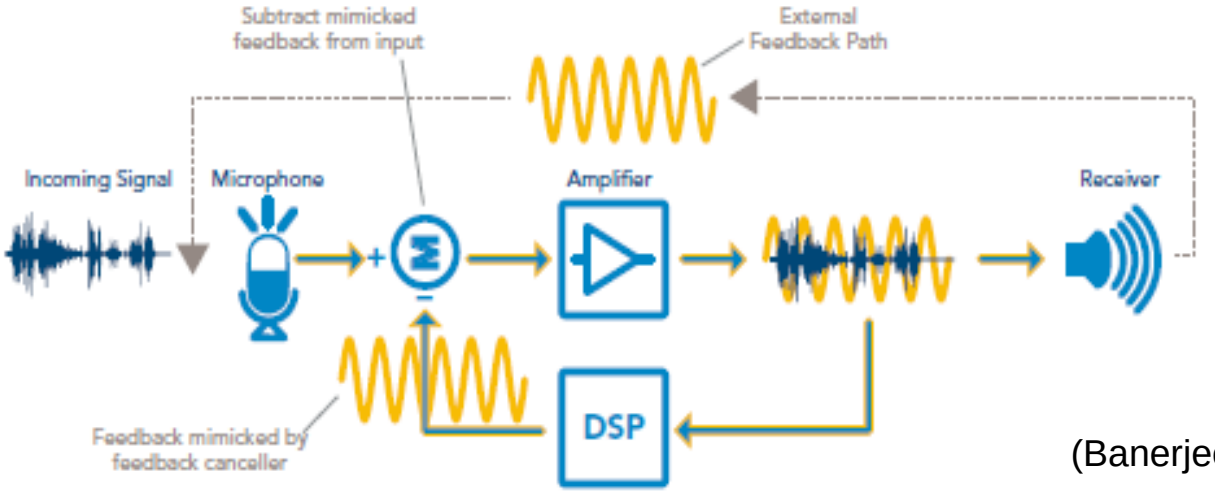


由助聽器喇叭輸出的聲音從耳道中洩漏出來後，又與其他聲音信號由麥克風輸入，並重覆放大，導致回饋音之產生

(Banerjee, 2006)

回饋音抑制的方法

- 回饋音抑制裝置之簡易圖



(Banerjee, 2006)

聲音輸出後被檢測到有回饋音時，助聽器聲音處理器內之適應性濾波器(adaptive filter)估測並模擬外部回饋音產生路徑的聲學響應特性，將喇叭的輸出經此模擬路徑後產生估計的回饋音，再經回饋音反相位處理系統(feedback phase-inverter)將此模擬之回饋音訊號反相位處理後與真實的回饋音相抵消

量測條件

兩組固定數據之聽損模式

	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz	8000 Hz
平坦型(dB HL)	50	50	50	50	50	50
向高頻傾斜型(dB HL)	20	30	50	65	65	65

助聽器

- 各於2011年及2006年生產之耳掛式助聽器A11、A06、B11、及B06。 A11、A06同為廠牌A，B11、B06同為廠牌B。A11、B11為2011年生產之新機種，A06、B06為2006年生產之舊機種 。
- 假設不同廠牌或同廠牌相距五年之助聽器產品代表不同的回饋音抑制演算法，。
- 各為當年代該廠牌自訂為市售之最高階助聽器，且皆以回饋音反相位處理系統(Feedback Phase Inverter)為其回饋音抑制演算法之設計。
- 分別以相同尺寸的open-canal fitting專用細耳管(slim tube)及最小尺寸的open-canal fitting專用耳塞(dome)為試驗工具。

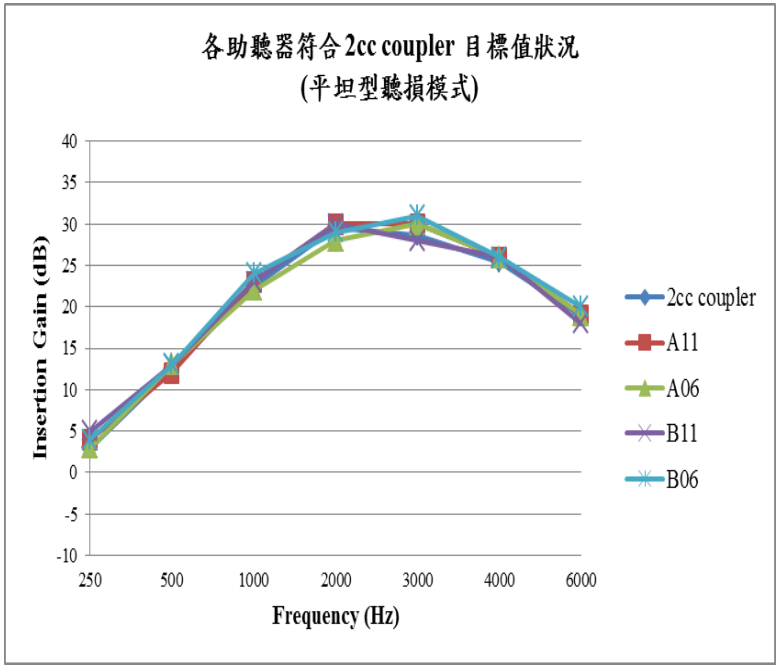
真耳分析儀

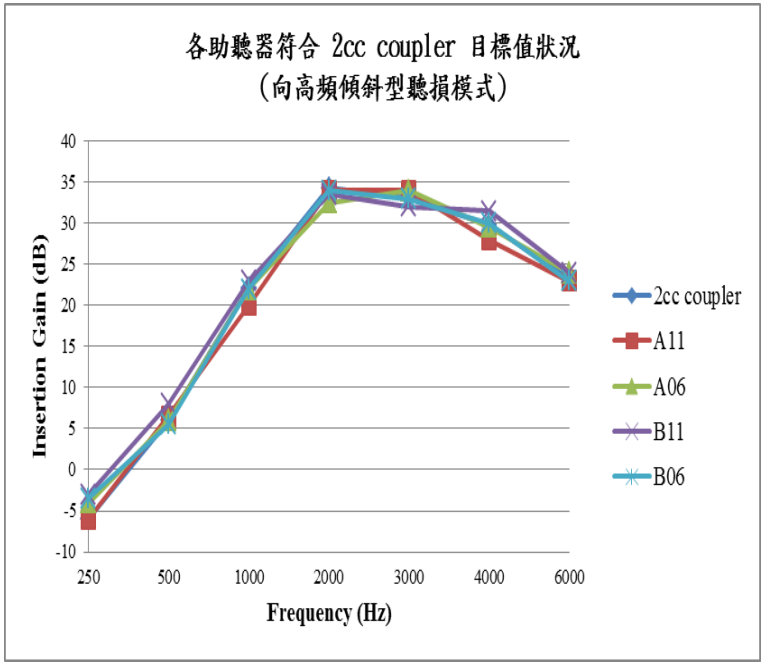
- 美國Frye Electronics公司的 FONIX FP-40真耳分析儀
- 由施測者配戴真耳分析儀的監聽耳機(monitor earphone)進行回饋音監控並記錄結果。

進行量測之前：聽損增益設定

- 助聽器方面

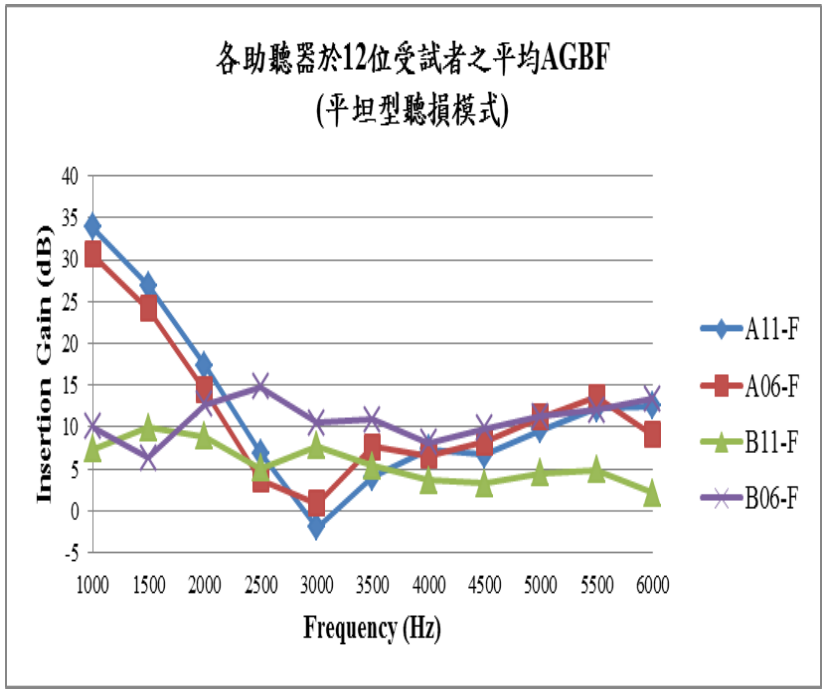
每台助聽器皆選擇NAL-NL1之選配公式，分別輸入以平坦型與向高頻傾斜型兩組聽損資料。除了回饋音抑制功能外，關閉所有可能會影響回饋音及輸出增益的其他特殊功能。將其壓縮閾值(Compression Threshold, CT)與壓縮率(Compression Ration, CR)等影響壓縮的參數控制為一致。以2cc coupler模式調整設定，使50 dB SPL輸入模式時的輸出曲線與上述各聽損模式的目標值吻合。

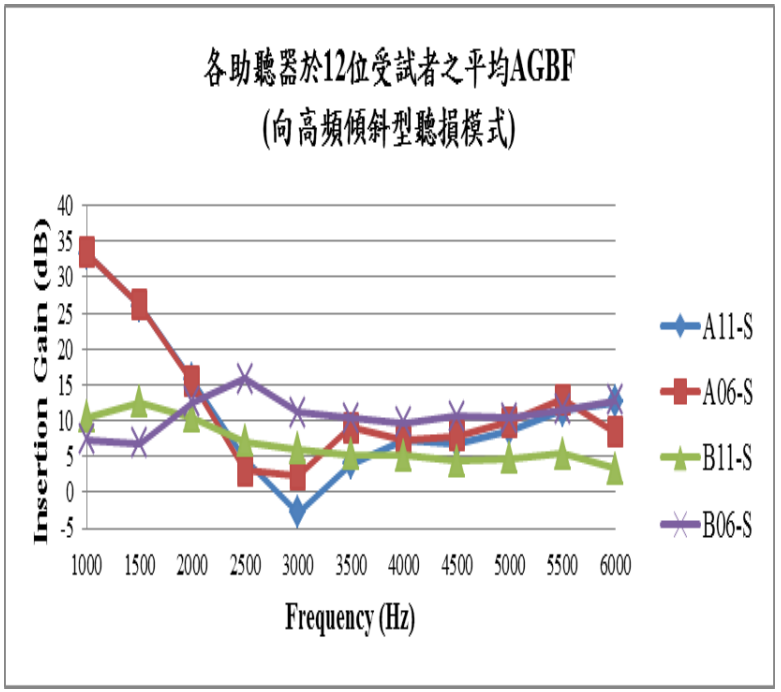




量測結果

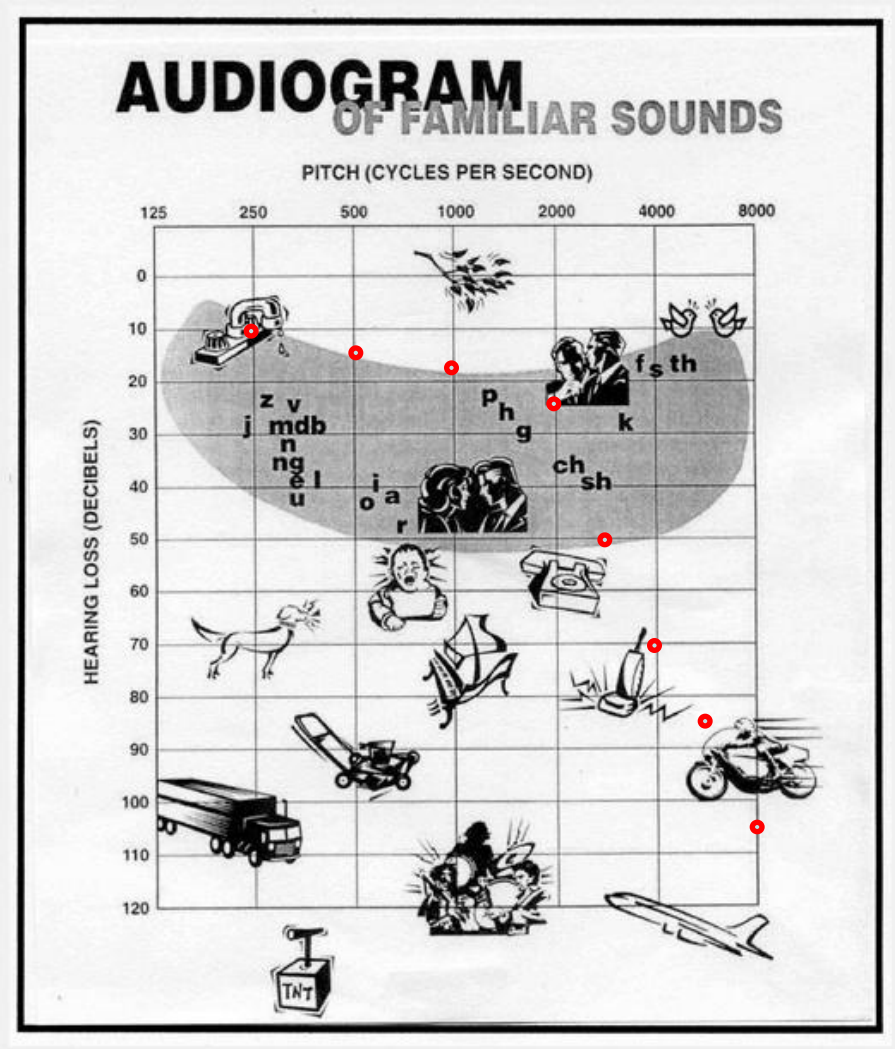
- 以配對樣本 t 檢定檢測平坦型與向高頻傾斜型聽損模式下，各助聽器各頻率於12位受試者平均AGBF之差異性。以p值為判定標準，$p < .05$ (95%信賴區間)表有顯著差異。結果顯示各助聽器A11(p=.168)、A06(p=.242)、B11(p=.303)、B06(p=.911)於平坦型或向高頻傾斜型之聽損模式皆無顯著差異





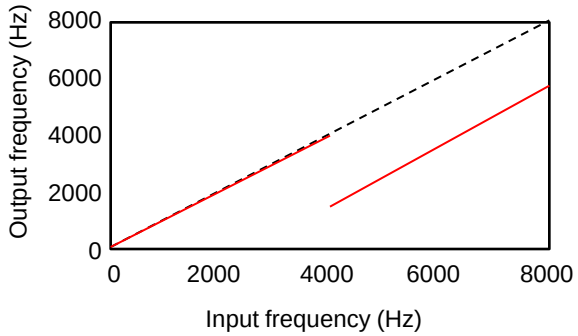
移頻處理概念介紹

聽力圖對照日常生活中常見聲音的頻率分布

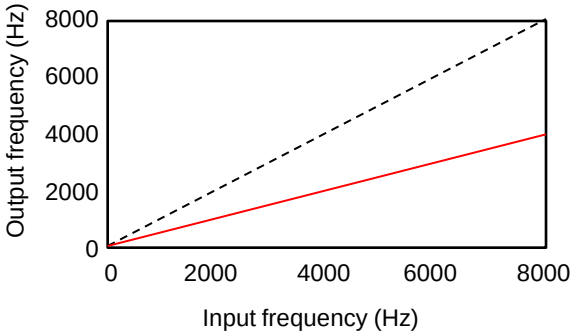


- Steeply sloping hearing loss
- Limitation of traditional means of amplification
  - Inadequate gin
  - Limited bandwidth
  - Acoustic feedback
- Dead region
- Consonants are softer in intensity but critical for speech understanding

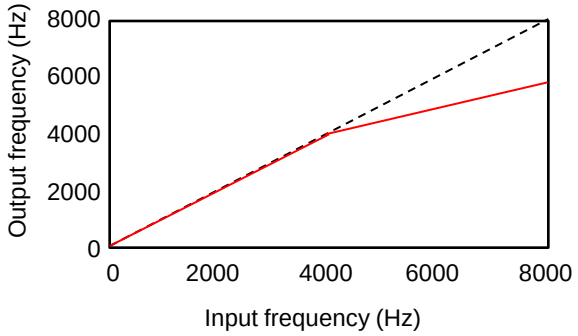
三種移頻處理的方式



- Frequency transposition: to reduce information in frequency by some constant number of Hz



- Linear frequency compression: the output frequency is a constant fraction of the input frequency

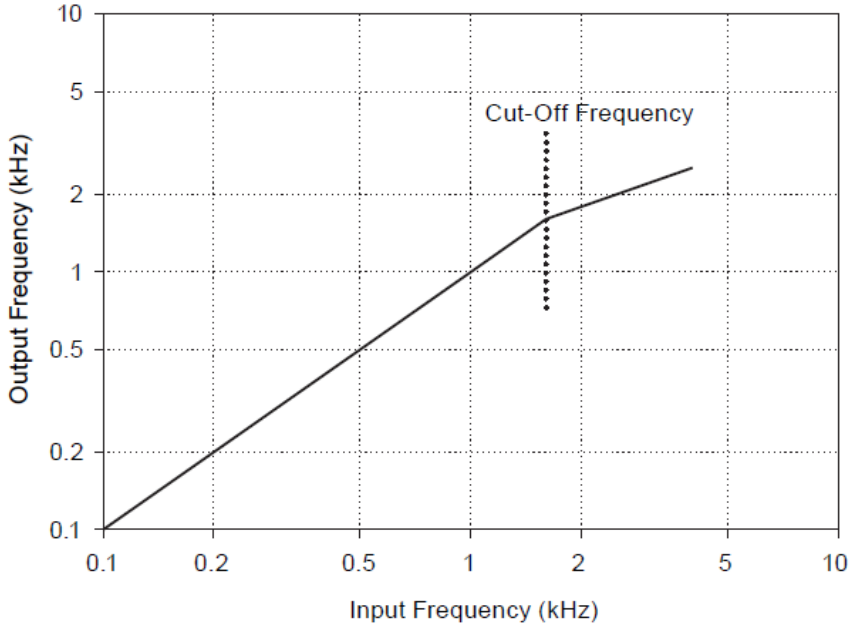


- Nonlinear frequency compression: lowering just the high frequencies

移頻處理: 開始進行移頻的頻率

- For nonlinear frequency compression
  - Cut-off frequency: lowest frequency where information is re-coded

$$F_{out} = \begin{cases} F_{in}, & F_{in} < F_{LPF} \\ F_{LPF}^{1-p} \times F_{in}^{p}, & F_{in} \geq F_{LPF} \end{cases}$$

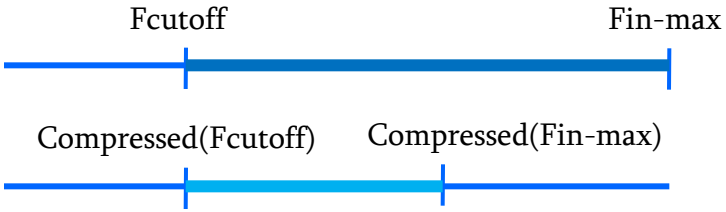


移頻處理: 壓縮率

- Compression ratio:

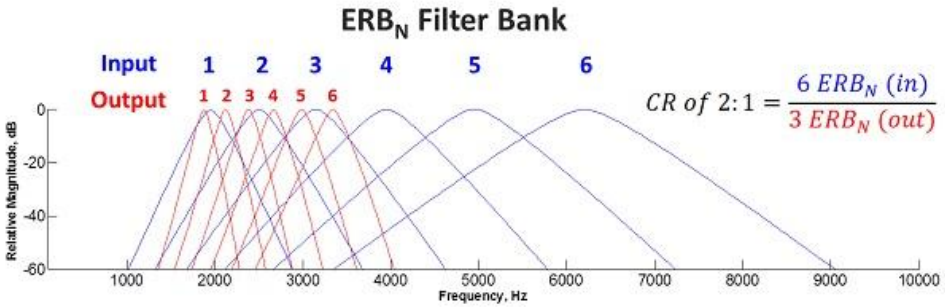
Definition #1:

$$\frac{\text{(Maximum input frequency} - \text{cut-off frequency)}}{\text{(Compressed maximum input frequency} - \text{compressed cut-off frequency)}}$$



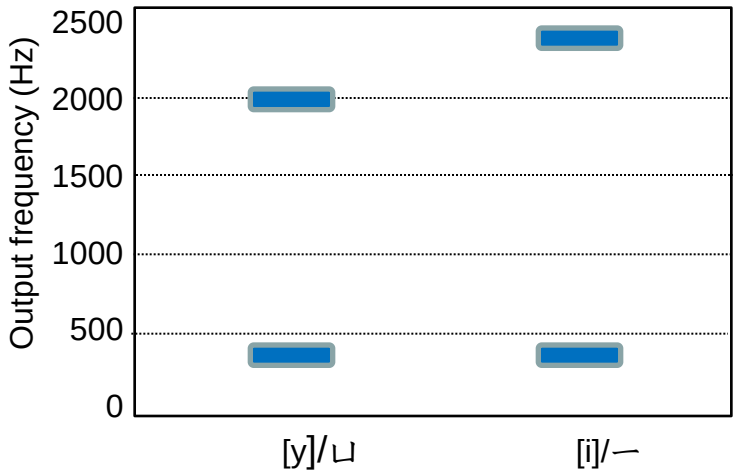
Definition #2:

$$\frac{\text{(Number of equivalent rectangular bandwidths for input compression BW)}}{\text{(Number of equivalent rectangular bandwidths for output compression BW)}}$$



Adapted from http://www.audiologyonline.com/articles/20q-highs-and-lows-frequency-11772

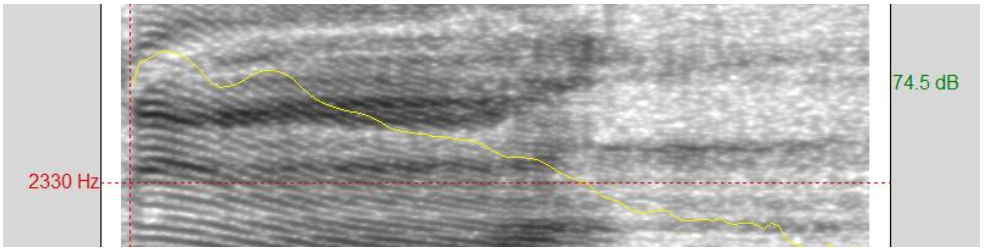
移頻處理對母音共振峰值的影響

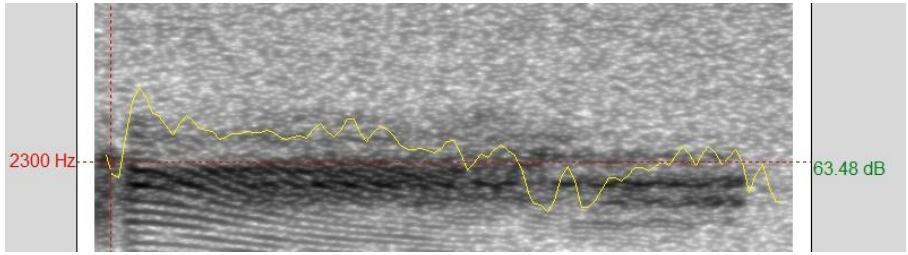


- For cut-off frequency = 1.5kHz and CR=2.3:1
- Higher frequencies are shifted downwards to a larger extent than lower frequencies
- F2 of [i] is shifted more
- The [i] is identified as [y]

移頻處理對頻譜特徵的影響

- Reduced differences between the peaks and valleys in the spectrum
- Merged formants



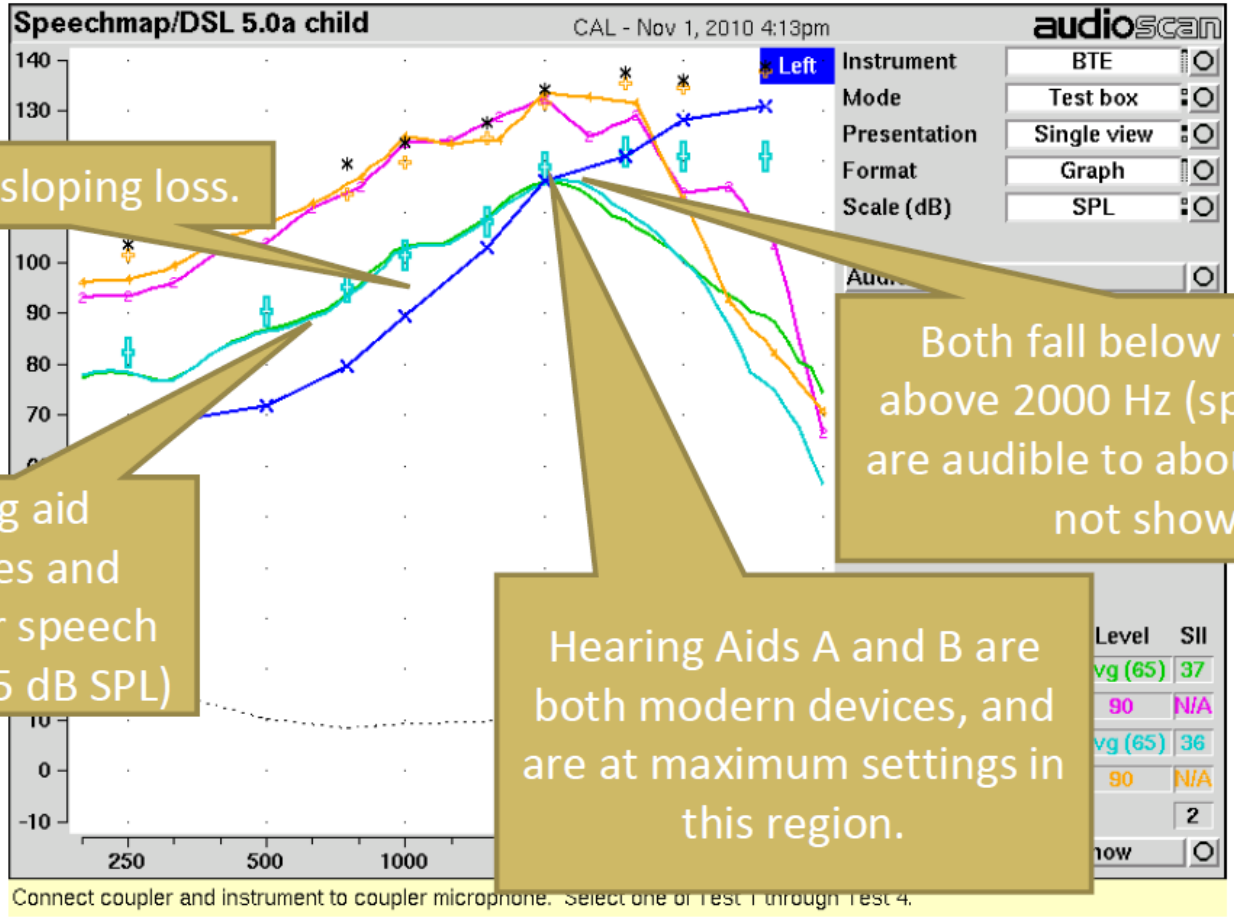


Compressed speech:
Reduced spectral contrast and intensity variation

移頻處理對噪音的影響

- May lead to different SNRs in auditory filters
  - Harmonic relationship
  - Spectral transition
  - Segmental-temporal characteristics
- Research study: the effect of nonlinear frequency compression on the performances of Mandarin Monosyllable Recognition Test (MMRT) in noise and Mandarin Hearing in Noise Test (MHINT)
  - Different performance outcomes and error patterns?
    - Material type
    - Noise type

使用 audioscan 進行量測



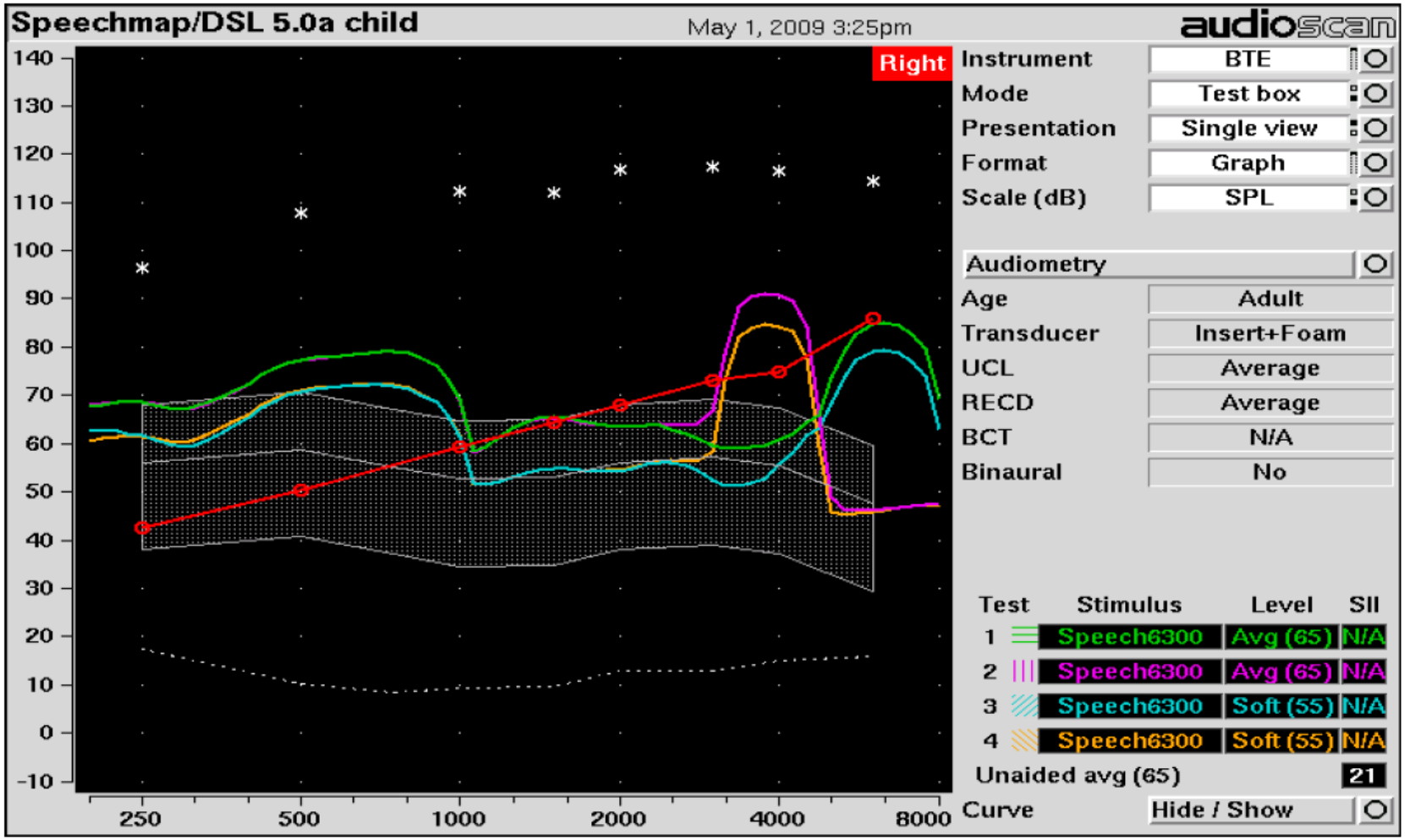
A severe sloping loss.

Hearing aid responses and targets for speech (input at 65 dB SPL)

Hearing Aids A and B are both modern devices, and are at maximum settings in this region.

Both fall below threshold above 2000 Hz (speech peaks are audible to about 2500 Hz – not shown).

高頻信號經過移頻後音量超過聽覺閾值



信號處理的效益與副作用

如何展示

還有哪些問題 (需求)

感謝聆聽

建議與回饋: ankh_li@mmc.edu.tw