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Measurements and Analyses of Calaid—FM Electroacoustic Performance

NAL
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COMMONWEALTH DEPARTMENT OF HEALTH

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MEASUREMENTS AND ANALYSES OF CALAID-FM ELECTROACOUSTIC PERFORMANCE

T. LOI, J. YIP

ADVANCED SYSTEM DEVELOPMENT AND EVALUATION SECTION

NATIONAL ACOUSTIC LABORATORIES

COMMONWEALTH DEPARTMENT OF HEALTH

1. Frequency Response of Calaid-FM system

1.1 Introduction

The frequency response of the Calaid-FM system has been documented (1,2). This section focuses attention at low and high frequencies where performance discrepancies were reported. In attempting to isolate sources of discrepancies, circuit sections were examined to determine their contribution to the response error. The majority of the testing were carried out with the noise microphone disconnected. This enables testing to be done with acoustic sources generated in a conventional test box. The development of the internal low frequency noise cancelling system (3) and ts performance have been documented (2,3). The section on AGC in this report includes aspects of the noise cancelling effects.

To avoid any effect of the AGC on frequency response, broadband noise were used as input signals. The response was analysed using a HP3582A spectrum analyser. When the analyser is used in its transfer function mode, the acoustic inputs were monitored with a B&K 4134 microphone.

The spectral power of the acoustic signal used in testing is given in figure 1-1. The power level below 500 Hz corresponds to that of average speech spectrum on assuming that the microphone is located 100mm from a speaker's mouth. Levels higher than expected-speech-levels were used at high frequencies for better signal to noise ratio, (indicated by a high correlation factor using the spectrum analyser). This does not affect the validity of response measurements as the AGC operation is linear at these levels. Figure 1-2 shows the response of the A151 integrated circuit incorporating the AGC when the input sound level is varied.

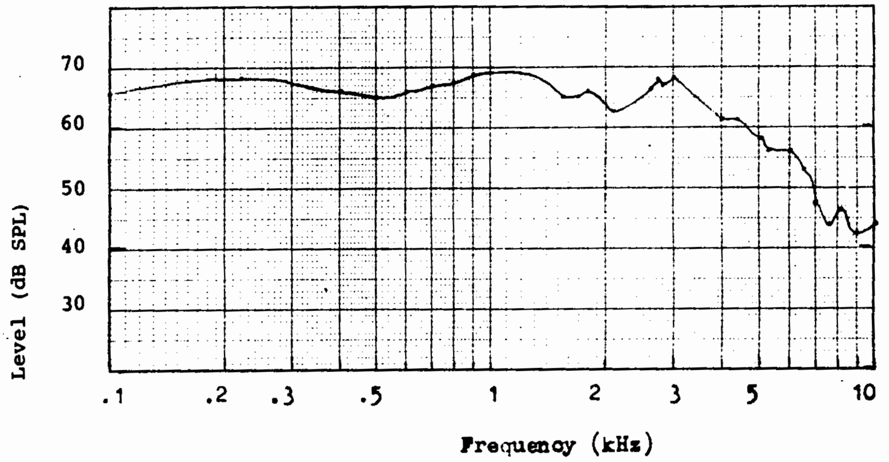


Fig 1-1 Acoustic test signal

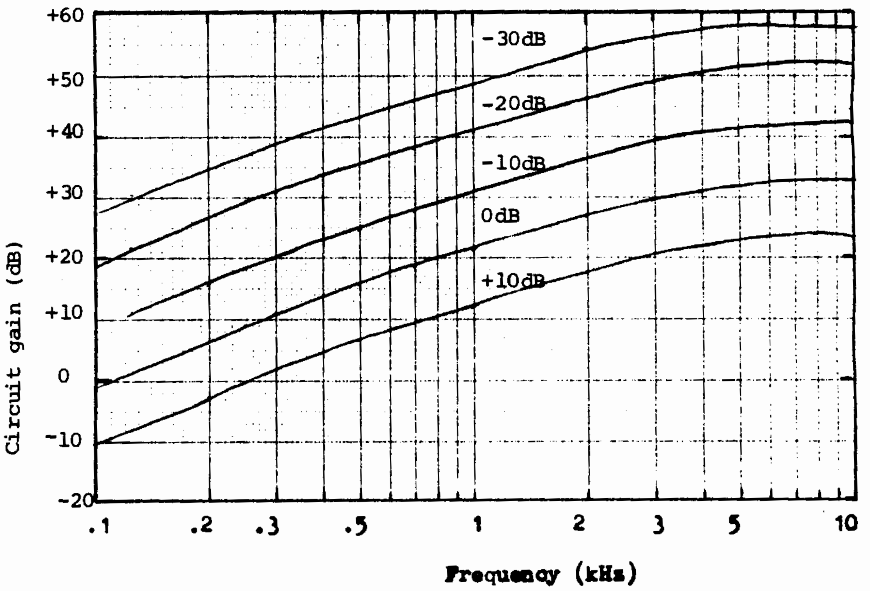


Fig 1-2 Transfer function of the audio IC A151 when the input signal level is varied in 10dB steps. Test signal given in Fig 1-1 gives curve "0dB"

1.2 Transmitter-receiver response

The frequency response of the Calaid-FM system had been specified as follows:

+ 6 dB/octave, 100 Hz to 4 kHz.

+ 12 dB/octave, below 100 Hz.

-6 dB/octave, above 6 kHz.

The pre-emphasis up to 2 kHz is implemented electronically in the transmitter audio circuit, and the rest of the rising slope from 2 kHz to 4 kHz is obtained by using a microphone with a rising response in this region. Fig. 1-3 shows the response of a "phase 2" transmitter and receiver. The response conforms to the design specification in the middle range, sloping away below 400 Hz and rolling off steeply above 4 kHz.

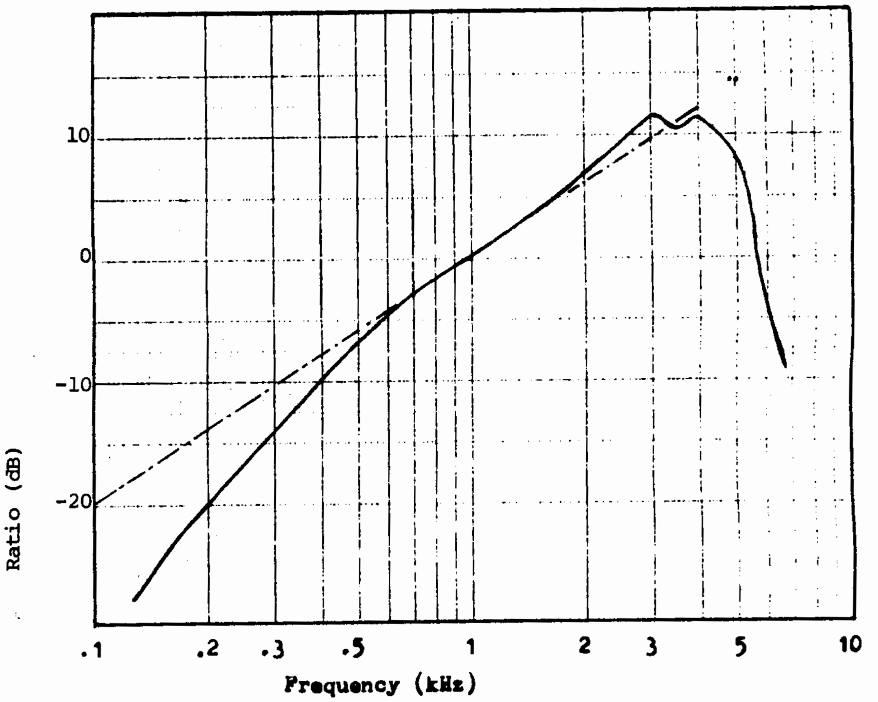


Fig 1-3 Response of phase 2 transmitter and receiver (fitted with BT 1751 microphone), dotted line representing 6dB/octave

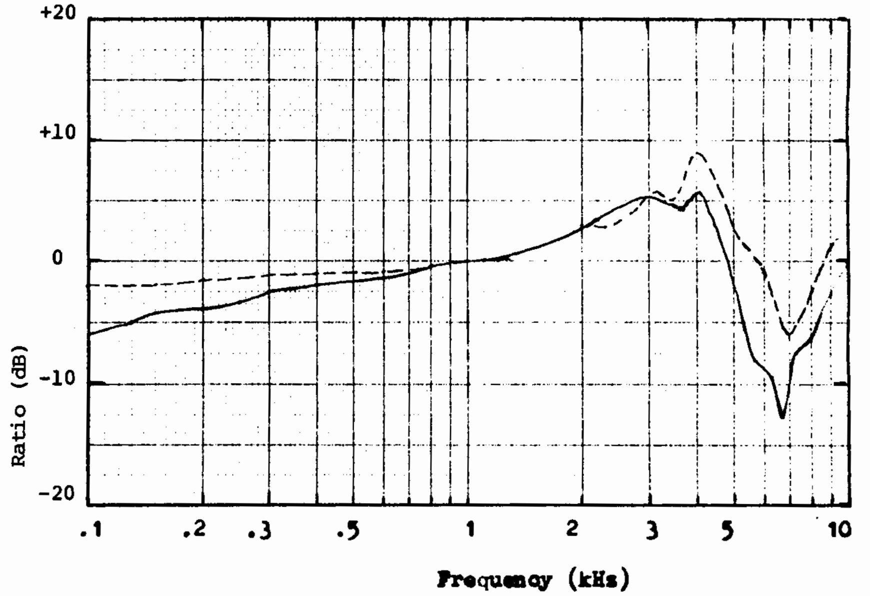


Fig 1-4 Response of microphone mounted in housing and boom

------ BT 1754

______ BT 1751

1.3 Microphone response

Most of the high frequency loss is accounted for by the roll-off in the Knowles BT 1751 microphone used. It also contributes toward the loss at low frequencies. The response both at low and high frequencies is improved with the use of BT 1754 microphones. Figures 1-4 shows the (difference between the two microphones). Mounting the microphone in the housing on the boom affects its response. Placing it too close to the "top" of the housing introduces peaks and makes the response sensitive to the placement of the housing cap. Stable responses can be obtained when the microphone is recessed at least 2 mm from the top. It is recommended that the microphone should be mounted as far back as possible (3.5 mm) in the housing.

off. The amount of attenuation depends on the DC level from the receiver and to a minor extent on the characteristics of the microphone. When the DC level is 1.5 volts, the amount of attenuation is nominally 50dB. When VR2 is set to give 26 dB attenuation of the microphone output when the DC level applied to the adaptor is 1.20V, the attenuation characteristic at 1 kHz for a sample hearing aid is shown in Figure 3-2. When the DC level from the receiver is above 1.1V, the amount of attenuation increases sharply. With the receiver in the combined mode, (no DC level from the receiver), the hearing aid its microphone, sensitivity is reduced by about 3 dB.

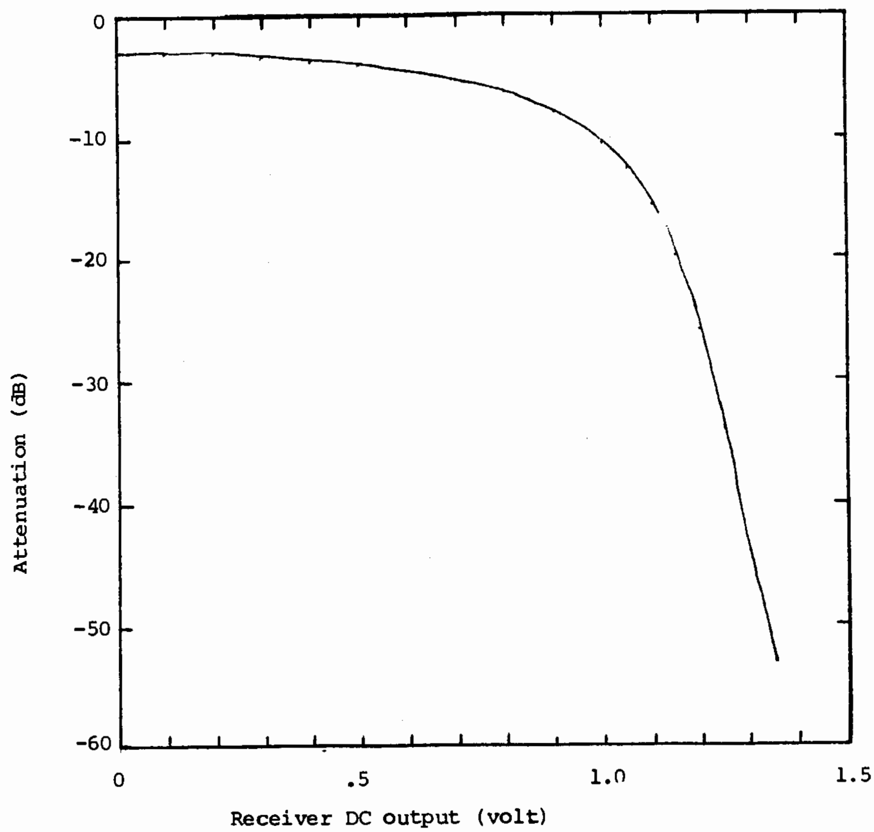


Fig 3-2 Attenuation of H-A microphone output vs the DC level of the receiver for a sample hearing aid

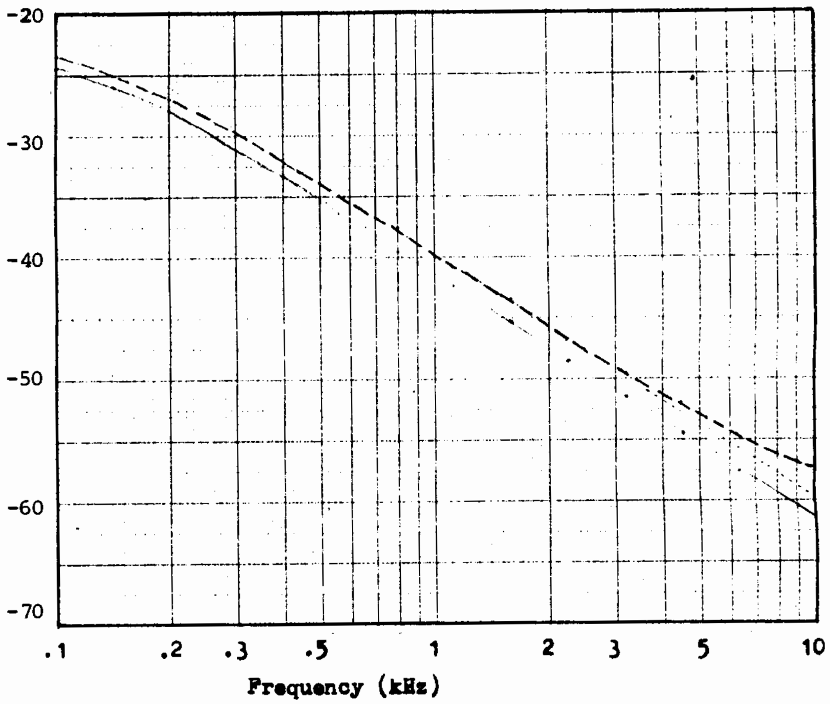


Fig 3-3 Adaptor frequency response

----- Measured

_____ Calculated

......-6dB/octave line

3.2 De-emphasis

Referring to the adaptor circuit diagram, VR1 and C_1 provide de-emphasis above 80 Hz. Capacitor C_2, required to isolate the DC component from ground, introduces a response roll-off commencing at 1.7 Hz. This results in an extra 20 dB of attenuation in the hearing aid frequency band when the volume control VR1 is set to maximum. Figure 3-3 shows the open circuit response of the adaptor with VR1 = 200 , and with S1 open. The measured response tracks the -6 dB/octave line within $\pm$ 1 dB between 200 Hz and 6 kHz. Above 6 kHz, the series resistance of capacitor C_1 starts to limit the high frequency roll-off. A series resistance of 2Ω was measured for the capacitor giving a breakpoint at 8 kHz. Since there is no pre-emphasis beyond the microphone response peak, de-emphasis need not be extended beyond 4.5 kHz, except perhaps for the purpose of noise reduction. The deviation at low frequencies increases the low frequency roll-off of the overall system, by an extra 3 dB at 100 Hz. This can be reduced to 1 dB by lowering the de-emphasis breakpoint a further octave down to say, 40 Hz. This can be done by doubling either C_1 or VR1. Doubling VR1 would be prefered, as this would maintain the response shape at higher frequencies.

3.3 Effect of loading

When the hearing aid is connected to the adaptor, it presents two distinct values of load resistance R_L to the adaptor. The value of R_L depends on whether the DC level is high or low. When the DC level is low, R_L is approximately equal to the output impedance of the Knowles EA1842 microphone. The nominal value is 3.5kΩ. This loads the adaptor output by around 8.5 dB with S1 open and around 2.5 dB with S1 closed. With a high DC level from the receiver, the hearing aid microphone output appears to be biased off and presents an impedance of around 20kΩ. The loading on the adaptor output is small: 0.5 dB when S_1 is closed and 2.5 dB when S_1 is open. The results effect of R_L causes a loudness change. With S_1 open, the normal mode of operation, the difference could be 6 dB, being louder when the DC level is high. In practice, the difference in volume from the hearing aid may be less than 6 dB. Because, with the DC level low, the hearing aid also responds to its own microphone, the combined output may decrease the difference in the perceived volume.

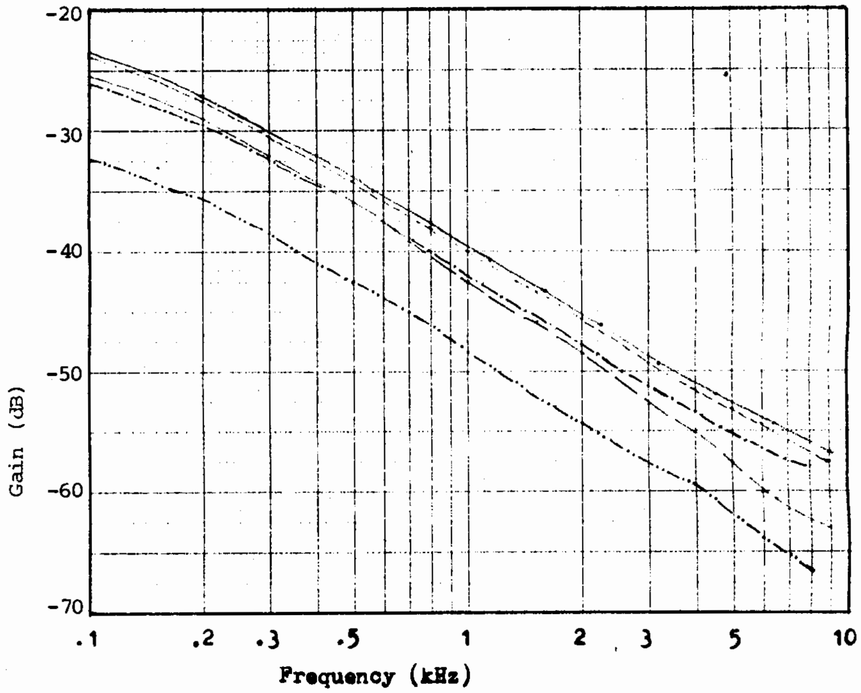


Fig 3-4 Adaptor response (transmitter in "FM")

_______ Hearing aid not connected

------- Loaded with hearing aid, Receiver in "FM", S1 closed

—■·— Loaded with hearing aid, Receiver in "FM", S1 open

—·—·— Loaded with hearing aid, Receiver in "C", S1 closed

—··—··— Loaded with hearing aid, Receiver in "C", S1 open

3.4 Volume control

The speech level from the FM system can be varied to match the level from the hearing aid microphone. Figure 3-5 shows that the volume control potentiometer can provide up to 15 dB of attenuation without affecting the frequency response. Beyond 15 dB, the control is too sensitive and the capacitor C_2 starts to affect the frequency response by limiting the attenuation at low frequencies.

Doubling VR1 as mentioned in Section 3.2 would alos increase the linear volume control range by 6 dB. In a noisy environment, when operated in the combined mode, it may be desirable to decrease the sensitivity of the hearing aid microphone to reduce the amount of noise pick up relative to the FM system. (The FM system incorporates noise cancelling and with its microphones located close to the talker's mouth, the signal to noise ratio is much better than that of the hearing aid microphone input). This facility is provided by a switch, S1.

As shown in Figure 3-4 and in Figure 3-6, closing the switch S_1 decreases the hearing aid microphone output by about 8 dB and increases the signal from the adaptor by 6 dB resulting in a relative change of 14 dB in favour of the FM system.

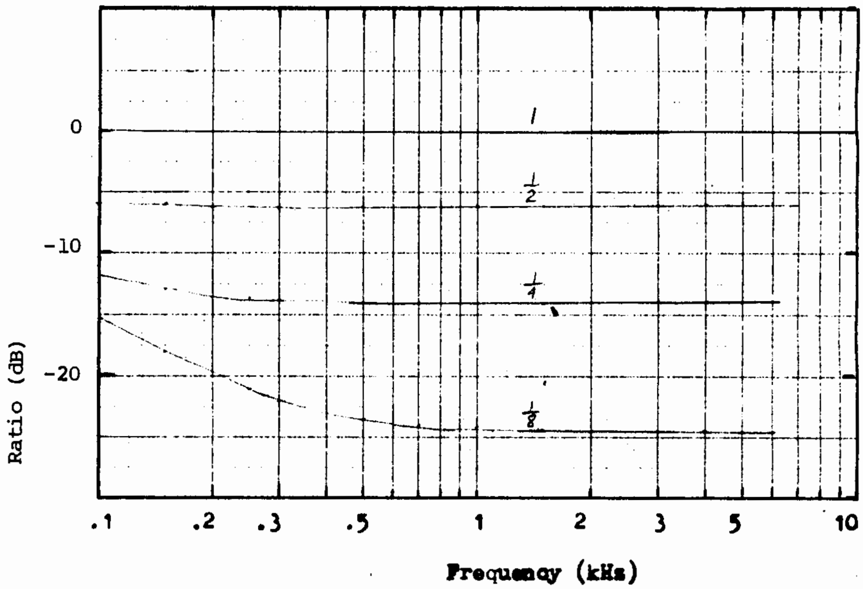


Fig 3-5 Action of adaptor volume control relative to maximum. Settings indicated only reflect approximate volume control positions.

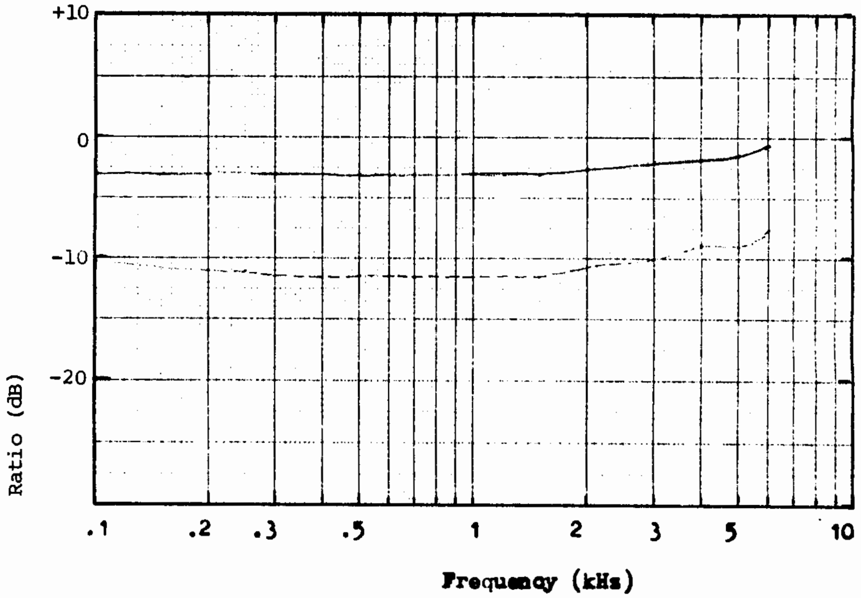


Fig 3-6 Reduction of Phonak aid microphone (EA 1842) sensitivity when the aid is coupled to the FM system via the adaptor. Receiver output in "C" mode.

____ S1 open

---- S1 closed

3.5 Hearing aid response

Figure 3-7 shows the responses of the hearing aid on its own and when coupled to the FM system via the adaptor. In both cases, the hearing aid switch positions were set to "Tone = L4, dB-SSPL = 136". Random noise spectrum-weighted to approximate average speech (4) were used for the measurements. The level chosen was 12 dB higher than in (4) as, with the use of the FM system, the speech microphone is nominally located 100mm away from the talker's mouth, while the level obtained in (4) uses microphone placed 400mm away from the source. To avoid saturating the hearing aid, the volume control of the hearing aid is set to "2". Using its own microphone, the aid output was 108 dBSPL. When coupled to the FM system (receiver output in "FM" and the adaptor volume at mid-travel position and S_1 open) the output level was 4 dB higher. Coupling the hearing aid to the FM system does not the affect the response between 100 Hz to 2.5 kHz. As shown in Figure 3-7 a loss in gain above 2.5 kHz is evident. The difference in the response shape is shown in Figure 3-8. The difference is accounted for by the microphone fitted to the hearing aid having a rising response at high frequencies around 4 kHz.

To minimise the difference, it would be desirable to have a similar boost in the adaptor output. The Knowles BT 1754 microphones used in the transmitter has a nominal high frequency boost of 11.5 dB at 4.7 kHz. This is 7 dB more than that of the Knowles EA1842 microphone used in the Phonak hearing aid. The boost is reduced, by the high frequency roll-off in the transmitter output stage (-2 dB at 4 kHz) and the de-emphasis circuit in the adaptor. The later cut the microphone boost by

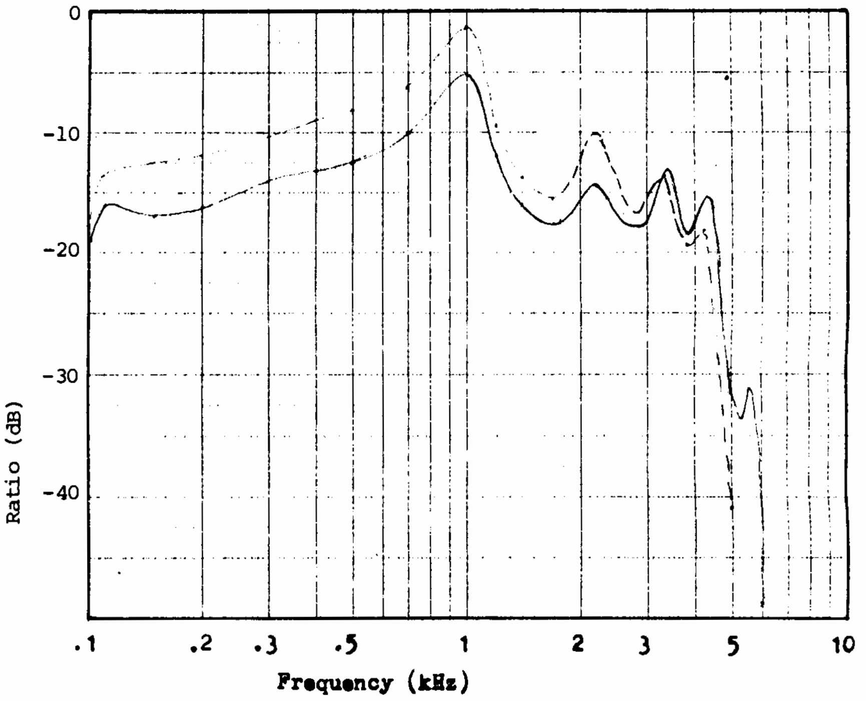


Fig 3-7 Frequency response of Phonak PPCL aid: volume = 2"

_____ on its own

----- coupled to the FM system: adaptor volume = 1/2"

6 dB at 4 kHz. (The pre-emphasis in the transmitter above 2 kHz relies on the rising response of the microphone). With a nominal BT 1754 microphone, there would still be a boost of 3.5 dB. The actual microphone used in the transmitter for the measurements has a response peak of only 9 dB at 4 kHz, (Fig. 1-4). While the hearing aid microphone has a boost of 6.2 dB at 4.1 kHz. The difference in the inputs to the hearing aid amplifier is shown in Figure 3-9. Nominally the BT 1754 microphone has sufficient boost to provide the pre-emphasis above 2 kHz, to compensate for the high frequency roll-off in the transmitter output stage, and still providing some high frequency boost. This applies only for frequencies up to the response peak, beyond which the respond rolls off much more rapidly than the EA 1842 microphone. From the manufacturer's data, the amount of high frequency boost for the BT 1754 microphone has a spread of 10 dB (between +6 dB and +16 dB). While the EA 1842 microphone would provide a boost of between 1.5 dB and 7.5 dB. For the FM system, the equivalent amount of boost would be between 9.5 dB and 15.5 dB. Some BT 1754 microphones would have insufficient boost (by up to 3.5 dB at about 4.5 kHz) for the FM system.

(Alternatively BT 1751 microphones can be used for the FM system. The de-emphasis circuit in the adaptor would need to be modified. That is, a -12 dB/octave slope between 100 Hz and 400 to obtain a flat low frequency response. Above 400 Hz, the - 6 dB octave slope should be maintain only up to 1.5 kHz. The advantages gained with this alternative is a less rapid response roll off above the response peak and a narrower spread of high frequency response boost. The existing method using BT 1754 microphone has advantages in a lesser component count in the adaptor, a better signal-to- noise ratio in the high frequencies and a lower level of harmonic distortion.

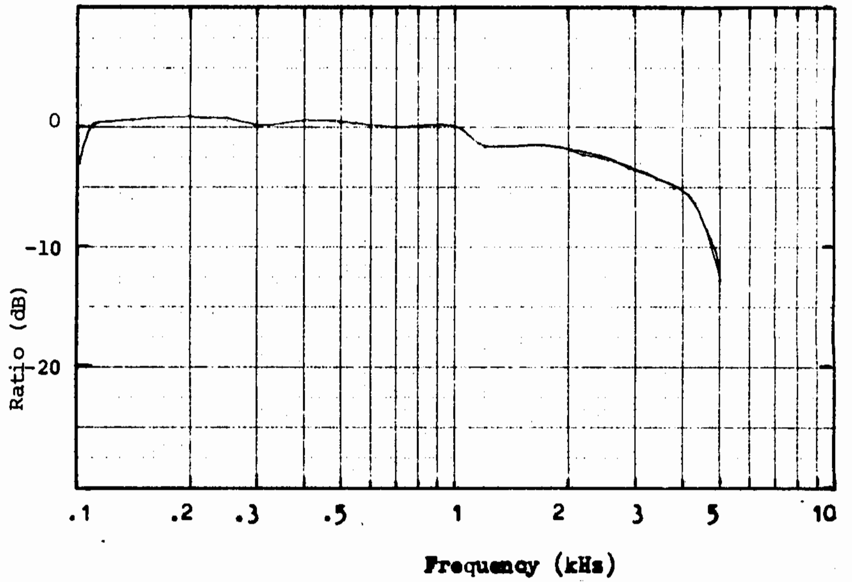


Fig 3-8 Discrepancy in response shape of the hearing aid when coupled to the FM system, with the same aid settings.

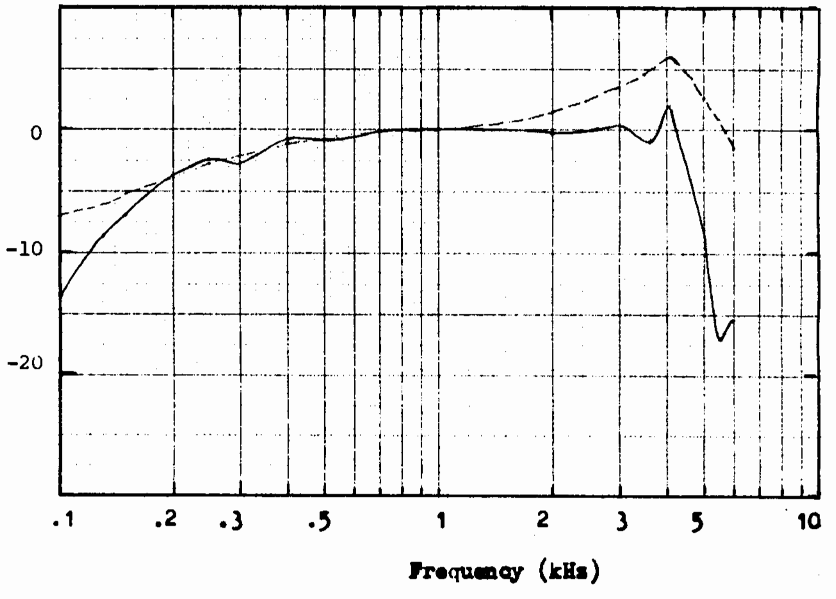


Fig 3-9 Input applied to the Phonak hearing aid amplifier

____ from transmitter and receiver and adaptor

---- from Phonak aid microphone

3.6 Adaptor output volume

The third function to be fulfilled by the adaptor is to adjust the signal level to be compatible with the audio input of the hearing aid. The task is made difficult by the action of the pre-emphasis and the automatic gain circuits in the transmitter. Figure 3-7 shows the acoustic gain of the hearing aid using its own microphone and via the FM system. The volume and tone settings were unchanged. The acoustic signal used corresponds in level and spectral power to that of "normal" speech, 100mm away from the speaker's mouth. The same signal is used for testing hearing aid response using its own microphone. With the adaptor volume set at mid-travel (6 dB below maximum volume), and with speech spoken at a "normal" level, the signal level from the adaptor is higher than that from the hearing aid microphone worn on the listener's ear. The transmitter microphone located close to the speaker's mouth would pick up a stronger speech signal. If the speech level is increased, the output from the hearing aid microphone increases proportionally. With the FM system, the AGC limits the output level. Figure 2-4 shows the maximum output obtainable from a receiver with a pure-tone input to the transmitter. At mid-frequencies, an output level of -5 dBV can be obtained. Reduction of maximum output at high and low frequencies is caused by the frequency response roll-offs in the circuits subsequent to the AGC. The necessary de-emphasis circuit in the adaptor greatly reduces the signal at higher frequencies. The solid curve in Figure 3-10 shows the maximum pure-tone output measured from the adaptor. A Phonak PPCL aid was used as a load for the adaptor. The results are in agreement with the adaptor transfer function shown in Figure 3-4 and the maximum output curve of the receiver shown in Figure 2-4. Figures 3-10 and Fig. 3-11 show curves of signal levels required to drive the hearing aids into saturation. The maximum output obtainable from the FM system (via the adaptor) is also shown. With the hearing aid "TONE" control set to L4, the adaptor output is sufficient to drive the hearing aid into saturation at low frequencies, and is insufficient at high frequencies except when the aid volume control is set to maximum. When the "TONE" control set to "N", the adaptor output is also insufficiently to drive the aid into saturation at low frequencies. This implies that when the hearing aid is used with the FM system, the volume control setting also affects the maximum output level. A reduction in volume decreases the maximum output obtainable from the aid. This is a result of the compression AGC being located ahead of the hearing aid volume control and compounded by locating the pre-emphasis circuit ahead of the AGC output point.

3.1.7 Summary

The hearing aid adaptor fulfils the function of providing de-emphasis, signal level interface and hearing aid microphone control and it imposes minimal compromise on system performance.

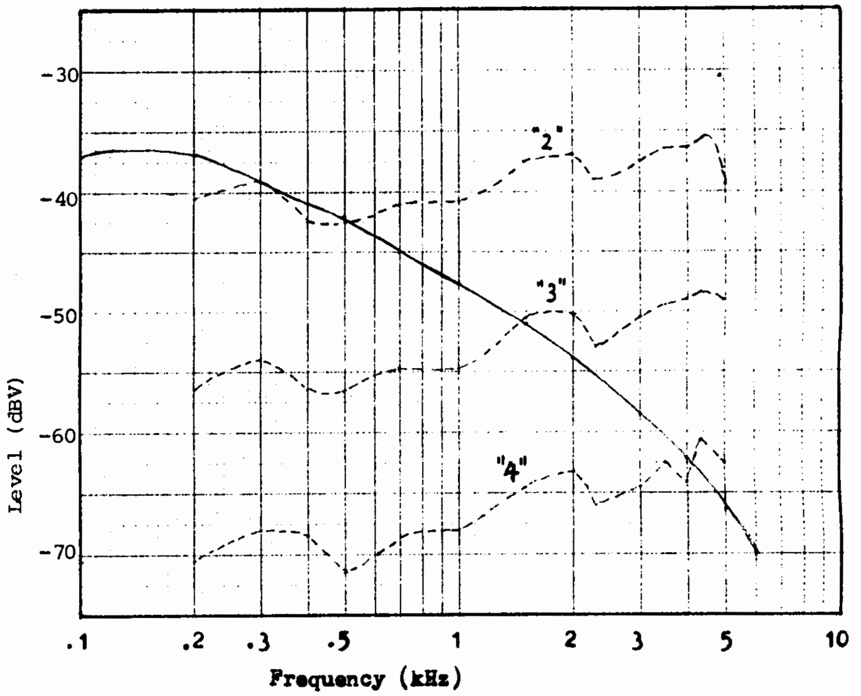


Fig 3-10 Adaptor output level and hearing aid sensitivity

_____ Maximum adaptor output for pure tone signals

..__-- Minimum input to saturate the hearing aid; dB-SSPL=136
Tone = L4 volume = "2", "3" and "4"

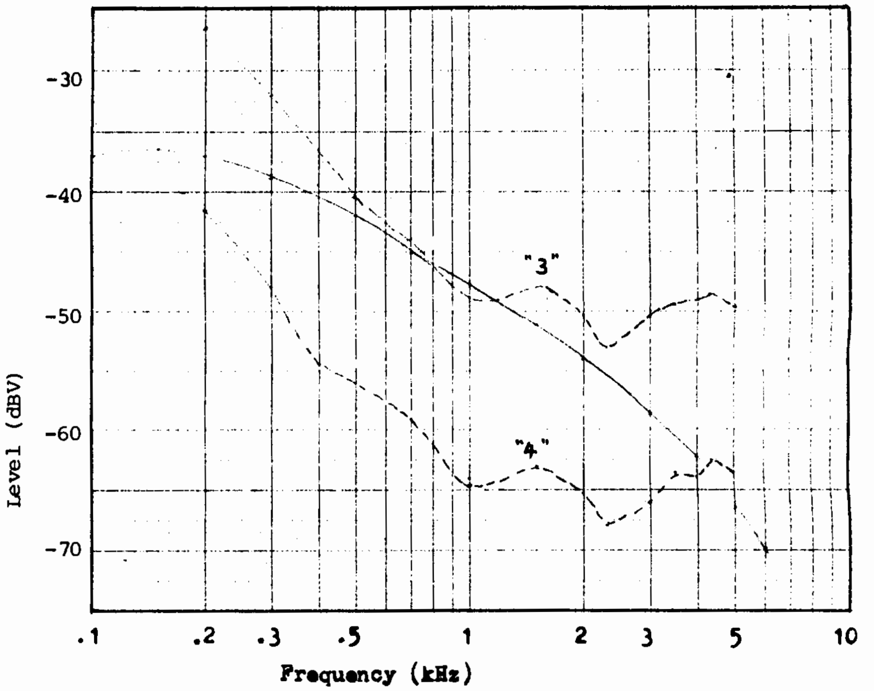


Fig 3-11 Adaptor output level and hearing aid sensitivity.
——— Maximum adaptor output for pure tone signals.
- - - - Minimum input to saturate the hearing aid, dB-SSPL=136,
Tone = N Volume = "4" and "3"

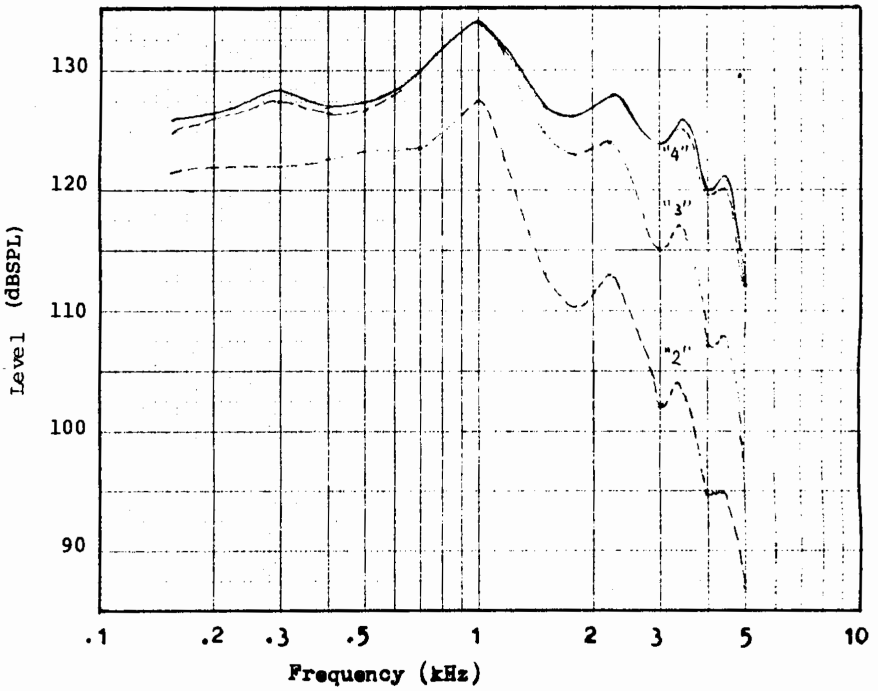


Fig 3-12 Maximum output of PPCL Phonak aid
(Settings: dB-SSPL = 136, TONE =L4)
______ using its own microphone
------ When coupled to the FM system.
aid vol. set to "4". "3", "2"

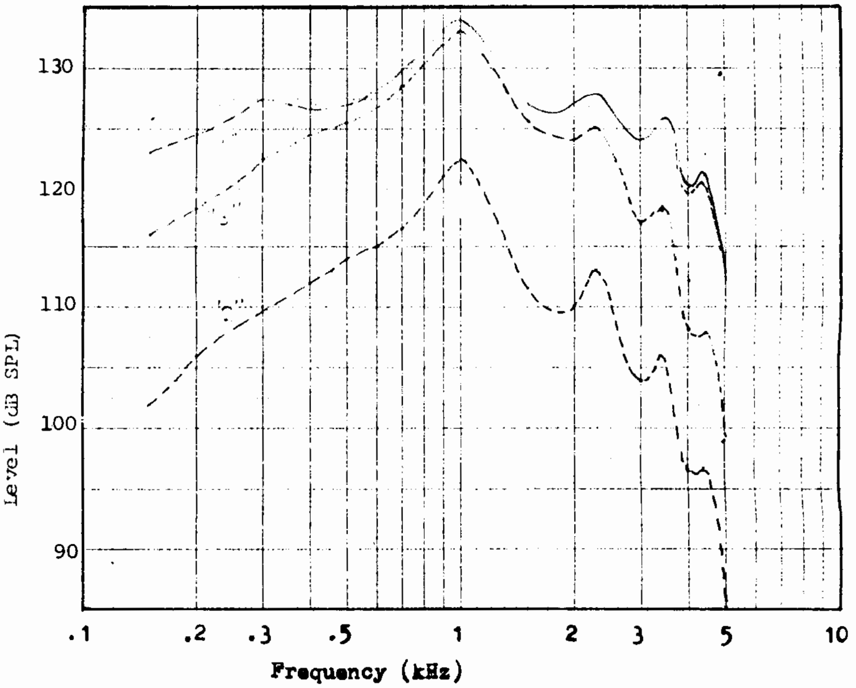


Fig 3-13 Maximum output of PPCL Phonak aid (settings: dB-SSPL = 136, TONE = N)

_____ Using its own microphone

------ When coupled to the FM system, aid Vol. set to "4", "3", "2"

4. Conclusions

The frequency response of the Calaid-FM system including hearing aids compares favourably with those of hearing aids in common use. When the Phonak hearing aid is coupled to the FM system, the resultant frequency response below 2 kHz is unaltered. At higher frequencies, the response would nominally be unaltered, but for a wider response spread similar to that of the Knowles BT 1754 microphones. Screening of microphones for 12 $\pm$ 3dB of response peaks would be desirable, though screening for 11 $\pm$ 3dB response peaks would be more practical in view of the probable statistical distribution.

The incorporation of the AGC and pre-emphasis circuits in the transmitter affect the operation of the hearing aid volume control. Reducing the volume control setting reduces the maximum output of the hearing aid.

The hearing aid adaptor successfully linked commercial hearing aids to the Calaid-FM system. The adaptor calibration procedure has since been eliminated by increasing the DC level of the receiver output. The value of R_1 has also been raised to increase the effective range of the adaptor volume control.

References

(1) Yip, J. (1982/83) Calaid FM phase 2 test records, various.

(2) Dillon, H. (1983) "Performance and optimisation of the CAL F.M. System". NAL internal report No. 41.

(3) Plessey Australia (1982) "Development of Calaid F.M. System for Department of Health". Technical progress report for the month ending 22.1.82.

(4) Byrne, D. (1977) "The speech spectrum - some aspects of it's significance for hearing aid selection and evalutaion" Brit. J. Audiol. 11, 40-46.

Figure 1-5 shows the improvement in system response using BT 1754 microphones. At 300 Hz, the response is improved by 3dB.

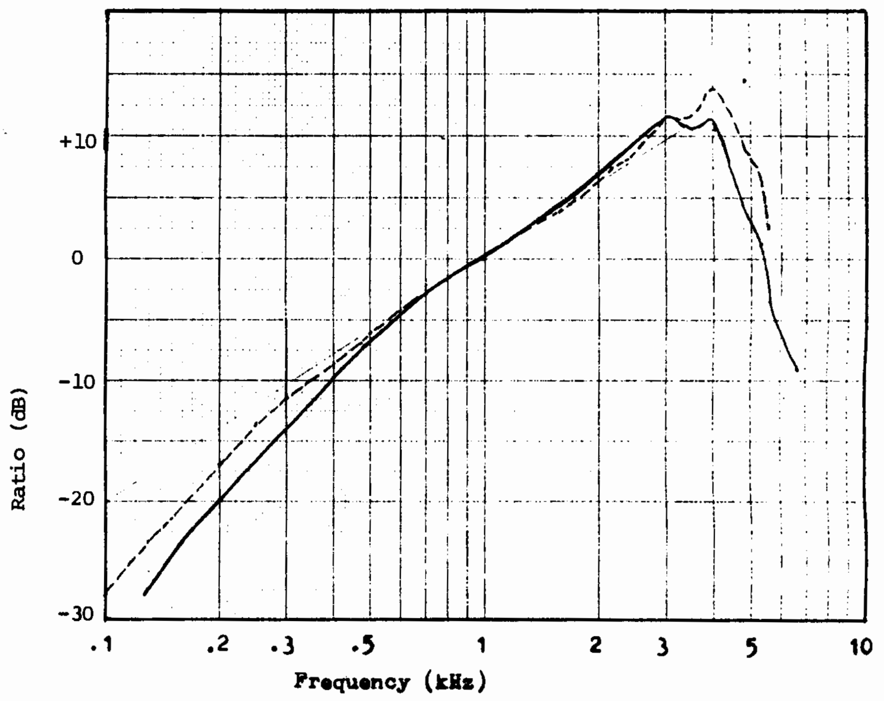


Fig 1-5 Transmitter and receiver response

--- Response using BT 1754 microphone

____ Response using BT 1751 microphone

-·-·- 6dB/Octave line

1.4 Response with de-emphasis.

To provide a nominally flat frequency response, the receiver output was connected to a de-emphasis network with a 3 dB break point at 58 Hz. The response of the transmitter-receiver with output de-emphasised is shown in figure 1-6. The use of BT 1754 microphones reduces the loss at lower frequencies (4 dB at 200 Hz) and extends the response at high frequencies. Microphones with responses peaking beyond 4 kHz have to be used for the transmitter to ensure an adequate high frequency response. Inherent in the design, the system response rolls off very steeply beyond the microphone response peak.

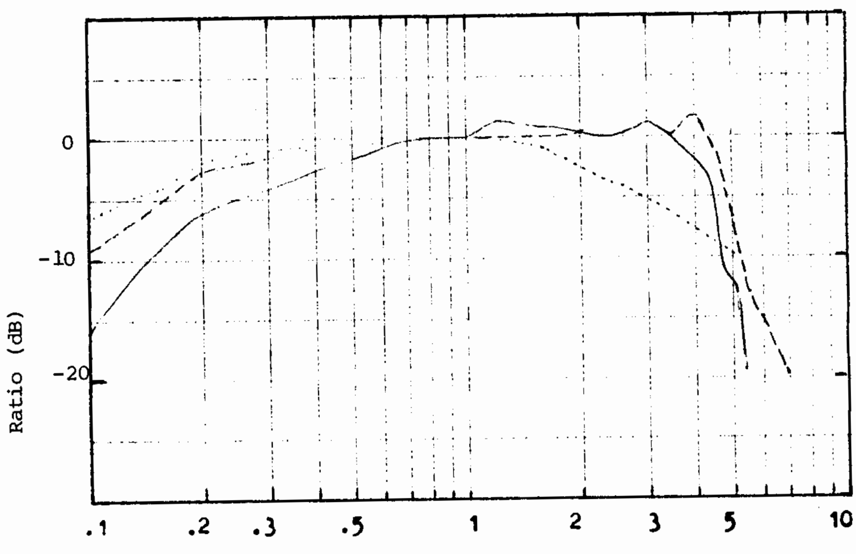


Fig. 1-6 Response of the transmitter-receiver with de-emphasis

——— BT 1751 microphone used

------ BT 1754 microphone used

.... direct electrical input

1.5 Circuit contributions

The dotted line of figure 1-6 shows the response of the system when a direct electrical input is applied. The behaviour at high frequencies is as expected since pre-emphasis above 2 kHz relies on the microphone response. The roll off at low frequencies is accounted for by the decoupling networks employed in the transmitter circuits. The main ones are the 160 Hz breakpoint at the input to the modulator and the 80 Hz breakpoint in the audio circuits.

Figure 1-7 shows the responses of these sections of the transmitter.

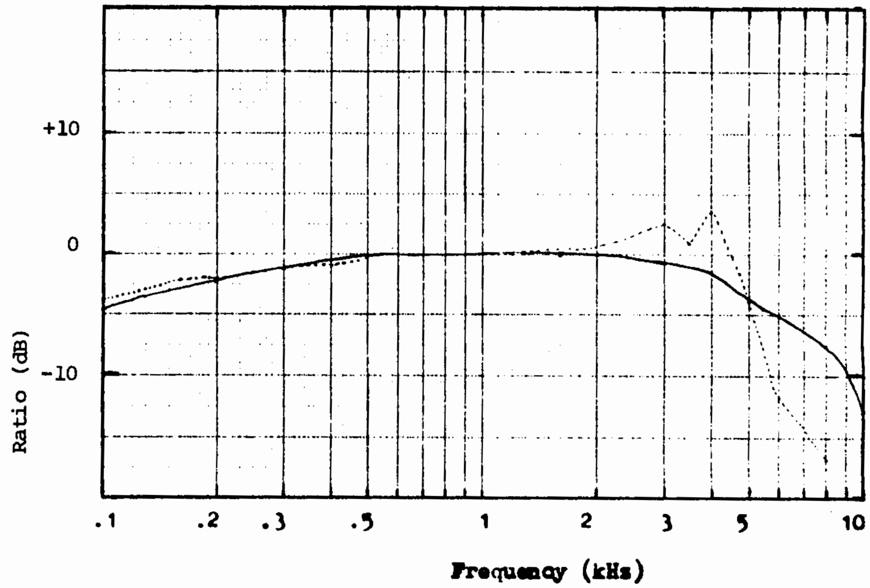


Fig 1-7 Transmitter circuit response

...... Microphone and audio ICs (measured with de-emphasis)

______ Modulator and output stages

The transmitter output section has a low frequency breakpoint around 150 Hz and a high frequency breakpoint around 4.5 kHz. Measurements with different capacitor values were not attempted as the capacitors are integrated in a hybrid circuit and are also related to the damping of the phase lock loop.

1.6 Summary

The CALAID FM system using the BT 1754 microphones provides good frequency response. The low frequency roll off (-3 dB at 200 Hz addtional to the noise cancelling system) is caused mainly by the coupling capacitors in the transmitter circuit. A trough in the microphone response (centred at 7 kHz of around -15 dB with respect to a peak at 4 kHz) and a -6 dB/octave roll off in the transmitter circuit with the break point at 4.5 kHz cause a steep cut off in the system response above 4 kHz.

The use of BT 1754 microphones improves frequency response at both high and low frequencies. Section 3.5 of this report indicates that it is desirable to screen the BT 1754 microphones for a high response peak of 11 ± 3dB at 4.5 ± 0.5 kHz.

To reduce resonse peaks and dips, the speech microphone should be located further back in the housing. (The microphone port should be at least 2 mm beneath the cap of the housing).

2 Transmitter AGC

Figure 2-1 shows the block diagram of the transmitter input stage. A2 is a voltage controlled attenuator. H(f) represents a high pass filter with a 6 dB/octave roll-off below 2 kHz. A3 is an amplifier with a fixed gain of 65 dB and G has a nominal gain of 10 dB. S selects the larger of the two inputs, Vo or G.Von to act on the generator C which sets the voltage Vc which controls the attenuator A2.

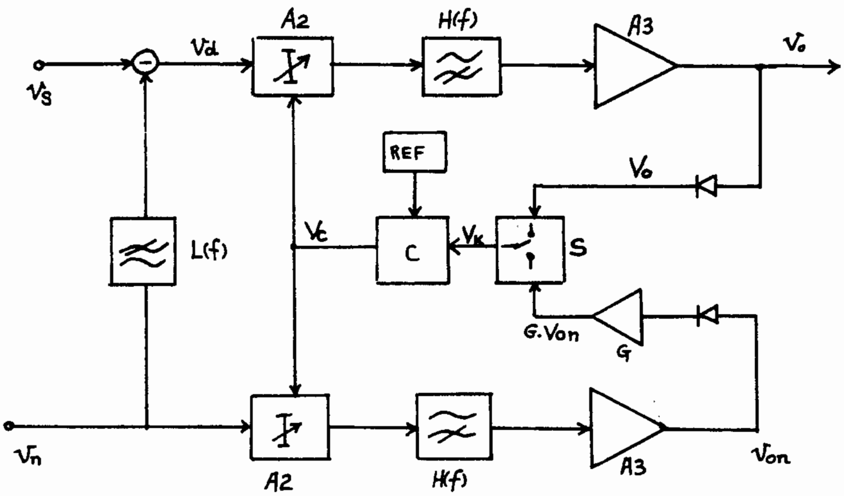


Fig. 2-1 Block schematic of the transmitter input circuit incorporating AGC.

The operation of the AGC circuit is governed by three signals: (1) a presettable internal reference: (2) the rectified speech signal Vo and (3) the rectified noise signal amplified by G(G.Von).

The following section analyses the operation of the AGC action. Vs and V_n denote the signals generated by the "speech microphone" and the "noise microphone" respectively. (Both microphones pick up speech and environmental noise, but at different relative amplitude and phase).

(1) Non-compression

When the preset internal reference is greater than either V^o or G.Von, the attenuator A2 provides a minimum attenuation, Ak, and therefore the output increases proportionally with input.

$$\tilde{V}o = \tilde{V}d.Ak.H(f).A3$$

$$= (\tilde{V}_s - \tilde{V}_n L(f)). Ak.H(f).A3$$

The value of Ak is related to the AMT setting.

(2) Compression

When either V_o or G.Von is greater than the internal reference, the AGC circuit operates in its compression mode keeping either Vo or G.Von constant (Vk).

When Vo is greater than G.Von, that is when Vo is more than 10 dB above Von, Vo is kept constant.

i.e. $Vo = V_k$ ..... (2)

When G.Von is greater than Vo, (that is when Vo is not greater than Von by 10 dB), then G.Von is kept constant by the AGC action.

ie G.Von = Vk ..... (3)

ie $Von = \frac{Vk}{G}$

The two channels are identical and therefore $\frac{Vo}{Von} = \frac{Vd}{Vn}$

Hence, from equation 3,

$$Vo = \frac{Vd}{Vn} \cdot \frac{1}{G} \cdot V_K \quad \ldots\ldots \ (4)$$

where

$$\frac{Vd}{V_n} \cdot \frac{1}{G} \leqslant 1$$

This means that, the speech microphone/noise microphone signal ratio affects the receiver output level. When the signal, Vd, is more than 10 dB above the noise signal, Vn, the output stays constant at the maximum volume. When the signal is less than 10 dB above the noise signal, the output decreases. The amount of reduction is equal to the amount that the speech signal is short of 10 dB above the noise signal. For example, if the speech signal is 3 dB short of 10 dB above the noise signal, the output will be 3 dB below the maximum.

Figure 2-2 shows some aspects of the AGC action. When the input signal to either channel is sufficiently high for the AGC to operate in the compression mode, the receiver output attains a maximum level of -4dBV. This is true only if the speech to noise signal ratio is above 10dB. If the ratio is below 10dB, the receiver output level decreases proportionally. When the signal ratio decreases beyond -5dB, (Fig 2-2(a)), signal reduction stops. This could be due to cross-talk between the two channels, caused by the noise cancelling circuit capacitively coupling the two channels. The amount of coupling decreases with increasing frequency. Figure 2-2(b) shows this frequency dependence. It is also significant to note that for a non-pure-tone signal, the RMS value of the maximum output may decrease. A speech-spectrum shaped random noise was used to test the AGC action. The results, shown in figure 2-2(c), depicts the same input-output relationship, but with the output level reduced by approximately 5dB. This may be explained by the fast attack and slow decay of AGC on signals with a high crest factor. The signal peaks incur the fast AGC attacks and together with the slow decay result in a reduction in the overall output level.

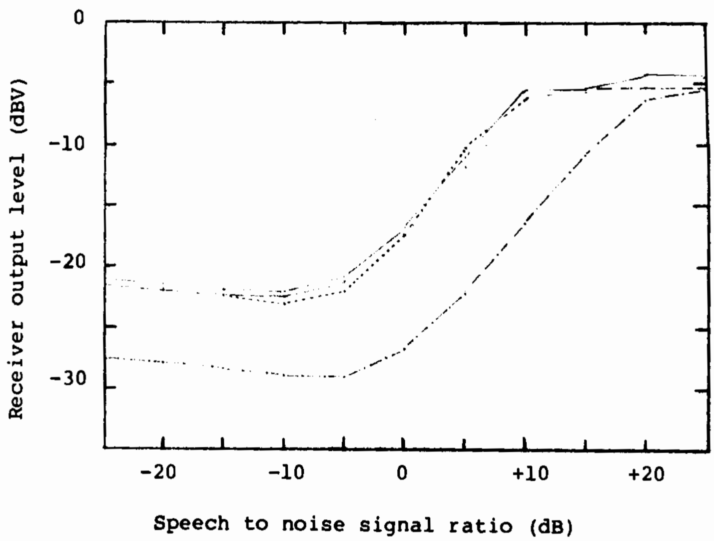
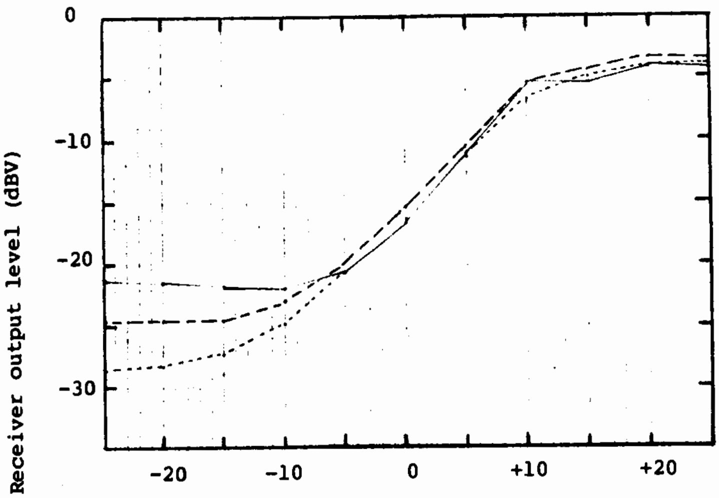


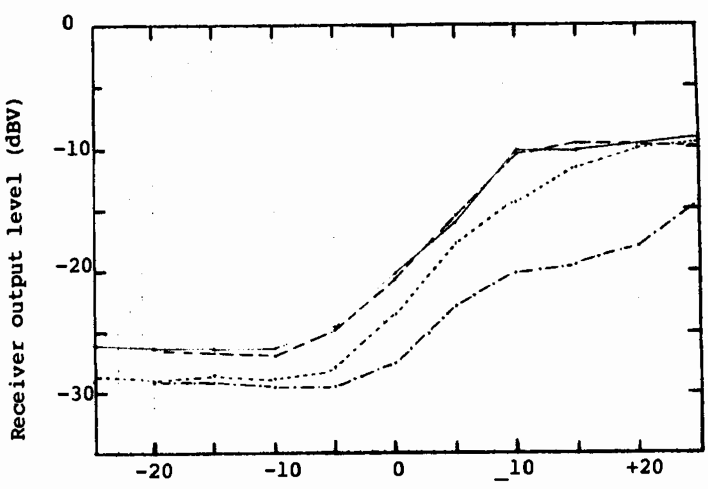
Fig 2-2 (a) Receiver output voltage at various speech to noise signal ratios. 1 kHz pure-tone used. Input level applied to the noise channel: ______ 100 mVRMS, 10mVRMS, 1mVRMS and _ . _ . _ _ 0.1mVRMS.



Speech to noise signal ratio (dB)

Fig. 2-2(b) Receiver output at various frequencies.
0dB = 100mVRMS

———— 1 kHz, - - - - - 2kHz and ·········· 4kHz



Speech to noise signal ratio (dB)

Fig 2-2(c) Receiver output voltage at various speech to noise signal ratio. Speech-shaped random noise used. Input level applied to the noise channel: ———— 50mVRMS,

- - - - - -10mVRMS, ·········· 1mVRMS and —·—·— — 0.1mVRMS

Combining equations (4) and (2), the following equation applies when the AGC circuit is in the compression mode

$$Vo = \min\left(1, \frac{Vd}{Vn} \cdot \frac{1}{G}\right) \cdot V_k \quad \ldots (5)$$

where $\frac{Vd}{Vn} \cdot \frac{1}{G} \leq 1$

(3) Combining equations (5) and (1)

$$Vo = \min\left[Vd.H(f).A_k.A3,\ \min\left(1, \frac{Vd}{V_n} \cdot \frac{1}{G}\right). V_k\right]$$

when $V_s \gg V_n$. Equation simplies to

$$Vo = \min(Vs.H(f).A_k.A3,\ Vk)$$

Then the system response can be expressed as follows:

$$V_o = \min[H_1(f).V_s.H(f).A_k.A3,\ V_k].H_2(f)$$

where $H_1(f)$ is the transfer function of the circuit preceding the AGC circuit. $H_1(f)$ is determined mainly by the microphone response (Fig. 1-3). $H_2(f)$ is the transfer function of the circuit after the AGC circuit. $H_2(f)$ is determined mainly by the transmitter output stages (solid line of Fig. 1-7).

$H_1(f).H(f)$ which occurs before the AGC feedback point determines the AGC threshold while $H_2(f)$ which occurs after the AGC circuit affects the output level when the AGC circuit is operating in the compression mode.

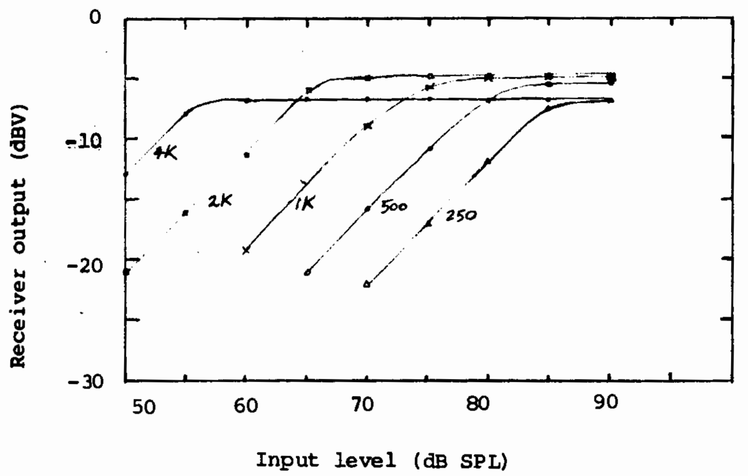


Fig 2-3 Input output curves for pure-tones at indicated frequencies. Noise microphone disconnected (Vs >> Vn)

Figure 2-3 shows the input-output curves for the transmitter-receiver system with pure tone input at indicated frequencies. The maximum output under compression varies with frequency. This is shown in Figure 2-4. The shape of the curve corresponds closely to the frequency response of the transmitter output stage (solid curve of Figure 1-7) confirming the proposition that the output stage is the main contributing factor. Figure 2-5 shows the compression threshold curve of the AGC circuit. The shape of the curve reflects the 6 dB/octave pre-emphasis preceding the AGC circuit feedback point. Deviation from the 6 dB/octave line corresponds to the response characteristic of the microphone and the audio circuit, (Fig. 1-7). The roll-off in the low frequencies results in the extra increase in compression threshold and a rising response between 2 kHz and 4 kHz results in an additional lowering of the compression threshold, for the frequencies in consideration.

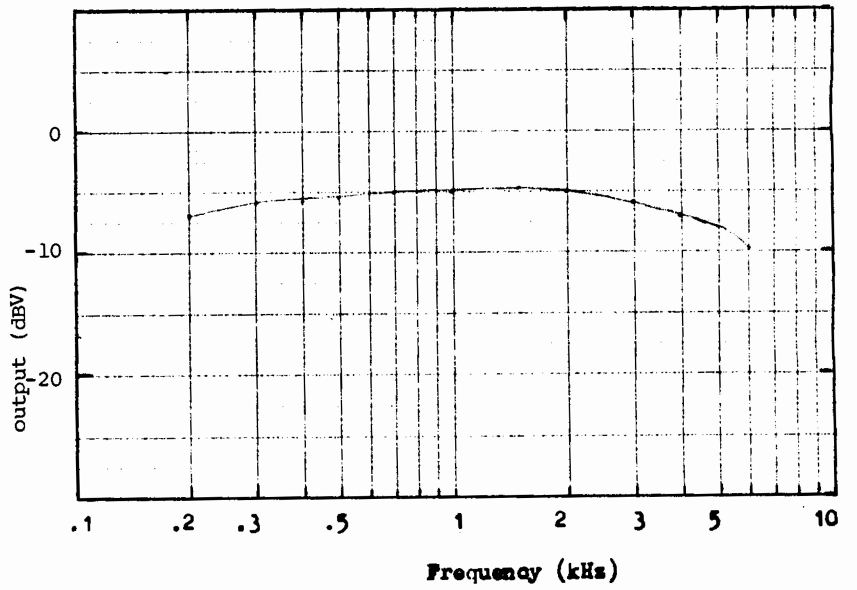


Fig 2-4 Receiver maximum output. Pure-tones used as test signals

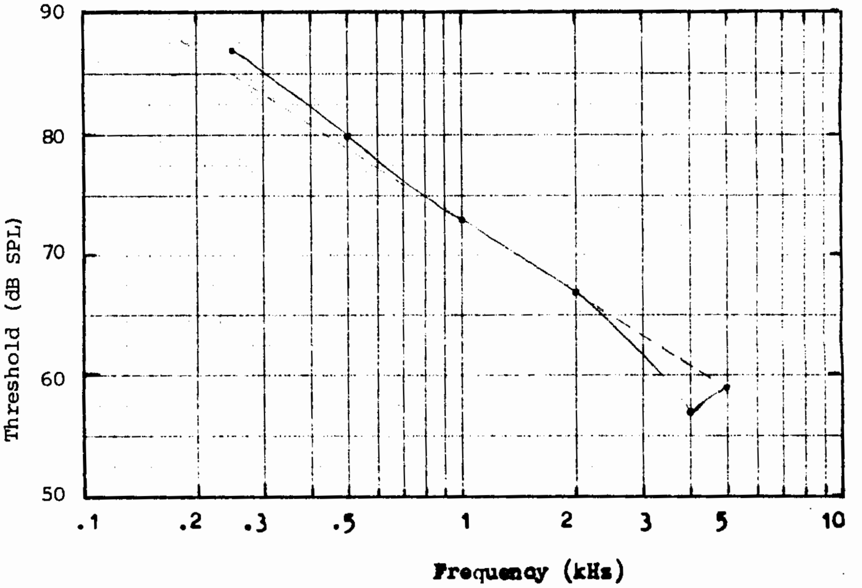


Fig 2-5 AGC circuit compression threshold

---- -6dB/octave line

3. Interface for Hearing Aids

The CALAID W hearing aid was specially developed to connect directly to the Calaid-FM receiver. The receiver output includes a switched DC level for hearing aid control. When the DC voltage is high (nominally 1.5V), the hearing aid microphone signal path is switched off and the hearing aid responds to the signal from the FM system. When the DC voltage is low, the hearing aid responds to both its own microphone and to the FM system. (The DC level is low when, the receiver output is in "C" mode). CALAID W also includes a de-emphasis circuit to match the pre-emphasis characteristic of the FM system (+ 6 dB/octave from 100 Hz to 4 kHz) to obtain an overall flat frequency response. Due to the pre-emphasis in the transmitter-receiver system, the receiver output was not designed to be used directly with commercial hearing aids that have the audio-input facility. A special adaptor has been developed to allow use of Phonak PPCL and PPC2 hearing aids (and other compatible types) with the FM system.

To interface commercially compatible hearing aids to the FM system, the adaptor performs the following main functions.

(1) Providing de-emphasis.

(2) Attenuating the receiver output level to be compatible with the hearing aid audio input level.

(3) Switching the hearing aid microphone on or off according to the DC level of the receiver output.

(4) Facilitating binaural fitting.

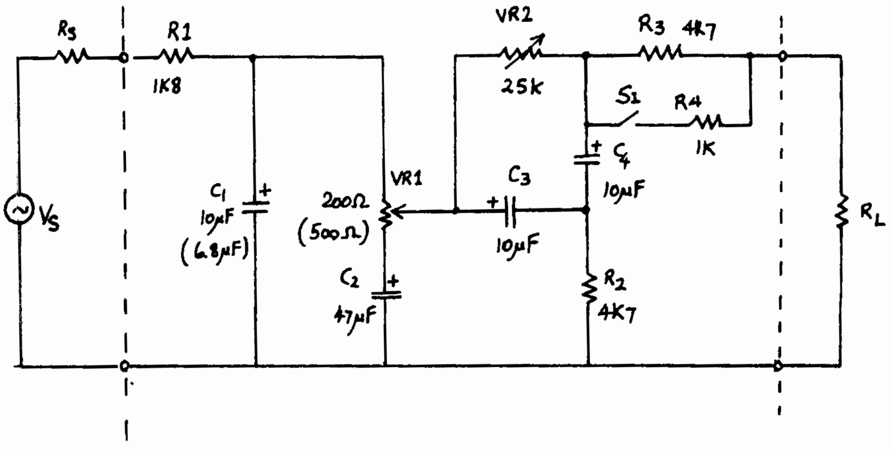


Fig 3-1 Circuit schematic of the hearing aid adaptor

( ): values for new version.

Referring to figure 3-1, which shows the circuit diagram of the adaptor, Rs represents the output impedance of the receiver. Measurements on a number of units give values between 35 and 70 ohms. This is relatively small in comparison with the value of R1. For binaural fittings, the circuit following VR 1 is duplicated, except for resistor R_2 and C_3.

3.1 Hearing aid microphone switching

The adaptor circuit has no DC path to ground. This allows the DC level from the receiver to be applied to the hearing aid input. When the DC level is high, the hearing aid microphone is effectively switched