ABSTRACT

Reduction of Acoustic Feedback Oscillations by Use of Spectrum Shifting.

by

Youssef Abdelilah

The problem of whistling noise or self-oscillation in public address systems operating in closed halls or rooms can be very disturbing to the listeners. This noise is sustained when the positive acoustic feedback, of the public address-room system, meets the oscillation criteria. This feedback system results from the acoustic sound signal reflected off the room walls and any other obstacles in the room, originating from the public address speakers and then re-entering the microphone.

When all of the audio signal components entering the public address system are shifted by a frequency increment Δf of 6 Hz, an increase of 5 dB in the useable signal level was achieved and whistling noise reduction is attained.

This thesis describes an apparatus for frequency shifting by small increments in steps of 1 Hz. The system is of simple implementation and effectively reduces the whistling noise and increases the value of achievable gain without introducing any speech distortions.

While the idea turned out to be not original, since it was described in the early sixties, the time for it may be now, because integrated circuits made it feasible in terms of cost, size and portability. Due to this, we feel that reintroducing the idea at this time may be fortuitous.

REDUCTION OF ACOUSTIC FEEDBACK OSCILLATIONS BY USE OF SPECTRUM SHIFTING

by

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APPROVAL PAGE

Reduction of Acoustic Feedback Oscillation by Use of Spectrum Shifting

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This thesis is dedicated to my parents and family...

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CHAPTER 1

INTRODUCTION

The idea for this thesis originated from the fact that when a person utilizing a hearing-aid, increases the amplifcation sufficiently, a whistling noise arises, which is very annoying. A similar problem arises in a Public Address(P.A.) System: When a speaker is talking into a microphone of a P.A. system in a hall or a room in front of an audience, and the voice signal is amplified to a certain gain, the positive acoustic signal feedback caused by the reflected voice signal out of the loudspeakers, off the room's walls and other obstacles into the microphone introduces a whistling or howling noise.

Our research consists of an actual realization of an experimental amplifying system, where we investigate the instability of this positive acoustic feedback that causes the whistling and offer a method to reduce it. The whistling occurs whenever the system starts oscillating, i.e. self-oscillation.

Our amplifying system consists of a two oscillator stages Single-Sideband (SSB) receiver, which takes an incoming acoustic voice signal (converted to an electrical waveform by a microphone) amplifies it to the proper level of audibility without introducing any whistling noise and maintaining a good intelligibility. The whistling occurs at certain frequencies where the positive P.A.-Room feedback causes oscillation. We are able to reduce this whistling by a continuous direct linear frequency translation with a small frequency shift Δf of all message frequency components without altering spectral shape and without creating any additional components. Figure 1.1 portrays a linear frequency translation system.

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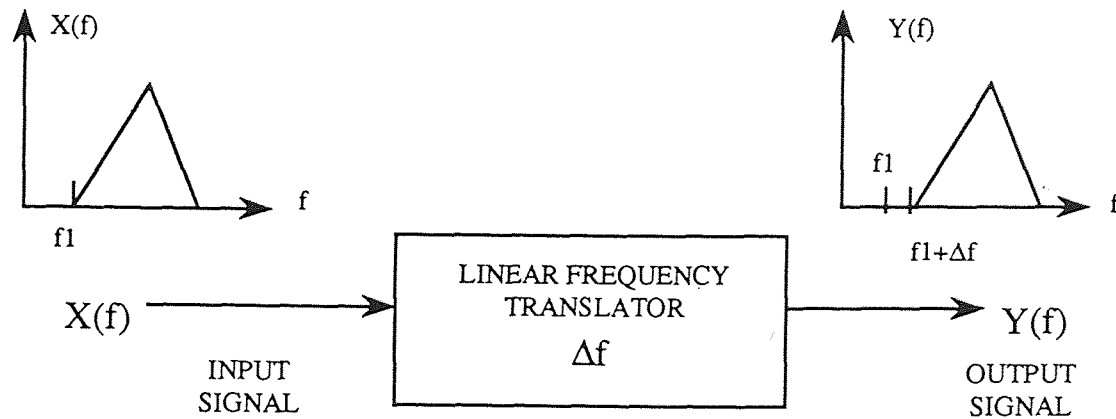
Figure 1.1: Linear frequency translator.

The frequency translation is accomplished by using SSB modulation. Figure 1.2 shows a prominent method, i.e., the two stages filter technique where an upward or downward frequency translation is accomplished by mixing(multiplying) an input waveform by a local oscillator, then filtering one of the sidebands and suppressing totally or partially the carrier. The resulting SSB waveform is again multiplied by another local oscillator whose frequency corresponds to the previous local oscillator frequency plus or minus the small frequency amount of shift desired. In other words, the difference between the two local oscillators frequencies represents the amount of frequency shift Δf desired.

The question asked is how much shift are we allowed before one pays a price in speech quality. The phone companies maintain that a shift as much as ± 10 Hz is possible without any recognizable effect on speech intelligibility. or subjective quality, while in the literature it is reported that for shifts between ± 5 and ± 50

Hz the quality suffers but the intelligibility is not degraded significally [1]. In our research, we investigated the shifting problem as well.

Input ─→ M1 (×) ─→ [BPF] ─→ M2 (×) ─→ [LPF] ─→ Output
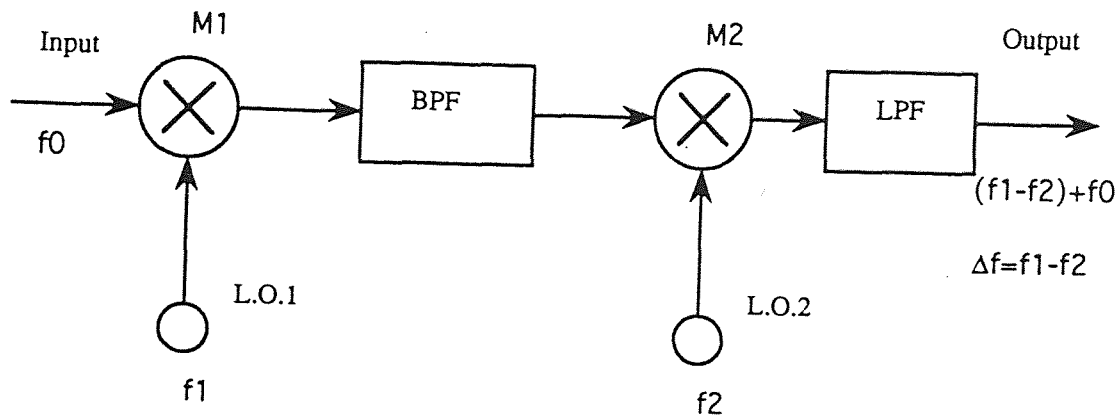
Figure 1.2: Frequency translation using two-stages filter techniques

The results of this research is this thesis, organized as follows: First, a chapter where some background on the basic theory of acoustics, sound theory, feedback theory, and frequency translation utilizing SSB modulation is reviewed. Second, a chapter containing surprising material that surfaced on October 15 th, 1991. That is when, our Attorney at Law Klauber & Jackson informed us about some published papers and patents related to our research and of which we were not aware. It is to be noted that our experiment is based on Dr. Klapper's original idea and a preliminary search of Dialog was made, but no publications were found. Nonetheless, in this chapter we included the work presented in those papers and patents, which we made reference to later on, and used it as a comparison to our results. Next, a major chapter where we describe in detail our experimental work

setup. While the idea turned out to be not original in that it was described in the early sixties, the time for it may be now, because integrated circuits make it feasible in terms of cost and size. Due to this, we feel that reintroducing the idea at this time may be fortuitous. The final chapter states our results and some suggestions for further research, plus a bibliography containing all relative references.

CHAPTER 2

BACKGROUND

2.1 Acoustic Phenomena

It is appropriate to briefly review acoustic phenomena and characteristics of sound
to establish some understanding of its nature and behavior.

Acoustics is the study of sound. It is concerned with the generation, trans-
mission, and reception of sound waves. These sound waves are created when energy
causes a disturbance in a medium. For example, when the drum-stick strikes a
drum, it disturbs the surrounding air (the medium). The outcome of that distur-
bance in air is the sound of the drums that we can hear. This sound is propagated
by air molecules or other material through successive displacements that correspond
to the original disturbance. The sound propogates in form of longitudinal waves.
It can undergo diffraction, in which bending occurs around obstacles. The amount
of diffraction is related to wavelength(the distance sound travels through one com-
plete cycle of pressure change, and is the physical measurement of the length of
one cycle)[2]. Longer wavelength diffract more than shorter ones. Thus, high fre-
quencies are considered to be more directional in nature. For instance, if one holds
a book in front of a loudspeaker, high frequencies will be blocked by the barrier,
i.e. the book, while longer wavelength or low frequencies will go around it. Sound
may refract, that is it changes direction with temperature. Sound-waves bend away
from warmer temperature to colder ones. While not a major problem indoors, it
represents difficulties for an outdoor P.A. System applications. Sound may be re-
flected and absorbed in a variety of ways. The mere passage of sound through air

acts to attenuate sound energy. High frequencies are more prominently attenuated in air. Humidity affects air attenuation; specifically, wet air absorbs sound better than dry air.

There is no end to intricacies of acoustics and the twists of faith that sound may experience as it travels through its medium. After sound enter its medium, it may be reflected, absorbed, or neither. In free field, outdoors for example, the sound continues to radiate outward from its source. The sound pressure level decreases with the inverse square of the distance from the source [2]. In an enclosure such as an enclosed room, some energy is reflected back creating echoes and reverberation. This is especially so in a highly reflective room with glass and tile, while an absorptive room with carpet and curtains has a short reverberation time with few echoes. In general, rooms have a preferred set of frequencies augmenting some over others according to the room dimensions and geometry and producing standing waves(sound waves with stationary low and high pressure regions, nodes and antinodes) at specific places in the room. In other words, the room frequency response, particularly at low frequencies, is a function of the location of the loudspeakers and listener, and any other barriers in the room. In particular, some of the acoustic voice signals coming out of the loudspeakers reflect and re-enter the microphone, hence, creating a feedback system which is reviewed in the next section.

2.2   Public Address Room Feedback System

This feedback system results from the acoustic sound signal reflected off the room walls and any other objects or obstacles in the room, originating from the loudspeakers and then re-entering the mike. The feedback system is portrayed in figure 2.1 where an example of a voice signal reflected off the room walls is shown.
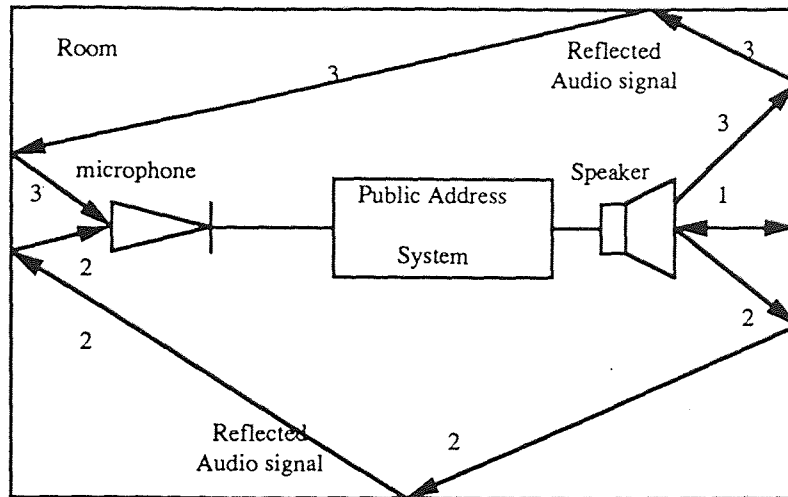
Figure 2.1: Public address-room feedback system.

The loudspeaker produces an amplified version of the sounds picked-up by the microphone. These sounds are the sum of the input audio signal and reverberated sounds in the room, that are amplified all together. How strong these echoed signals are depends upon the distance the signals travel between the speakers and the microphone as well as the absorbtion by the medium. Because of the attenuation in the air and absoption by the walls, the larger the distance, the weaker the signal that reaches the microphone. Furthermore, due to finite propagation speed of sound waves, there is a time delay between the signal produced by speaker and that picked-up by the microphone. This represents an acoustic feedback system which can be either positive or negative feedback. This accoustic feedback system which consists of a PA-room system is represented in the block-diagram form in figure 2.2 as a positive feedback system.

External Audio Signal → Total Audio Input to Mic. → Σ (+ +) → Public Address System → Loudspeaker Audio Output

Reflected Audio Signal

(a)

Input
$X(\omega)$ → Σ (+ +) → Forward-Path Gain $A(\omega)$ → Output $Y(\omega)$
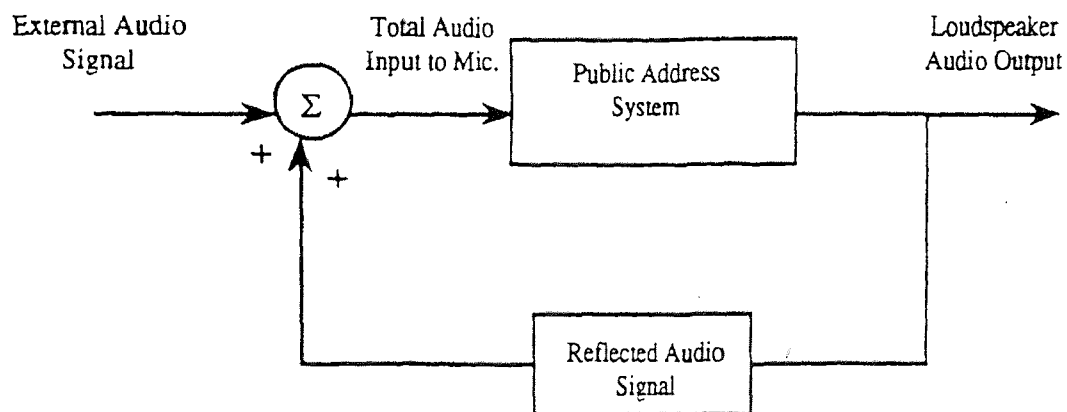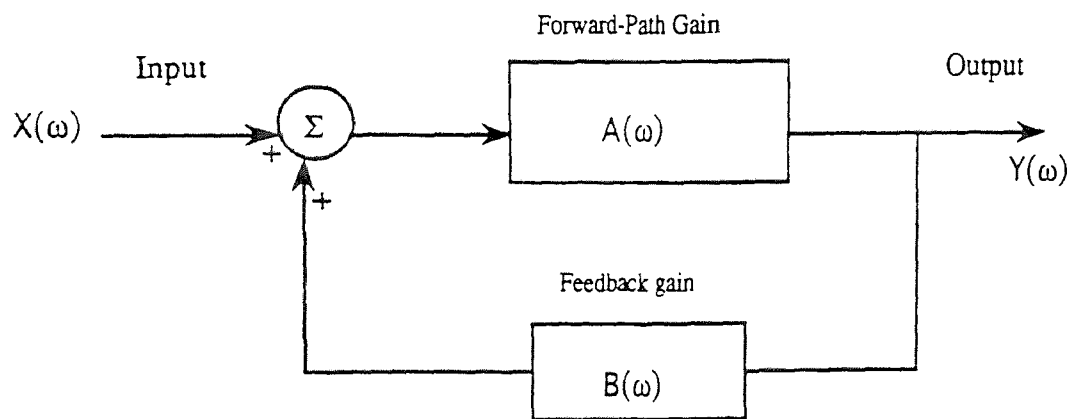
Feedback gain $B(\omega)$

(b)

Figure 2.2: Acoustic feedback system:(a)System model, (b) Electronic block diagram.

Here $A(\omega)$ is the forward loop gain representing the voice signal amplification gain, and $B(\omega)$ is the feedback gain representing the attenuation of the delayed output propagation signal which is added to the external input voice signal $X(\omega)$; all are function of frequency. The effect of this feedback may either increase or decrease the total gain. According to Nyquist theory in control systems, the overall transfer functon of this feedback system is given by:

$$H(\omega) = \frac{Y(\omega)}{X(\omega)} = \frac{A(\omega)}{1 - B(\omega)A(\omega)} \qquad (2.1)$$

where the product $A(\omega)B(\omega)$ is called the open-loop transfer function, a complex quantity of magnitude $|A(\omega)B(\omega)|$, and phase of $\arg(A(\omega)B(\omega))$. Since the forward path gain $A(\omega)$ and feedback path gain $B(\omega)$ depend on frequency, the feedback can increase the total gain at one frequency and decrease it at another. At some particular frequency ω_0, the frequency dependent P.A.-room feedback system starts to oscillate and creates an undesired whistling or squealing noise.

It is well known from Nyquist stability criteria that such system will oscillates if the following conditions are met:

$$1 - A(\omega_0)B(\omega_0) = 0 \implies A(\omega_0)B(\omega_0) = 1 \qquad (2.2)$$

$$\implies \begin{cases} |A(\omega_0)B(\omega_0)| = 1 & \text{open loop gain mag.} \\ \arg(A(\omega_0)B(\omega_0)) = 360°n & \text{n=0,1,2,....} \end{cases} \qquad (2.3)$$

To recapitulate, we state that in a positive feedback system given by the overall transfer function equation (2.1), if the open loop gain has a total phase shift multiple of $360°$ at ω_0, the system will oscillate at ω_0 provided that the open loop gain is unity as stated in equations (2.2)and (2.3). If the gain is less than unity at that frequency where the phase shift is $360°$, the system will be stable, whereas if the gain is greater than unity, the system will be unstable. Thus to eliminate whistling or squealing of the acoustic feedback in a P.A-Room system,

one or both of the conditions in equation (2.3) must be avoided, and this thesis presents a method to reduce the whistling noise in this acoustic feedback problem. This will require some theoretical background given here, and experimental study.

2.3   Frequency Translation

Before discussing a particular frequency translation technique, it is worth defining what we are attempting to achieve. Frequency translation is provided by using any modulation technique: Amplitude Modulation (AM) and its hybrids, Phase Modulation (PM), or Frequency Modulation (FM). But in all modulation operations, the resultant modulated signal contains more spectral components than the original message waveform, and in many cases bears no resemblance to the spectrum of the original message waveform.

The desire here is to realize a direct linear frequency translation of all the message signal frequency parameters without altering the spectral distribution and without creating any additional terms. Such process is often refered to as Single Side Band modulation (SSB) or true Linear Modulation (LM). Amplitude Modulation (AM) consists of a carrier plus an Upper Sideband (USB) and Lower Sideband (LSB), and to create SSB both the carrier and one sideband are removed so that only the Upper Sideband (USB) or Lower Sideband (LSB) remains.

The three common methods of SSB modulation are the filter technique, the phase- shift technique also called the Hilbert transform method, and the Weaver technique. These last two techniques are also known as Quadrature frequency modulation. Today, the filter method is used almost exclusively in analog circuit implementations while the phase-shift and Weaver methods are favored in digital circuit implementations [1,3].

Filtering Technique:

In the filtering technique an upward or downward frequency translation can be achieved by mixing (multiplying) the input waveform with a local oscillator whose frequency corresponds to the desired frequency shift, and selecting the appropriate uppersideband (USB) or lowersideband (LSB). Provided that the local oscillator frequency is greater than twice the highest frequency of the input signal, complete rejection of the unwanted sideband and carrier is possible (assuming ideal brickwall filters) as displayed in figure 2.3. If however the local oscillator frequency is less than twice the input frequency, the sidebands partially overlap, and perfect frequency translation is impossible. With practical filters having a finite transition region, the translation of low-frequency components cannot be realized without either undue suppression of the low-frequency terms in the wanted sidebands, or poor rejection of low frequency terms in the unwanted sideband.
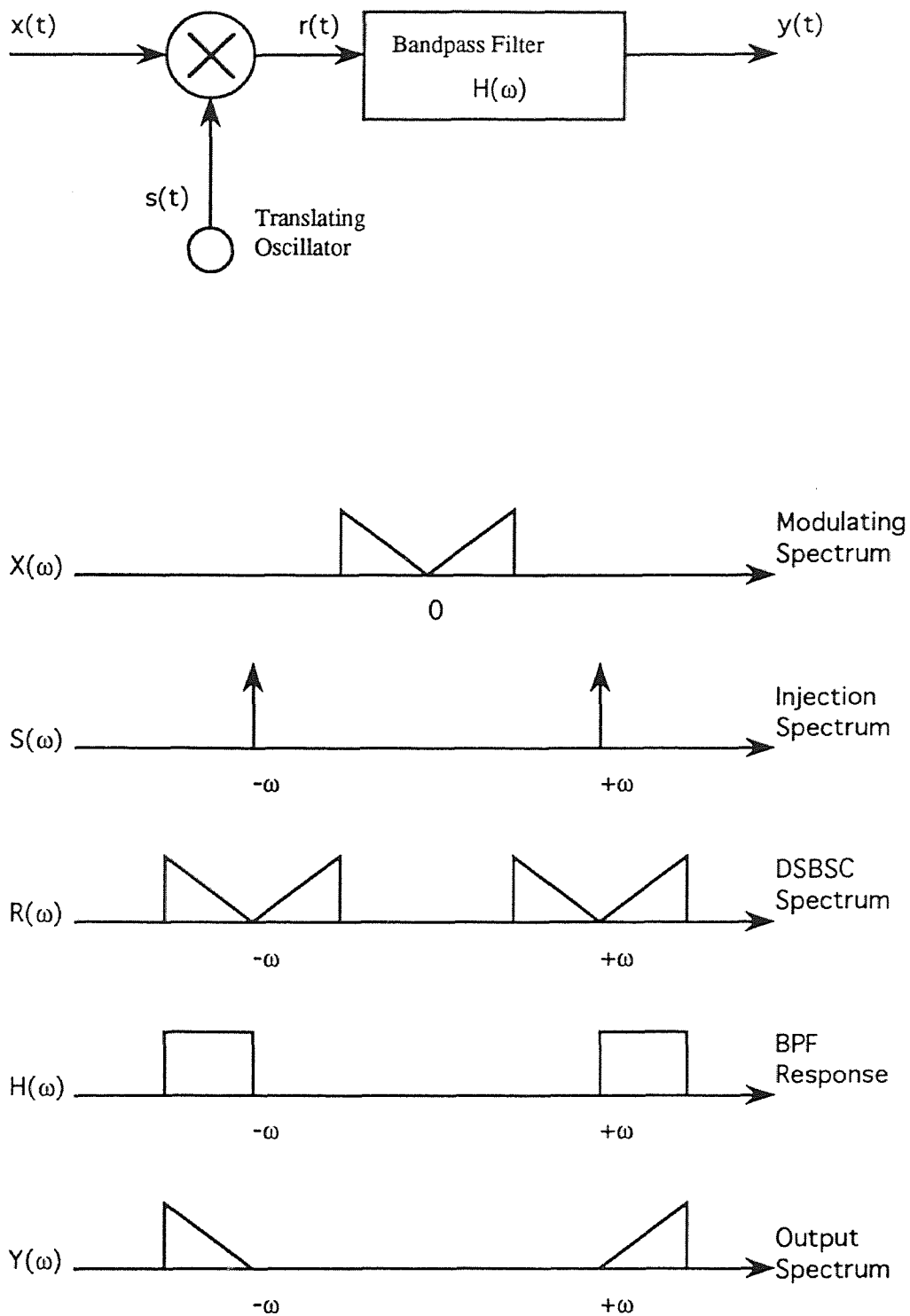
Figure 2.3: Filtering technique of SSB frequency translation.

This is the major drawback of the filter technique for frequency translation, coupled with the restriction on the use of low-frequency local oscillators, and hence small frequency shifts, due to sideband overlap. This limitation can be overcome by using two stages of filter-based frequency translation as shown in figure 2.4 with the algorithm depending on the sign of the small shift Δf: translation is upward if Δf is positive and downward otherwise.
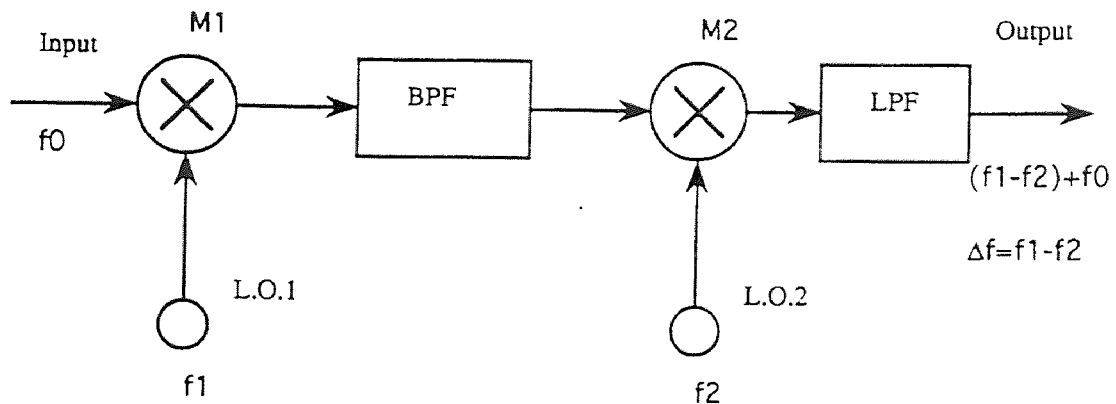
Figure 2.4: Two-stages SSB frequency translation for small freqency offsets, using filtering technique.

Phase-shift or Hilbert transform technique:

The Hilbert transform is just a phase shifter that shifts a bandlimited signal by $-90^\circ(\frac{-\pi}{2}rad)$ to generate the exact quadrature of the waveform: in other words, it changes the phase of every frequency component by -90°. The Hilbert transform

is mathematicaly represented by the following equation[1] :

$$\hat{x}(t) = \frac{1}{\pi} \int_{\infty}^{\infty} \frac{x(\alpha)}{t - \alpha} d\alpha = \lim_{\epsilon \to 0} \left(\int_{\infty}^{t-\epsilon} \frac{x(\alpha)}{t - \alpha} d\alpha + \int_{t+\epsilon}^{\infty} \frac{x(\alpha)}{t - \alpha} d\alpha \right) \qquad (2.4)$$

where the Cauchy principal value is taken because the integral is undefined at $\alpha = 0$. Physically, this operation is performed by passing the signal x(t) through a filter with an impulsive response h(t)=1/t, producing the output $\hat{x}(t)$. The transfer function of the Hilbert transform in the frequency domain H(ω) is displayed in figure 2.5 and defined by the following equation:

$$H(\omega) = -jsgn(\omega) = \begin{cases} -j & \text{if } \omega > 0 \\ j & \text{if } \omega < 0 \end{cases} \qquad (2.5)$$
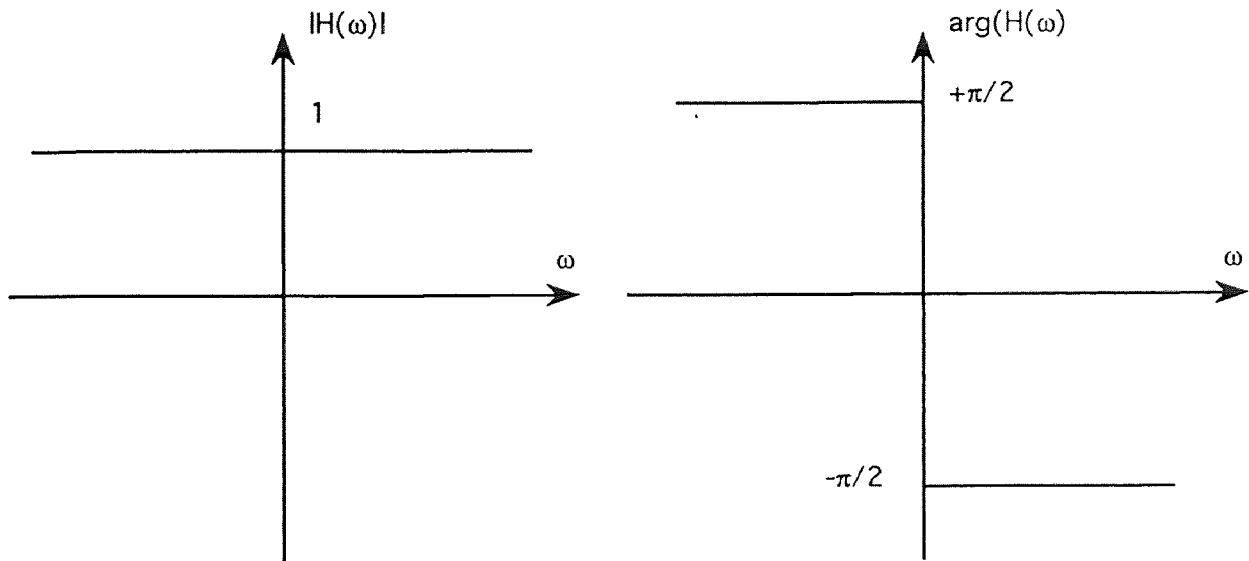
Figure 2.5: Transfer function of an ideal Hilbert transform.

Note that a practical implementation of an ideal 90° phase-shifter (Hilbert transform) is unrealizable. The wider the frequency bandwidth of the waveform, the greater the difficulty in realising an accurate quadrature replica but at most,

an approximation of the phase-shifter over a finite band is possible. The frequency translation using the phase-shift or Hilbert transform technique is represented in figure 2.6. In this method, a SSB signal is generated by synthesizing a quadrature (Q) component which leads or lags the in-phase (I) component by 90°.
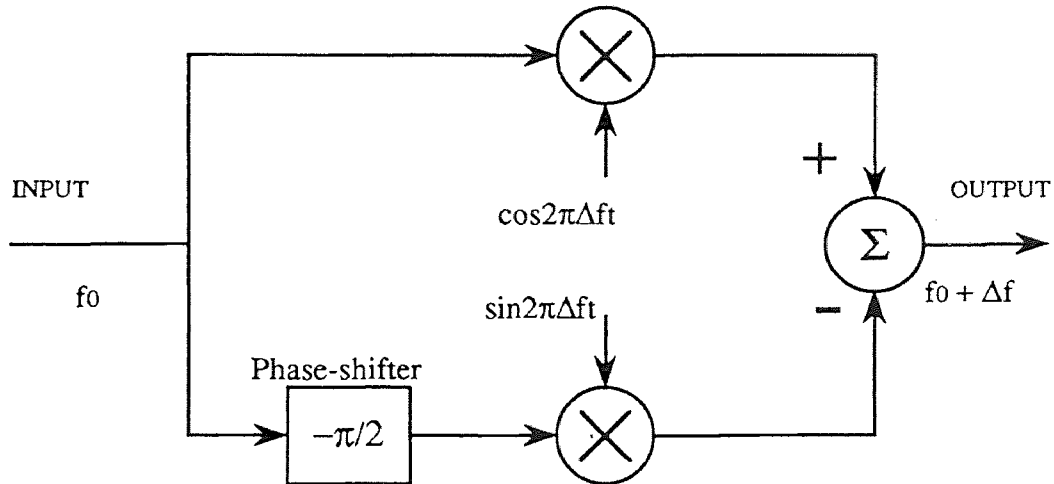
Figure 2.6: SSB frequency translator using Hilbert transform.

The circuit in figure 2.6 performs the mathematical function:

$$y(t) = x(t)cos(2\pi\Delta ft) \pm \hat{x}(t)sin(2\pi\Delta ft) \tag{2.6}$$

where x(t) is the signal to be translated, $\hat{x}(t)$ is the Hilbert transform or quadrature version of the message signal, y(t) is the translated output signal and Δf is the desired frequency shift. Each multiplier generates upper and lower sideband terms, see figure 2.6. However, one of the sidebands cancels out at the final combiner stage. depending on the sign of the combiner (adder). If the two terms are added, then the lower sideband is selected and downward frequency translation is achieved.

If they are subtracted, the message is translated upward in frequency.

In practice, there is usually some error (amplitude or phase) in the quadrature waveform generation process with the result that the unwanted sideband from the mixing process is not completely suppressed.

The ability of the Hilbert transform technique to realize small frequency shifts is brought about by the elimination of the output filter stage and low frequency oscillators. A double mixing quadrature frequency translation by small offset can be implemented as shown in figure 2.7 where the frequency shift corresponds to the difference in frequency between the two local oscillators.
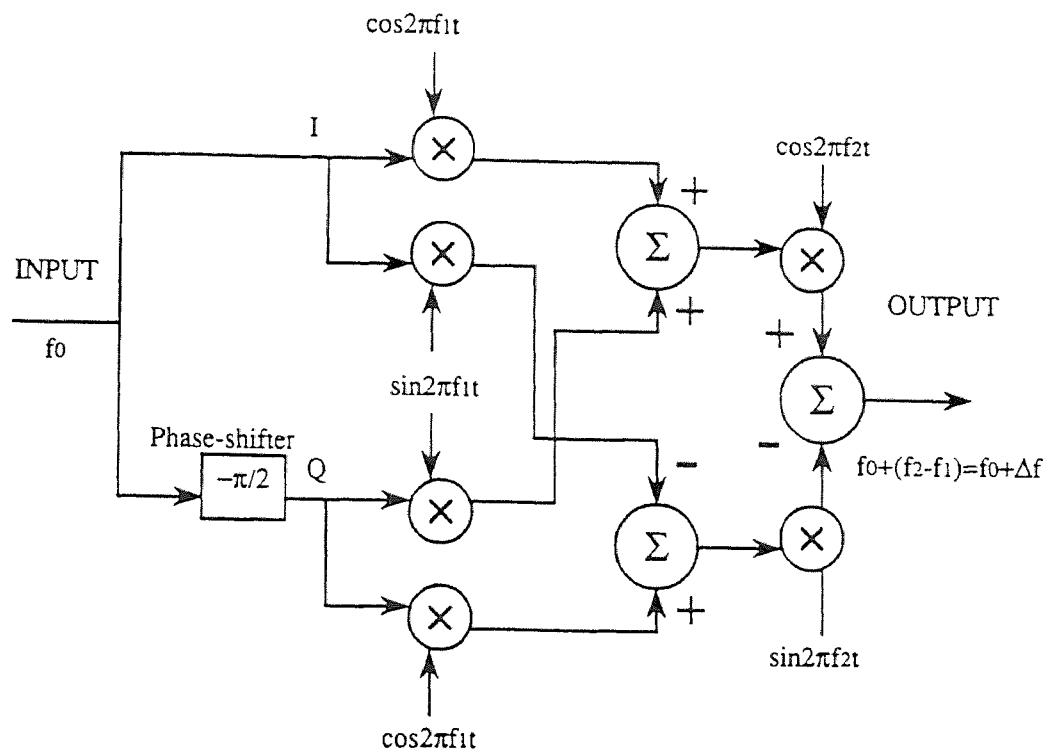
Figure 2.7: SSB frequency translation for small frequency offset using Hilbert transform.

This method is more preferable for digital implementation of small frequency translation when using recursive techniques.

The main drawback of the phase-shift or Hilbert transform for small offset translation is in the circuit difficulty and quadrature imbalance which may cause a poor sideband suppression. This results in residual sideband components that may cause distortions to neighboring signal components of the frequency band occupied by the translated waveform signal components. The problem is solved by using a frequency conversion process known as the Weaver method. This technique removes the need for wideband quadrature generators and is easy to implement.

Weaver technique:

A schematic of the frequency translations using the Weaver technique is given in figure 2.8.
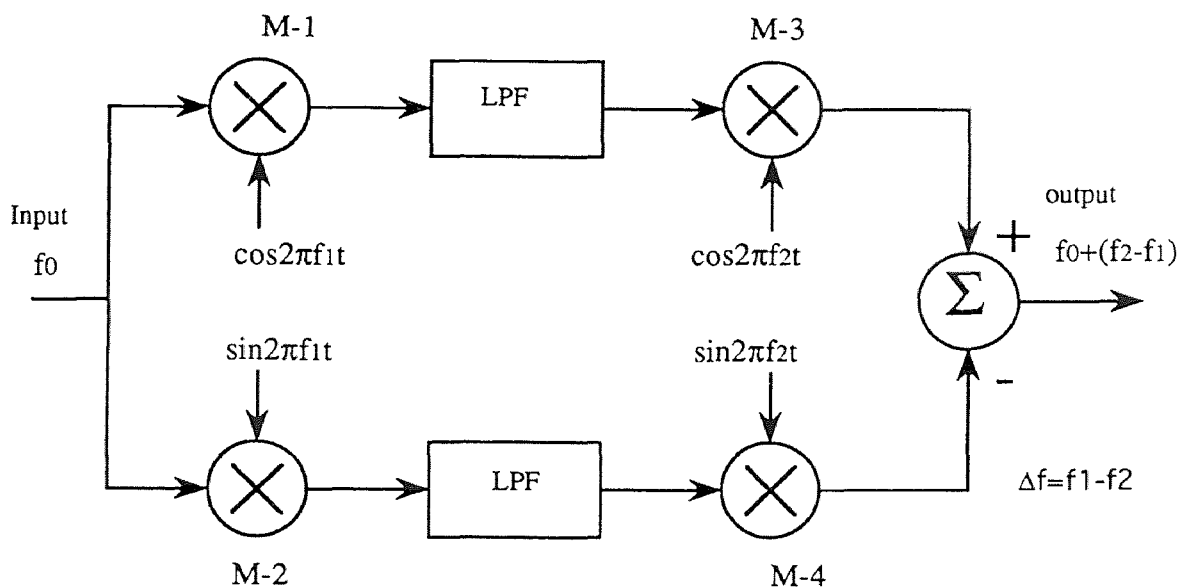
Figure 2.8: Weaver method of frequency translation.

The Weaver technique operates by first down-converting the input signal using quadrature local oscillators, with the local oscillator frequency chosen such

that the lower sideband folds approximately, symmetrically about zero Hertz. After filtering to remove the upper sideband (USB) from the mixing process, two quadrature lower sideband (LSB) signals remain. These two signals are then modulated with a further pair of quadrature oscillators in an identical manner to the Hilbert transform method described above. When the mixer outputs are combined, one of the sidebands from each mixer is suppressed (assumming perfect quadrature oscillators), giving the desired frequency translation output waveform. The translation frequency is simply the difference in frequency between the two quadrature oscillator pairs.

Mathematical analysis of the Weaver method:

Reference is made to figure 2.8. Let the input signal x(t) be given by:

$$x(t) = \cos(2\pi f_0 t) \tag{2.7}$$

After multiplication of the input signal by the first quadrature carrier f_1, the mixers M-1, and M-2 output is given by r_1 and r_2 respectively:

$$r_1(t) = x(t) \cos(2\pi f_1) t \tag{2.8}$$

$$r_1(t) = 1/2 \cos 2\pi (f_0 - f_1) t + 1/2 \cos 2\pi (f_0 + f_1) t \tag{2.9}$$

$$r_2(t) = x(t) \sin(2\pi f_1) t \tag{2.10}$$

$$r_2(t) = 1/2 \sin 2\pi (f_1 - f_0) t + 1/2 \sin 2\pi (f_1 + f_0) t \tag{2.11}$$

After lowpass-filtering, the USB is suppressed and the outputs of both LPF are given by w_1 and w_2:

$$w_1(t) = LPF(r_1(t)) = \cos 2\pi (f_0 - f_1) t \tag{2.12}$$

$$w_2(t) = LPF(r_2(t)) = \sin 2\pi (f_1 - f_0) t \tag{2.13}$$

After multiplication of the LPF outputs by the second quadrature carrier f_2, the mixers M-3, and M-4 outputs is given by s_1 and s_2 :

$$s_1(t) = r_1(t) \cos(2\pi f_2) t \tag{2.14}$$

$$s_1(t) \;=\; 1/2 \; \cos 2\pi(f_0 - f_1 - f_2)\,t + 1/2 \; \cos 2\pi(f_0 - f_1 + f_2)\,t \qquad (2.15)$$

$$s_2(t) \;=\; r_2(t) \; \sin(2\pi\,f_2)\,t \qquad (2.16)$$

$$s_2(t) \;=\; 1/2 \; \cos 2\pi(f_1 - f_0 - f2)\,t + 1/2 \; \cos 2\pi(f_1 - f_0 + f_2)\,t \qquad (2.17)$$

The output of the adder after manipulation and cancellation is given by:

$$y(t) = s_1(t) - s_2(t) = \cos 2\pi(f_0 + f2 - f_1)\,t = \cos 2\pi\,(f \pm \Delta f)\,t \qquad (2.18)$$

Where Δf is the difference between the two quadrature frequency oscillators.

The Weaver technique's major attraction compared with the Hilbert transform method is that no wideband 90° phase-shift network is required, since the processing is accomplished using quadrature local oscillators which can be very accurately derived. This means that the likelihood of poor sideband suppression due to quadrature imbalance is greatly reduced.

CHAPTER 3

PRIOR RESEARCH AND WORK

3.1   Introduction

The chapter presents various methods, proposed in these journals, for improving the acoustic feedback stability of public address systems, all of which suggest the frequency shifting scheme. The optimum frequency shifting of audio signals was found by all to be about $\Delta f = 5$ Hz.

3.2   SSB Suppressor Method

This method was described by Schroeder[4][5], and Prestigiacomo and Mclean [6]. Their idea of frequency shifting was effected by a Single Sideband-modulation technique shown in figure 3.1, [4].

In the above system, the microphone signal after amplification modulates a 20 Khz carrier generated by a crystal-controlled oscillator. The signal passes through a SSB-filter between 20 and 30 Khz which removes the carrier and the lower-sideband(LSB). The upper-sideband (USB) is then modulated by a carrier of 19.995 Khz and lowpass filtered. The resulting signal has had all frequency components shifted by 5Hz with respect to the input signal.

In the theoretical study done by Schroeder[4], it was shown that frequency shifting permits one to increase the gain by an amount equal to the difference between the maximum gain and the mean gain of the steady-state frequency response of the open loop reponse of the acoustic feedback of the P.A.-room system with

negligible direct sound transmission. Figure 3.2 displays such a response [4,5]. The

peak levels of the room response curve are less than 0 dB.
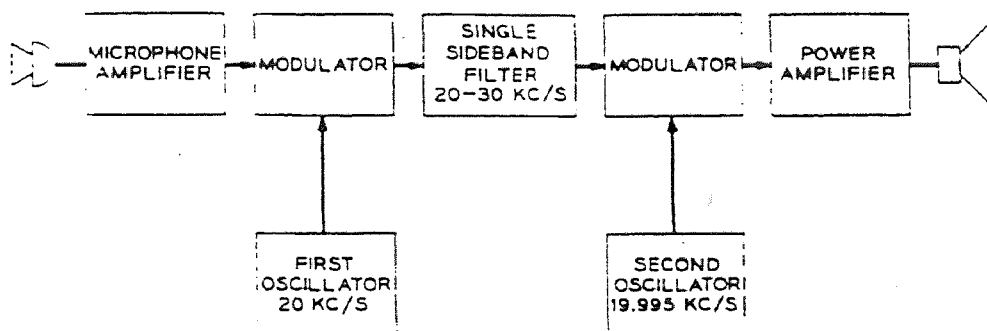
Figure 3.1: Single-sideband method of frequency shifting.
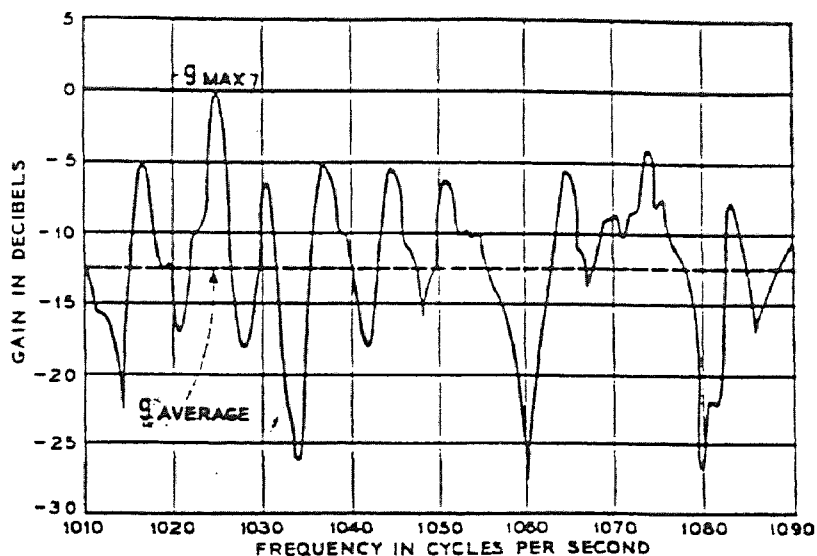
Figure 3.2: Frequency room response of a large room-P.A. system without frequency shift. Avg. frequency spacing of 5Hz, avg. reverberant time 1 sec.

From a stability point of view, the theoretical permissible increase in gain when using frequency shifting can be predicted by the following equation given by Shroeder[4]:

$$g_{max} - g_{avg} = \{10 \, \log(\ln(T \, W)) + 2.5\} \, dB \qquad (3.1)$$

where g_{max} is the maximum gain of the signal response of a public address system without frequency shiftiing operating in a room(when g_{max} exceeds 0 dB system becomes unstable), g_{avg} is the average gain level of frequnecy response employing frquency shifting of 5 Hz, T is the average reverberation time of the room, and W is the bandwidth of the P.A. system. It is to be noted that the equation above was derived by considering only the magnitude of the open-loop gain, neglecting its phase(see Appendix for detailed derivation of equation 3.1 and the definition of reverberation time). As a result, this gain increase is slightly higher than the gain for the stability of a public address systems without frequency shifting. There is a discrepency of about 2 dB. This formula is a theoretical prediction of the additional stable gain defined as the difference in gains for stable operation with and without frequency shifting. As shown in figure 3.2, the maximum permissible gain should be less than the peaks labeled (g_{max}) of the particular room frequency response, which should be less than 0 dB in order to maintain stability of the public address system. If the room response could be flattened and made to have an average gain, g_{avg} (indicated by dashed line), for all frequencies, the gain of the public address system could be increased by the quantity given by equation (2.3). By shifting all the frequency components of the microphone signal by an amount equal to the average distance between the response peaks and adjacent valleys, typically about 5Hz but may vary from one response to another, the effective room response may be flattened. From a subjective point of view, the performance has to be evaluated differently. It was approximated, from a practical point of view that the useable additional gain is about 4dB less than the theoretical quantity given by the differ-

ence between the two levels given by equation (3.1). The useable additional gain due to frequency shifting can be predicted by the following equation [5]:

$$\text{useable additional gain} = \{10 \, \log(\log(T \, W/22)) + \Delta\} \, dB \qquad (3.2)$$

where $1 \, dB \leq \Delta \leq 4 \, dB$;

For a typical choice WT=5000, so that the useable gain is between 5 and 8 dB. Measurement of the additional stable gain as a function of the magnitude and sign of the frequency shift have been made in different room sizes using the system setup of figure 3.1. The results of these experiments are plotted in figure 3.3, 3.4, 3.5, see [5]. A detailed qualitative explanation of these figures is presented in the last section of this chapter.
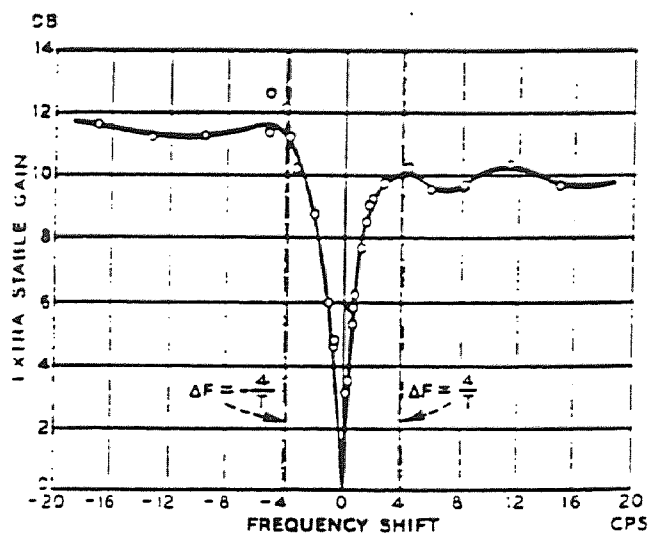
Figure 3.3: Additional stable gain in a large auditorium (volume 2500 m^3, average reverberation time T= 1 sec).
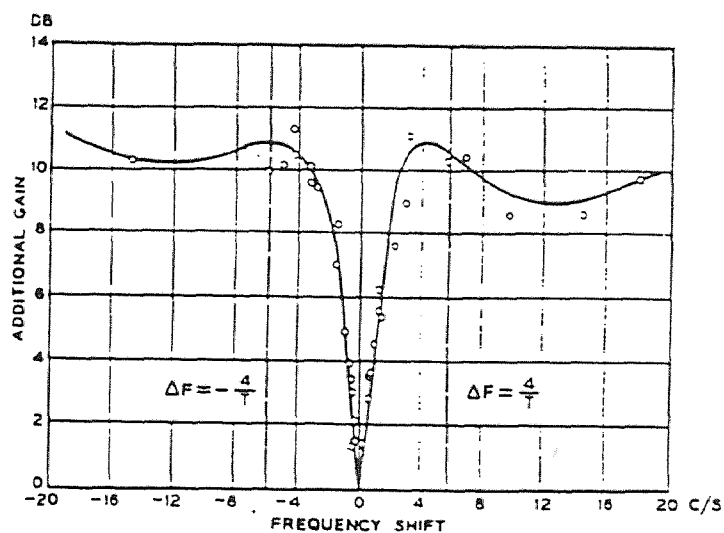
Figure 3.4: Additional stable gain in a medium size room (volume 100 m^3, average reverberation time T=0.7 sec).
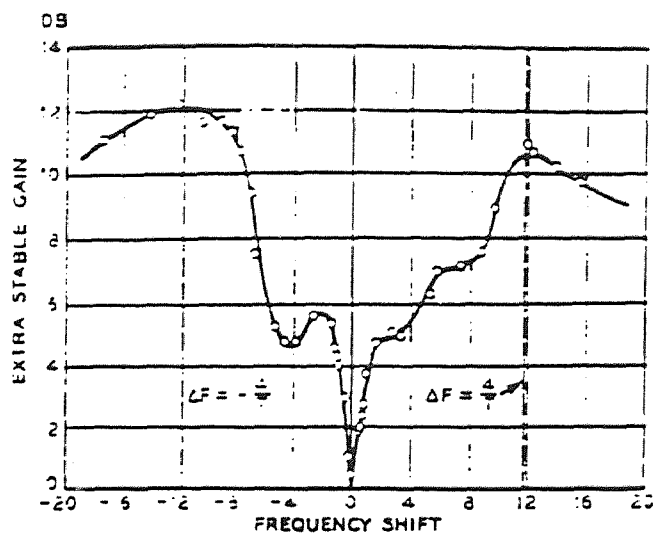
Figure 3.5: Additional stable gain in soundproof booth (volume 14 m^3, average reverberation time T=0.34 sec).

3.3 Phase-Shifting Method

In a paper by Bode and Moog[7], two frequency shifting systems using phase-shifting methods were proposed. Only the simplest one is discussed here. This method is similar to the Hilbert transform technique discussed in section(2.3).

The system consists of an electronic instrument which is capable of shifting all the frequencies of the audio spectrum by the same amount. The frequency shifter utilizes a phase-shifting method described by the following block diagram figure 3.6,see [7].
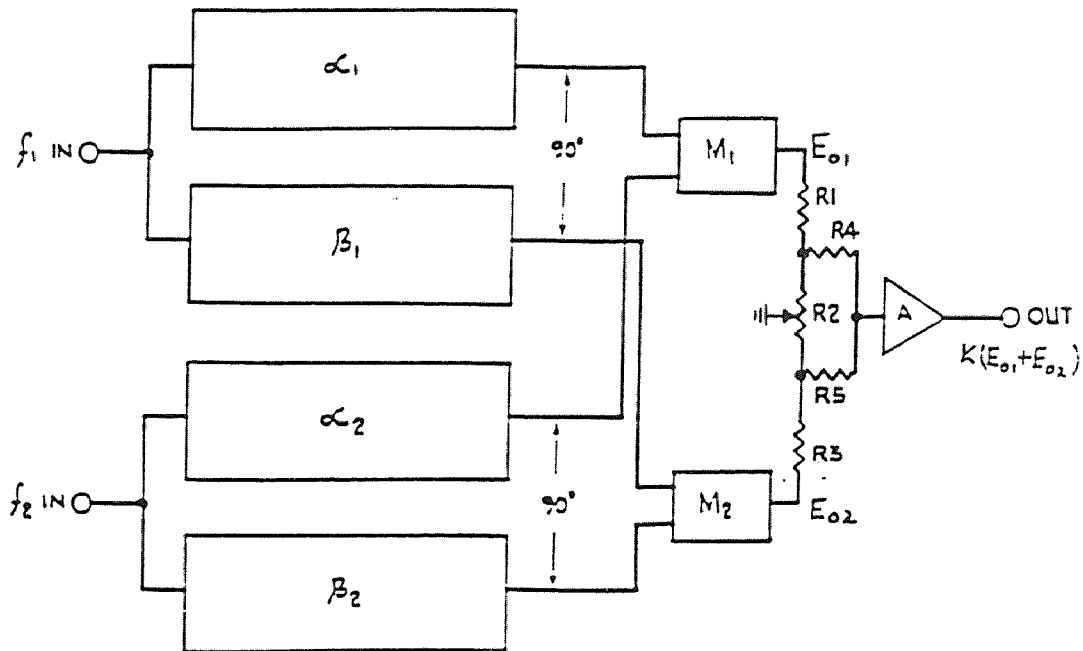
Figure 3.6: Block diagram of phase-shifting type frequency shifter.

Here the signal is fed into the input f_1 IN of the two phase shifting circuits α_1 and β_1, which are designed so that at their outputs all audio frequencies are $90°$ apart in phase. These frequencies are then fed to two multipliers M1 and M2.

In the same way the carrier frequency, which is fed to f_2 IN of the phase shifters α_2 and β_2, is processed for obtaining a relative phase relationship of $90°$ of carrier frequencies. These outputs are fed into the second inputs of the multipliers M1 and M2. The output signals of the multipliers M1 and M2 are added and amplified resulting in shifted signal spectra.

The following equations recapitulate the basic equations for the functionality of the system above [7].

If the signal outputs of the phase filters α_1 and α_2 are considered to be:

$$E_{\alpha_1} = \cos(\omega_1 t) \tag{3.3}$$

$$E_{\alpha_2} = \cos(\omega_2 t) \tag{3.4}$$

then after multiplication in the four-quadrant multiplier M1 the output will be:

$$E_{o_1} = \cos(\omega_1 t)\cos(\omega_2 t) \tag{3.5}$$

Due to the $90°$ phase relationship between the outputs of the phase filters β_1 and β_2 versus α_1 and α_2 the output will be:

$$E_{\beta_1} = \sin(\omega_1 t) \tag{3.6}$$

$$E_{\beta_2} = \sin(\omega_2 t) \tag{3.7}$$

After multiplication in the four-quadrant multiplier M2 the output will be:

$$E_{o_2} = \sin(\omega_1 t)\sin(\omega_2 t) \tag{3.8}$$

After summing the output of M1 and M2, and simplifying the expression the final result is given by:

$$E_{o_1} + E_{o_2} = \cos(\omega_2 - \omega_1)t \tag{3.9}$$

which is the frequency shift of ω_2 by negative amount of ω_1

3.4 Base-Band Shifting Method

The base-band shifting method is proposed by Boukhard[8][9]. A block diagram of the system is shown in figure 3.7.
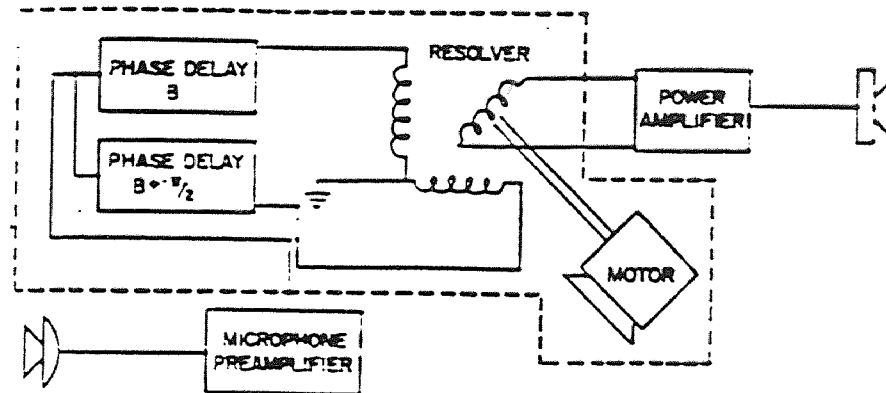
Figure 3.7: Block diagram of base-band frequency shifting.

Here the signal from the microphone passes into a phase splitting network which provides two signals displaced in phase by 90^0 relative to each other. These signals feed the two quadrature primary winding of a resolver. A resolver is a variable coupling transformer in which the secondary windings is rotatable relative to the primary. The output of the resolver is a constant amplitude signal whose phase varies according to the position of the rotor relative to the stator windings. Thus the phase of the output signal can be varied through 360^0 with each shaft rotation, and if the resolver is continuously rotated, a frequency shift of 1 Hz per revolution/second occurs. With a constant speed motor driving the resolver at a

value of 300 rpm, a 5 Hz frequency shift is obtained. A schematic diagram of the resolver is shown in figure 3.8, [8].
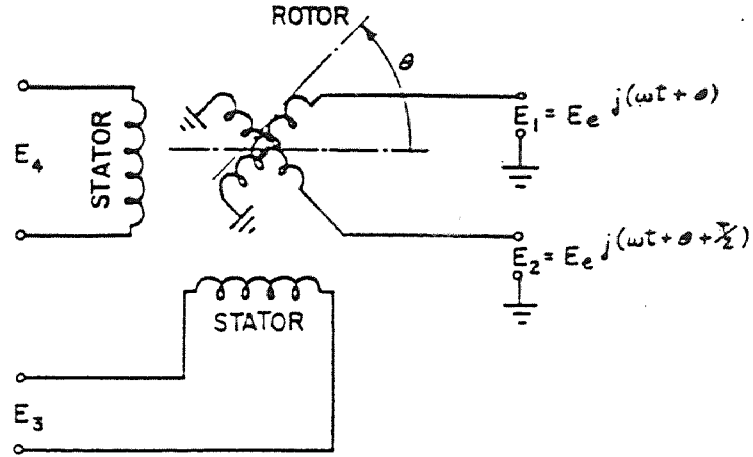
Figure 3.8: Schematic representation of an inductive resolver

The signal at the output of the resolver in the schematic above to be applied to the power amplifier can be written as:

$$E_1 = E_4 \sin\theta + E_3 \cos\theta \qquad (3.10)$$

$$E_2 = E_4 \cos\theta - E_3 \sin\theta \qquad (3.11)$$

where:

$$E_3 = E e^{(j\omega t)} \qquad (3.12)$$

$$E_4 = E e^{j(\omega + \frac{\pi}{2})t} \qquad (3.13)$$

After substitution

$$E_1 = E e^{j(\omega + \theta)t} \qquad (3.14)$$

$$E_2 \quad = E e^{j(\omega + \theta + \frac{\pi}{2})t} \tag{3.15}$$

Here θ is the instantenous angle of the resolver. The output winding of the resolver however is rotated continously by the resolver drive, and the rate of phase change in the output signal from the resolver is $d\theta/dt$. Thus a rotation of the phase angle θ through 2π radians Δf times per second produces a shift of Δf;i.e, [8]:

$$E_1 = E e^{j(\omega + 2\pi \Delta f)t} \tag{3.16}$$

The frequency shift of the order of 4 Hz made it possible to obtain a subtantial increase in the gain without engendering oscillation in the public address system. The following figure 3.9, [9], gives the additional gain for an advancing or retreating frequency shift.
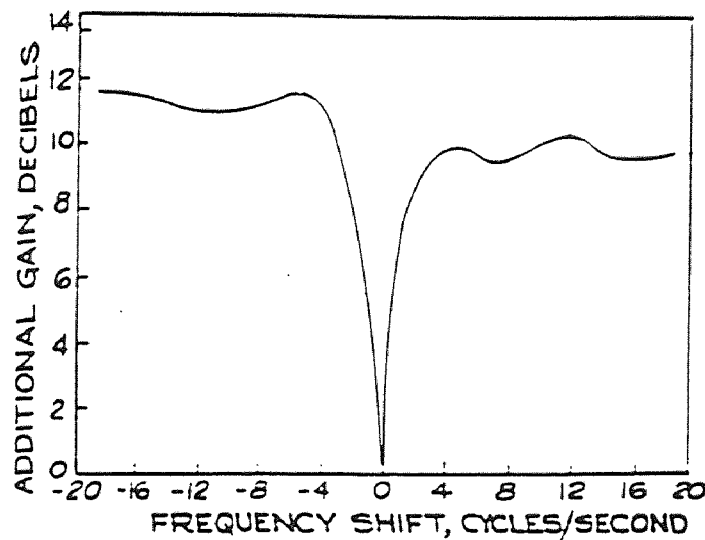
Figure 3.9: Additional stable gain using base-band shifting method in a typical room.

It is to be noted that the figure above is for a specific room environment.

For different room conditions such as different room size or temperature, data may vary accordingly as shown in previous figures.

3.5 Qualitative Discussion of Results

It should be understood that all of the figures(3.3-3.5, and 3.9) representing the additional gain versus frequency shifts are applicable, only to a given reverberant room or other specific environment such as temperature and room sizes, thereby establishing different optimum condition for the amount of frequency shift that will give a maximum gain increase of the system.

It was found by Schroeder[4][5], through experimental measurement that the optimum shifts of the specific room is about 4/T Hz, as pointed out in the figures by dashed lines. This measured value, (4/T) where T is the average reverberant time, is equal to the average spacing between peaks(maxima) and adjacent valley(minima) of the room frequency response. This way, the excess energies generated at the gain peaks are quickly absorbed in the valleys.

Also shown in the figures, as expected, no additional gain can be realized without frequency shifting. Furthermore, there is no significant consistent difference between positive or negative shifts. The somewhat larger gain increase for negative shifts, seen in some of the figures, could be due to the presence of deep valley in the frequency response just below a major peak. Moreover, larger shifts do not give any significant additional improvement and, in some cases are even less effective. Finally, though all the figures show that an average stable gain of about 10 dB margin should be realizable, subjective evaluation by the authors reveals that only 6 dB gain is realizable by the frequency shifting. This value of 6 dB is the practical realization that offers not only stability but also reduction of the whistling noise,

while 10 dB is only for stability in the public address-room system,i.e the open loop gain is less than 0 dB. It is known that only subjective test can tell as always the speech quality of the system.

CHAPTER 4

OUR EXPERIMENTAL SETUP

4.1   Introduction

This chapter describes an experimental setup for evaluating the frequency shifting method using filter techniques to reduce the whistling noise and improve the acoustic feedback instability of a public-address system operating in a closed room environment.

As discussed earlier, instability occurs when voice signals reflected off the room walls and other obstacles, reenter the microphone. When these acoustic feedback signals are in phase and of amplitude larger than the original signal at some frequencies, self-oscillation occurs causing an irritating whistling noise.

This whistling or howling noise can be reduced by avoiding the oscillation criteria of equation (2.3). One way to elude these criteria is by using directional microphones and speakers with proper placement. Proper results cannot be achieved if microphones are moved quite often, and it is impractical because acoustic signal distribution varies from one frequency to another and from one microphone location to another.

The proposed method makes use of the frequency translation techniques discussed in section (2.3). The idea is to shift the speech spectrum by a small frequency offset so that the amplified output spectrum is different from the spectrum at the input. Thus, gain buildup at the critical frequency is reduced and self-oscillation is avoided.

This chapter presents the experimental investigation of this speech translator

and the amount of shift necessary to maintain good speech quality and to avoid oscillation.

4.2 Frequency Shifting Discussion

The reflected sound from the speaker to the microphone of a public address system shown in figure 4.1 (switches in position A), contains a large number of of reflection and of varying path length, amplitude, and phase. Part of the energy generated at the output is returned to the input. The fed back signals add to the original input signal causing a gain buildup during a short time interval. When the amplitude buildup is at a frequency where the phase shift around the loop is a multiple of $360°$ and the loop gain is greater than unity, oscillation occurs. Because this feedback is positive, the amplitude would continue to increase until the open loop gain is saturated or reduced to just match the oscillation gain criteria; otherwise the oscillation amplitude would increase indefinitely.

For oscillation to build up again, a short period of time (oscilation buildup time) is needed to allow for the open loop gain to build up to an appropriate amplitude. However, if the amplitude or phase of the fed back signal is interrupted in a period of time shorter than the oscillation buildup time required, then oscillation would be withheld until the next oscillation buildup time. Thus, shifting the frequency or phase of the signal by a small amount would interfere with the criteria for oscillation.
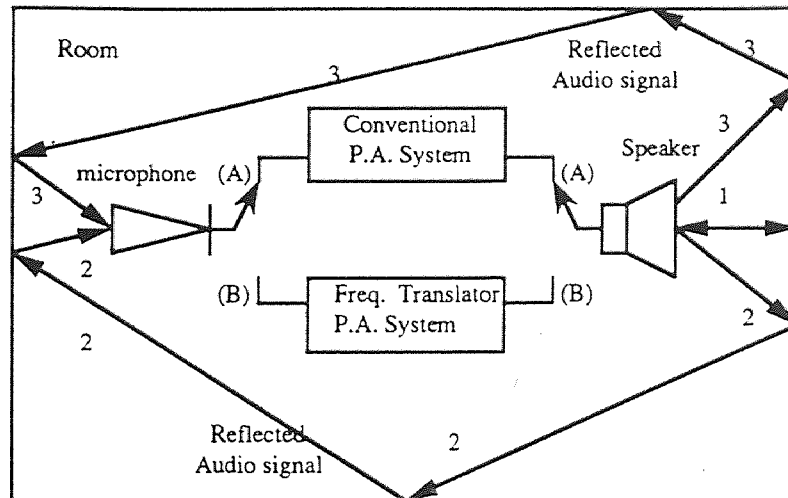
Figure 4.1: A conventional public address system and frequency shifter public address system.

Reference is made to figure 4.2, showing an arbitrary relationship between the phase and amplitude used for this discussion. If a frequency- shifting or phase-shifting mechanism is added to the public-address system, as shown in figure 4.1 (switches in position B), that causes a change in phase of the open loop system, say a 20^0 lag or lead, the oscillation frequency will change by Δf Hz, either increased or decreased until the total phase shift is a multiple of 360^0 again. Similarly, if an effective frequency-shift in an amount of Δf is added or subtructed to the oscillation frequency, a phase lag or lead of an amount of $\Delta \phi$ is attained, and the relative open loop gain is decreased. Therefore, the oscillation criteria of equation(2.3) are avoided. Figure 4.2 shows the effect on the phase and amplitude of the open-loop gain if the oscillation frequency is shifted by a small offset of Δf Hz.
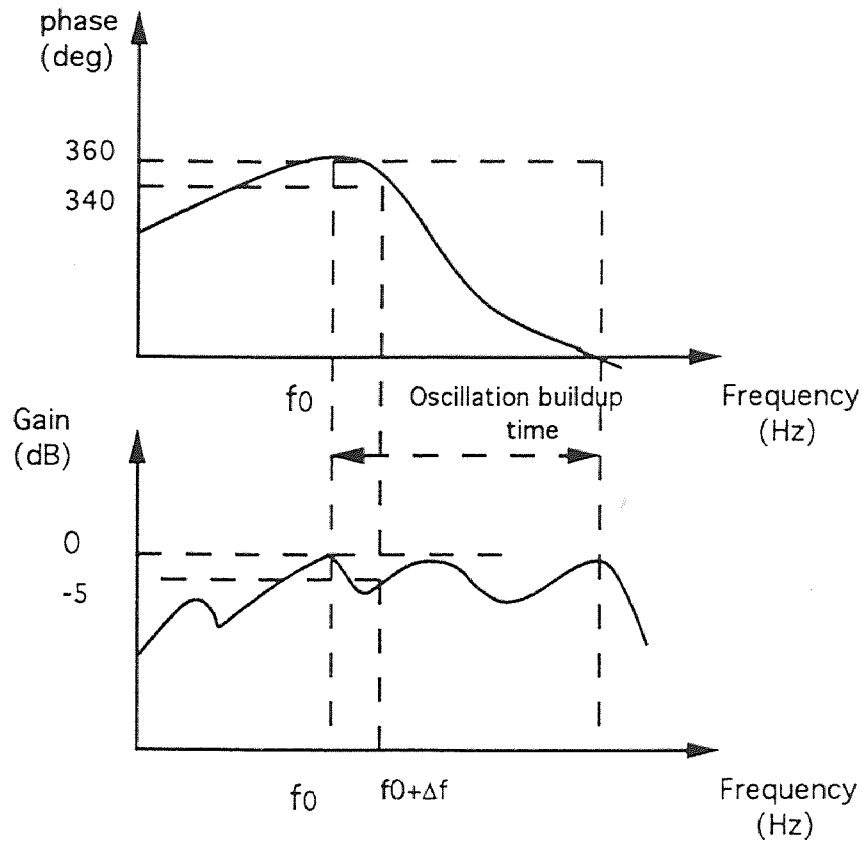
Figure 4.2: Effect of frequency shift on the phase and amplitude of an open-loop gain.

If all of the signal frequencies are shifted continuously by a small offset, producing an effective phase change, then the oscillation build up is disrupted. Thus, effectively interrupting the phase coherence of the acoustical fed back signal from speaker to microphone via room walls and other obstacles, makes it possible to increase the overall gain of the system and allows a considerable increase in the signal sound level in the enclosed room.

The linear frequency shifter inserted between the preamplifier and the power amplifier of the public address system, would shift the input signal by a small offset Δf Hz, each time it passes through it. The first time the audio signal enters the

microphone, it is shifted in frequency by Δf Hz before it is emitted by the speaker into the room. The sound is deflected and fed back into the microphone, which is then shifted again in frequency by Δf Hz. As it is passed the second time around and through the signal channel of the public address system it is shifted by a total of $2\Delta f$ Hz. As the signal is fed back several times from the speaker to the microphone, it is shifted in frequency by a total of $n\Delta f$ Hz, where n is the number of times the signal goes through the system. Therefore, the loop gain of the public address frequency-shifter system is always kept at less than unity, since the fed back signal maximum amplitude was reduced below the original 0 dB and is shifted to a lower amplitude. If Δf is chosen appropriately, then the signal which is first amplified at the peak of the response is shifted to a valley of the response curve. Thus frequency shifting is reducing oscillation noise and increasing the permissible overall gain.

4.3   Objective

The speech spectrum translator (frequency shifter) is to accept the frequency range of 300 Hz to 3300 Hz, and shift this spectrum to a selected band. Except for the spectral shift, the signal is not to suffer other perceptible distortion, and should sound like the original voice signal when translated back to the original band. The frequency shift is to be settable in steps of 1 Hz. The choice of the frequency range of 300 to 3300 Hz is based on telephone voice quality. The shifted output signal is to be amplified to a high level for the purpose of driving a transducer(loudspeaker). The implemented experimental device of frequency-shifting should be of simple construction.

4.4 Speech Spectrum

The spectral characteristics of speech vary over a wide-range because of the different component sounds involved and because of wide differences among different speakers. Figure 4.3, given by Bennett[10], shows the average energy-versus-frequency distribution for representatives talkers(three males and three females).
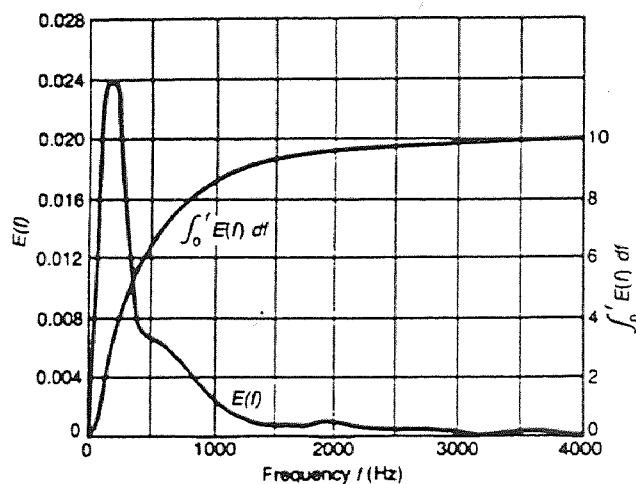
Figure 4.3: Average energy-versus-frequency distribution for representative talkers. E(f)df is energy in band of width df at frequency f.

It is seen from this figure that the energy is zero at frequencies near zero, rises to a peak at about 300 Hz, falls rapidly until about 400 Hz, and then decays more slowly to several thousand Hertz. The importance of the various portions of the spectrum to speech understanding is not determined by this relative power distribution curve. In fact the low-frequency components can be eliminated with

little loss of speech intelligibility which is a practical criteria of telephone quality of received speech.

Intelligibility is determined experimentaly by testing the subject's performance using sentences, words, and syllables of various speakers. A typical test result is shown in figure 4.4, given by Bennett[10]. In this test a group of listeners record the sounds that they think they heard. The percentage of correct observations is called the "Sound Articulation",[10].
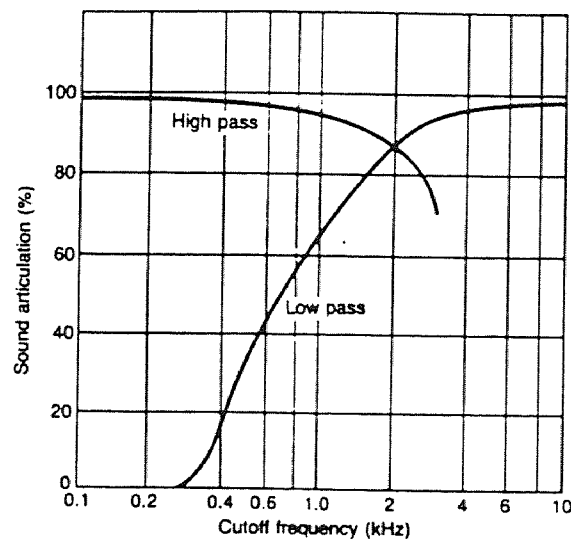
Figure 4.4: Importance of frequency range in articulation.

The curve marked high-pass shows the result of removing the low- frequency components of speech. It is seen that almost no effect is noticeable for removal of frequency components up to about 300 Hz. For the low-pass curve, the high frequency components are attenuated and it is observed that components above 3500 Hz contribute only slightly to improved performance. This type of test and other related tests have let to the adoption of a voice baseband for communication

purposes lying in the range 300 to 3500 Hz. In this frequency band, there is little attenuation, but outside those limits its strength falls off rapidly. Indeed, it is found that telephone circuits which respond well to this range of frequencies give quite satisfactory commercial telephone service.

For sound perception, similar tests were conducted. It is found that the human auditory system is in the frequency range of approximately 16 Hz to 16 Khz. The upper limit falls off with increasing age; among the young it occasionally reaches 20 KHz; in old age it may be as low as 10 KHz. At the low end of the frequency range, the perceived sound becomes a pulse train; at the high end, it fades off into silence.

4.5   Laboratory Experimental System Setup

The experiment was carried out in a closed room. The room has a rectangular shape with parallel walls of hard structure. It has a volume of about 2185 ft^3 and dimensions of about 10ft x 23ft x 9.5ft. It also contains desks, file cabinets, workbenchs, and other furniture that were fixed to a specific location throughout the experiment to account for any reflection that might occur.

A block diagram of the experimental setup is shown in figure 4.5. The input voice signal from a microphone or tape recorder is passed through a filter to cut off frequency components above 4000 Hz. The bandlimited voice-signal is amplified to a suitable level, then applied to an analog multiplier,(M-1). The second input port of the analog multiplier receives a 68.750 KHz carrier. After multiplication, a double-sideband suppressed carrier signal is generated. The single sideband filter cuts off the lower sideband and further rejects the carrier. The upper sideband signal in the range of 69.0 to 72.5 KHz is applied to an input port of (M-2). The

second input port of (M-2) receives another carrier from the second local oscillator at 68.750 KHz tunable in 1 Hz steps upward or downward. The output of (M-2) contains the desired translated signal in the range of 250 to 3500 Hz $\pm\Delta$, as well as undesired component formed as a sum in the range of 137.750 to 140.750 Khz. A low-pass filter is used to suppress these components above 5 KHz. The result is a speech signal translated in frequency by the difference between the first and second oscillator carrier frequencies. The translated speech is applied to an amplifier (A-2) which boosts the signal to a sufficient level to drive the power amplifier (A-3). A potentiometer between (A-2) and (A-3) controls the level of the power. A meter is provided at the output of (A-3) indicating the level of the signal voltage applied to the transducer(loudspeaker) . All filters, mixers, oscillators and other instruments used in this experiment were selected after a preliminary study and testing.

System outline:

Reference is made to the block diagram of figure (4.5). As is well known, in implementing a system there are numerous approaches possible. In this experimental system design cost effectiveness and commercially available components were endeavered. All of the system components used were tested and calibrated for best performance and reliability.
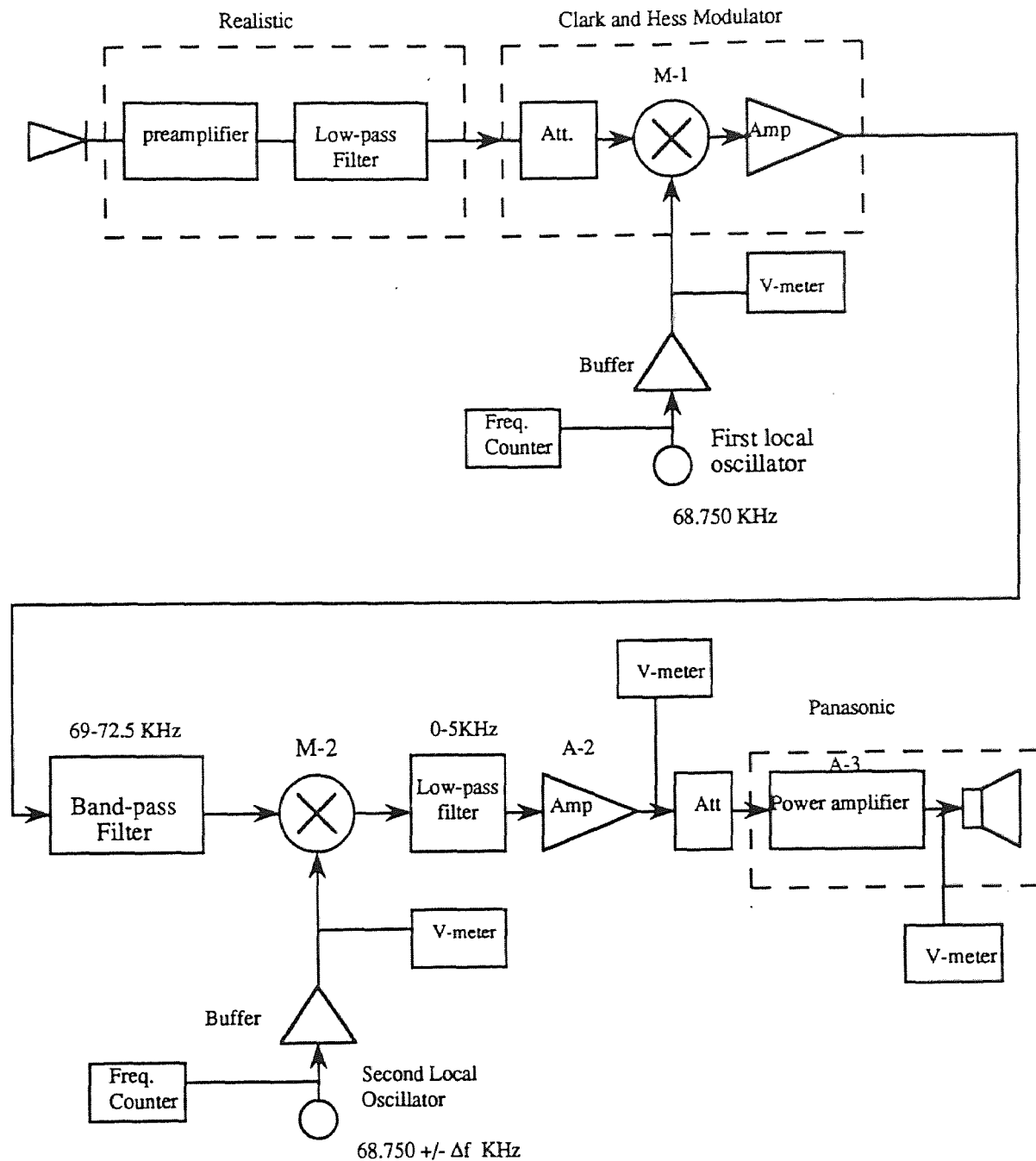
Figure 4.5: Block diagram of our experimental P.A. SSB translator system

The mixers used in this experiment are Clarke-Hess model-200 modulator/multiplier (M-1) and HP model 1053A (M-2). These mixers offer an outstanding combination of high performance, versatility, and were available in the laboratory. Some of the advantages are incorporated in low insertion loss, excellent balance of port, low intermodulation products and a high degree of carrier suppression at about 45 dB which reduces the BPF requirement. Moreover, the Clark-Hess mixer contains an amplifier at the output port and an attenuator at the input stage. The amplifier amplifies both the difference and the sum of the product to proper level. The attenuator controls the input signal level such as not to overdrive the mixer. There are two LEDs indicating any overdrive.

General Radio Generators model-1001-A are utilized for the local oscillators to generate the two carrier frequencies. They offer a high degree of frequency stability over a period of time and controllability in terms of temperature, noise, and changing environment. The tuning range is in the step of 1 Hertz.

The Allen Avionics LPF model-F2000 of cutoff frequency of 5KHZ is used. The bandpass filter (BPF) utilized is the Marconi Instrument model-TM 7730/2 which is a tunable L-C realization of 69.0-72.5 Khz bandpass region. This filter offers conservation of spectrum shape, and reduction of unwanted signals. In addition, it gives better rejection of carrier and better frequency selectivity.

Various amplifiers were used to amplify the signal to suitable levels. Some of the instruments have built in amplifiers offering good performance. At the input stage the Realistic model MPS-5 from Radio Shack is used. It contains a microphone, a pre-amplifier,and speaker. At the output stage, the Panasonic Tape Player model-RQ 7065 is used. It contains an adjustable power amplifier that amplifies the translated voice signal to a level to drive the loudspeaker. Other amplifiers used are the Hewlett-Packard model-400D. These amplifiers are placed accordingly during calibration for a better gain distribution throughout the system so that there is no

overdrive or clipping of the signal level at one end or the other.

The following is the list of other test instrument used either for performance check, and/or calibration of the experimental system.

• Philips model-PM6667 frequency counters of 120 MHz were used to monitor the frequency carrier readout of local oscillators.

• Tektronix oscilloscope model-2215A, a 60 MHz oscilloscope. Used to measure the displayed signal level and to analyse the signal waves at various test points.

• Simpson voltmeter model-200, used to control the voltage level at test point.

• H.P. Wide range oscillator model-200 CD.

• H.P. Attenuator model-350D.

• H.P. Spectrum Analyser model-8568B.

Mathematical analysis:

Only single tone input is considered for simplicity. Reference is made to figure 4.5. Let the input signal x(t) be given by:

$$x(t) = \cos(2\pi f t) \tag{4.1}$$

where $0.3\,KHz \leq f \leq 3.3\,KHz$;

After first multiplication in M-1 with 68.750 KHz carrier, the M-1 mixer output r(t) becomes:

$$r(t) = x(t) \cos(2\pi\,68.750)t \tag{4.2}$$

$$r(t) = 1/2 \cos 2\pi(f - 68.750)t + 1/2 \cos 2\pi(f + 68.750)t \tag{4.3}$$

After bandpass-filtering, the LSB and carrier are suppressed and the USB becomes:

$$s(t) = BPF(r(t)) = 1/2 \cos 2\pi(f + 68.750)t \tag{4.4}$$

When s(t) is multiplied by the second mixer M-2 carrier the conversion output w(t) becomes:

$$w(t) = s(t) \cos 2\pi (68.750 \pm \Delta f)t \tag{4.5}$$

$$w(t) = 1/4 \; \cos 2\pi(f \mp \Delta f)t + 1/4 \; cos 2\pi(f + 137.750 \pm \Delta f)t \quad (4.6)$$

where Δf=1,2,3... Hz.

After lowpass filtering and amplification the final signal output y(t) is given by:

$$y(t) = LPF(w(t)) = \cos 2\pi \, (f \pm \Delta f)t \quad (4.7)$$

which is the original input signal shifted in frequency either upward or downward in steps of Δf Hz.

4.6 System Calibration

A systematic approach was utilized to calibrate and troubleshoot the overall performance of the experimental set-up of figure 4.5. Each component from oscillators to mixers, as listed in the previous outline was tested seperately and compared against similar products for better performance once integrated in the overall system.

The most common systematic method is signal tracing after signal injection. The overall system was monitored at all times by tracing signals at each one of all the test points (TP) as shown in figure 4.6.

A frequency generator, H.P. model-200CD was utilised as an input source to supply appropriate single tone signals with different frequencies and amplitude levels. The reason for using single tone signals at first, was because it is easier to trace and to identify it at various stages.
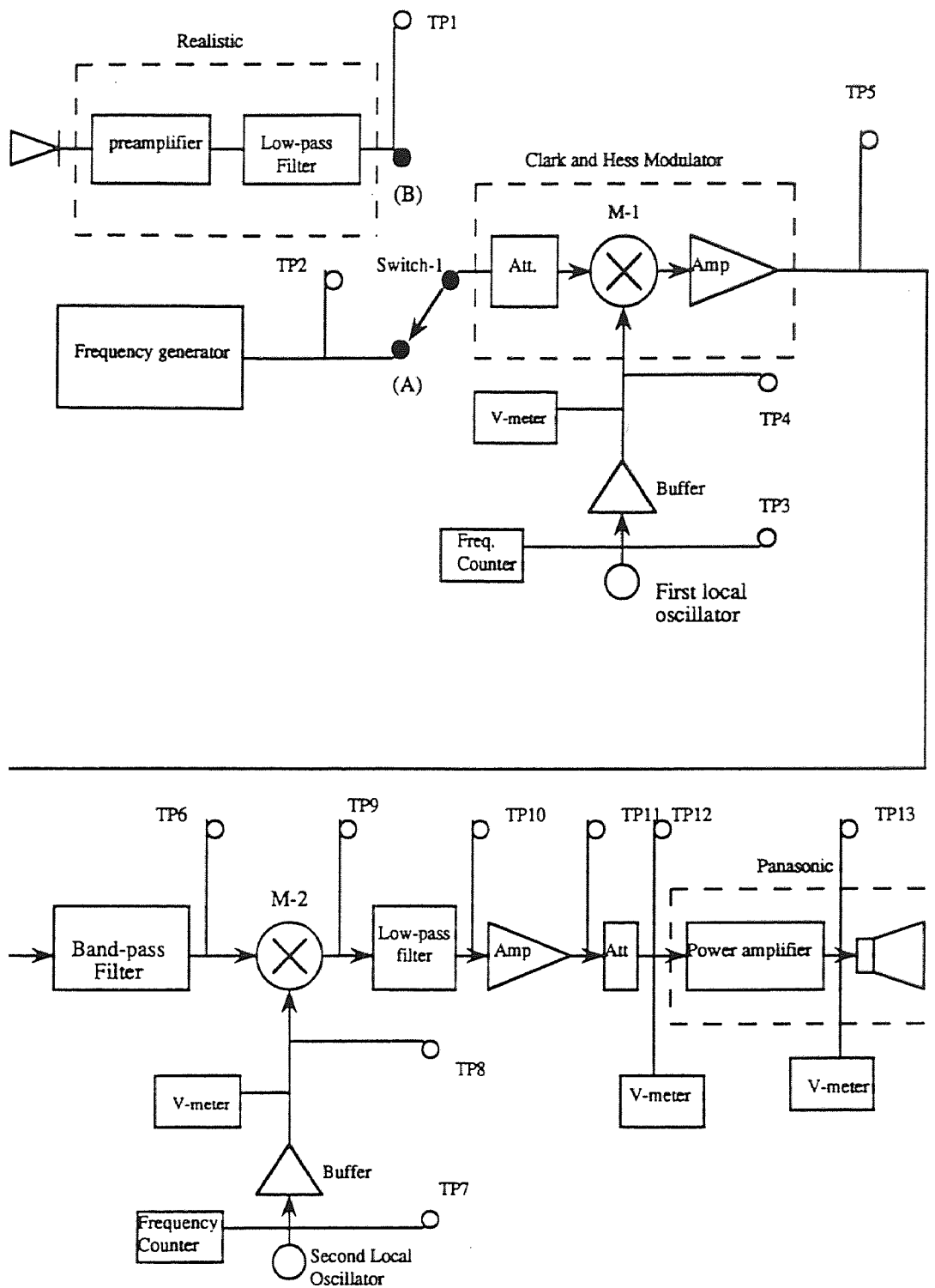
Figure 4.6: Experimental system showing test points (TP) used for calibration.

Furthermore, one can adjust the signal levels properly to maintain good gain distributions throughout the system and not to overload any mixers, or overdrive any amplifiers. A dual-trace oscilloscope was used to display the waveform shape present at the input and output of each stage simultaneousely. A probe is used for low level signals,when needed to trace any distortions and phase relationship on the dual trace mode. A spectrum analyser is used, especially to display the modulated signals DSBSC and the SSBSC spectrum in order to identify any distortion.

Procedure:

Reference is made to figure 4.6. First the system was isolated and kept very steady at the fixed location. Then a source signal, a pure sine wave set to an appropriate ampliude and frequency not exeeding the speech band, was fed into the input stage of the first mixer (M-1)(switch-1 in position A). The probe lead was placed at each single test point marked as (TP), to check for presence of test signal. The reference signal at TP1, pure sine wave, was displayed at all time on the scope X-trace in order to compare it to other signals at different test points(TP).

Attention was paid to the oscillators output stages, especially the second stage oscillator, since any changes in frequency will result in signal phase and voltage level variations. This is monitored by displaying test points TP2 and TP6 at all time. Whenever a small freqency shift Δf is introduced to the second local oscillator carrier, voltage level adjustment was made to maintain the 2V peak-to-peak level of the carrier signal of both oscillators. The 2V peak-to-peak was used throughout the entire experiment. Some difference between the two local oscillators might still result, because of frequency drift over a long period of time, but not major to cause any gross distortions. Phase distortion would not have major effect on speech signal intelligibility, since the human ear is insensitive to any phase distortions. Still, the effect of phase on the speech signal can be reduced when the same local oscillator is feeding both mixers. To allow small frequency shifting, and reduce frequency

drifts one should utilize a frequency synthesizer in order to shift the speech signal by an amount as small as 0.001 Hz. Frequency synthesizers will be investigated in the next section.

For the mixers stages, signal levels were minimized such that no overloading occurs at their input ports. The modulated signal at TP4 was diplayed on the scope in dual trace X-Y mode to verify the Double Sideband Suppressed Carrier (DSBSC) wave. This was verified in the frequency domain when displayed on the Spectrum Analyser. Similarly, the Single Sideband Suppressed Carrier (SSBSC) spectrum was obtained by cheking the signal at TP5. The results were as expected.

For amplifier stages, appropriate test points were checked to verify the signal voltage level amplification. A rough check of each stage gain was made by reading the signal level at the input and output of the amplifier stages to make sure that the output signal has been amplified properly.

The overall performance of the experimental frequency shifting system was monitored constantly by using the signal injection and signal tracing methods to check each stage sequentially. After the speech signal was inserted, switch-1 in position(B), and all of the test points were checked again to verify the overall performance; adjustments were made where needed. The speech spectrum was clearly displayed on the Spectrum Analyser, and the results were as expected.

4.7   Experimental Data

After calibrating the experimental system of figure 4.5 and making necessary adjustments(i.e. adjusting amplitude level of the input signal to a level just before whistling), data were taken. In order to minimize errors due to human factors, data were collected several times and verified against each other.

During the experiments every component was maintained at a fixed location and at the same distance throught out the experiment. The room atmosphere was kept at a normal temperature, because theoretically any major increase or decrease in the temperature due to heating or air conditioning, may affect the speed of acoustic sound which affects the reverbration path in return. Hence, this temperature gradient may introduce some frequency shift, which may or may not affect the overall acoustic feedback of the public address system operating in the room; it depends on how drastically is the change in the room's temperature. Nonetheless, the room atmosphere was kept at normal temperature at all time.

The room's door was kept closed during collection of data so that no resonance may occur and cause any variations of the room's mode (node and antinode). An open door may change the room modes just as the case of the acoustics in closed or open pipe.

All other furniture in the room were fixed to the same coordinates. The microphone was placed properly far enough from the loudspeaker, in a position of low radiated sound, in order to eliminate directional feedback noise. The distance from the microphone to the loudspeaker was maintained at all times to a 58 inches radius.

All of this monitoring to maintain the room atmosphere stable was done so that no major disturbance would affect the acoustic feedback and to keep errors to a minimum.

Data aquisition:

Reference is made to figure 4.5. Oscillation frequencies were determined directly by increasing the system gain until oscillation occured. Afterward, the gain was attenuated to keep the gain just below the 0 dB level, the stability level. The speech signal was then fed into the microphone of the public address frequency shifter system, and the frequency shifting of the signal was attained by slowly sweeping

the second oscillator carrier in small steps of $\Delta f=1$ Hertz, and simultaneously increasing the gain until it reaches the maximum level just before whistling. This gain increase was recorded and the effect of the frequency shift on the speech intelligibility was monitored, and described. The subjective performance is the criteria used to determine the intelligibility of the shifted speech signal. This means, using the human ears to hear the voice signal and determine wether the intelligibility and quality is satisfactory or not.

At first, the voice signal is shifted by a frequency amount in steps of 5 Hz upto 100 Hz, in order to determine the maximum amount of frequency shift possible before the speech signal is deteriorated. Table 4.1 shows the subjective results of shifting the speech signal vs. the amount of the shift. The maximum amount of frequency shift allowed before the speech signal becomes distorted was determined to be 20 Hz.

Table 4.1: Subjective results of a shifted speech signal.

Δf (Hz)	Subjective Speech intelligibility.
1-10	Intelligibility and Quality are excellent.
11-20	Intelligible but little noisy.
21-50	Intelligible but very noisy.
50 $\leq$	Deteriorated, not intelligible.

An upward and downward conversion in step of 1 Hz up to 20 Hz was conducted , and its effect on speech quality was observed. The results are shown in Table 4.2-4.3. The amount of relative gain increase introduced in the public address system after shifting all of the speech signal components was also recorded. These results are plotted in figure 4.7 which displays the relative gain increase versus the amount of frequency shift.

Table 4.2: Results of speech signal upconversion.

Δf (Hz)	Gain Increase (dB)	Subjective Speech intelligibility.
0	0	No improvement.
1	2	Excellent and loud.
2	3	Excellent and loud.
3	3	Excellent and loud.
4	4	Excellent and louder.
5	4	Excellent and louder.
6	5	Excellent and loudest.
8	5	Excellent and Loudest.
10	5	Very good and Loudest.
12	4	Good and Loud.
14	4	Average.
15	3	Below Average.
18	3	Mediocre.
20	3	Mediocre to bad.

Table 4.3: Results of speech signal downconversion .

Δf (Hz)	Gain Increase (dB)	Subjective Speech intelligibility.
0	0	Quality is excellent.
-1	1	Excellent and loud.
-2	3	Excellent and loud.
-3	3	Excellent and loud.
-4	4	Excellent and louder.
-5	4	Excellent and louder.
-6	5	Excellent and loudest.
-8	5	Excellent and loudest.
-10	5	Very good and Loudest.
-12	5	Good and Loud.
-14	4	Average.
-15	4	Average.
-18	4	Mediocre.
-20	4	Mediocre to bad.
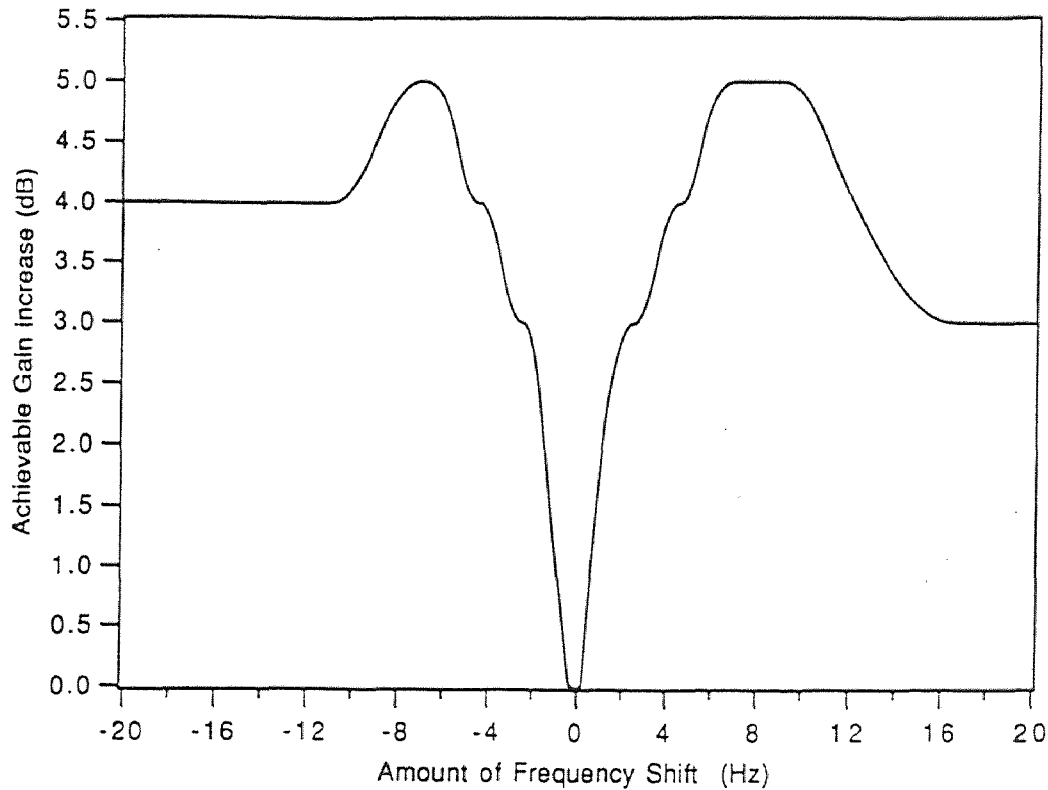
Figure 4.7: Subjective relative gain increase versus frequency shifting of speech signal in a room of 2185 ft^3 volume.

The measurement of this sort are extremely laborious, but all of this data was verified several times and erratic data, due to human error, when compared to each other was rejected in order to minimize any error.

4.8 Discussion of Results

When considering these results, certain aspects of the experiments should be remembered, such as the room size and atmosphere conditions. Different room acoustical characteristics will produce different results.

After shifting the input voice signal to the P.A. system by a constant amount

of frequency, it was determined that a frequency shift in the range of 10 Hz does not alter neither the speech quality nor the intelligibility. Reference is made to table 4.1. A frequency shift between 12 and 20 Hz effects the quality of the voice signal slightly, but the intelligibility is unharmed. Any frequency shift above 20 Hz, but less than 50 Hz will distort dramatically the quality though the speech may stay intelligible. In general, it was observed that the quality of the speech decreases faster than the intelligibility with frequency shifts larger than 10 Hz.

In these experiments it was found that a frequency shift range from 2 to 10 Hz is preferred (refer to figure 4.7 and tables 4.2-4.3). A frequency shift of less than about 2 Hz will not be sufficient enough to reduce the whistling noise, and a frequency shift greater than 10 Hz will lead to speech quality distortion even though the intelligibility may be unharmed. Although the results show that a higher frequency shift up to 20 Hz may still give higher gain increase, the speech quality will be degraded. Furthermore, the plot of figure 4.7 display almost the same behavior in the frequency shift range from -10 Hz to 10 Hz, because positive or negative frequency shift has no 1subtantial effect, as expected. However, in different room responses one frequency shift may give higher gain increase than the other. This is mainly due to the existence of a deep valley next to a major peak. This may be observed for positive or negative frequency shift of 15 Hz, in our case.

In our room size and atmosphere conditions, the optimal frequency shift value obtained that permit a maximum subjective gain increase with best quality and intelligibility of the speech, is 6 Hz. When all of the speech signal frequency components were shifted by 6 Hz in a P.A. system, a 5 dB increase in usable signal level was achieved without any whistling or distortion of the speech signal. Also as shown in figure 4.7, there are other frequency shifts permitting a 5 dB gain increase, bu the subjective quality of the speech is not as good as the optimal value of $\Delta f =$ 6 Hz. It is to be understood that the results given in figure 4.7 are for a specific

room, and that different room acoustical architecture and environment may give different optimal values.

4.9   Frequency Synthesis Investigation

As mentioned previously, when one local oscillator is feeding both mixers, it was observed that a slight noise reduction is achieved. This is mostly due to eliminating any frequency drift distortions that might arise when using two different local oscillators. But if one local oscillator is used, no frequency shifting can be achieved unless a frequency synthesizer is incorporated, especially if small frequency generation is desired.

Frequency synthesizers generates a large number of different output frequencies, all related to a single, highly stable frequency reference source generated by a crystal controlled oscillator. Frequency syhthesizers are suited for a frequency translation of very small amount as our experiments call for. Figure 4.8 shows the incorporation of such synthesizer.
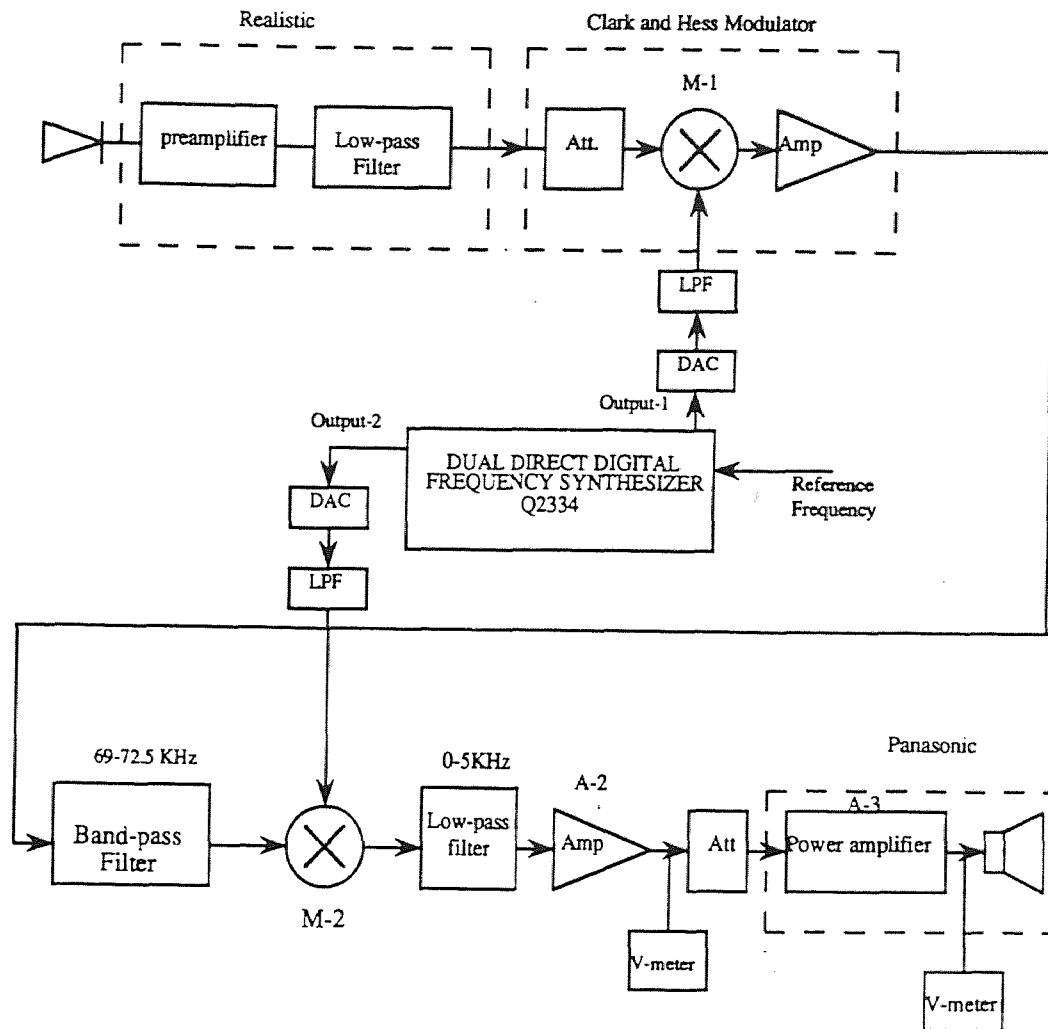
Figure 4.8: SSB frequency translation using a frequency synthesizer.

Frequency synthesis can be accomplished in three ways: direct synthesis, indirect synthesis (also known as Phase Lock Loop (PLL)), and digital frequency synthesis (DDS). A detailed litterature on the subject is described in references [11][12].

Direct synthesis refer to the generation of new frequencies using a combination of multipliers, dividers, bandpass filters, and mixers.

Indirect synthesis employs a Phase lock Loop (PLL) with a programmable divider which provides an easy method for synthesizing a large number of frequen-

divider which provides an easy method for synthesizing a large number of frequencies all of which are integral multiples of the reference frequency. The reference signal is generated by a Voltage Controlled Oscillator (VCO).

Direct Digital Frequency Synthesis (DDS), is achieved by either solving a recursive equation, or by storing the sine wave values in a lookup table. The latter is the most practically used. The DDS produces a sinusoid of a given frequency by digital integration of the frequency at a clock rate consistent with sampling theory, translating the resulting phase to a sinusoidal waveform via a table lookup, and converting the digital representation of sine wave to analog using a digital-to-analog converter (DAC), followed by a reconstruction filter.

Both PLL and DDS offer better performance than the first technique. DDS has become a viable alternate or complement to PLL synthesis for wide application. Depending on the need one method is preferable over the other. For our experiment, DDS is better suited than PLL. DDS offers excellent frequency resolution (i.e. fine frequency translation in step of 0.005 Hz) and settling time. For tradeoffs between the two techniques, one should refer to [12].

After some investigation, it was found that Qualcomm new Q2334 dual DDS offers excellent performance at a reasonable price. Detailed specs and a block diagram are shown in the appendix.

However, after finding of the Publications and realizing the time and money that it would consume, the idea of frequency synthesis was not pursued. Another suggestion, discussed in next chapter offers a better solution and is more economical. These suggestions are left for further research.

CHAPTER 5

CONCLUSION

The frequency shifting method using the two stages SSB filtering technique presented here, provides a simple implementation for effectively reducing the whistling noise due to the acoustic feedback of a public address system operating in a room. Because of this, it considerably increases the value of the achievable gain and sound pressure without introducing any distortion to the speech signal. This method has various advantages. It offers an overall stable system. The SSB frequency shifter using the filtering technique, depends only on the quality of the Bandpass filter, and the stability of the oscillators. The public address-frequency shifter system performance depends on the room sizes and conditions(atmosphere). For different room environment, one can adjust the frequency shift increment to adapt to specific room conditions.

It has been found in our experiments that the polarity of the increment of frequency shift has no substantial effect and gives about the same gain increase, but for different room responses one may generates somewhat different gain increases.

The preferred frequency shift range is from 2 to 10 Hz (see figure 4.7 and tables 4.2-4.3). A frequency shift of less than about 2 Hz will not be sufficient to reduce whisting noise, and a frequency shift greater than 12 Hz, will lead to speech quality distortion. Although a higher frequency shift up to 20 Hz has improved the gain slightly, it did deteriorate the speech quality. In our room size and atmosphere 6 Hz is the optimal value found that permitted a maximum gain increase with best speech quality and intelligibility. When all of the speech signal was shifted by 6Hz in a public address-frequency shifter system operating in a closed room, a 5 dB

increase in usable signal level was achieved without any whistling or distortion of speech quality or intelligibility. This is still a substantial improvement in the public address system quality. It is seen from figure 4.7, that there are other frequency shifts permitting greater gain increase, but the subjective quality of the speech is not as good as the optimal value of $\Delta f = 6$ Hz generates. It is to be understood that the results plotted in figure 4.7 are only for a specific room, and that differnt optimal values might be determined for different room acoustical architecture and environment.

Because each room has different acoustical characteristics, our results were slightly different than the results reported by the publications presented in chapter 3. Yet, all of us have reached the conclusion that a continous frequency shift of an amount of 5 to 6 Hz will effectively suppress the whistling noise introduced by self-oscillation that a public-address system in a closed room creates and afford a gain increase without speech distortions.

This method has many of practical applications such as in hearing-Aids, hands free speakerphones, telephone answering machine, or in conventional public address systems used in lecture halls, large meeting rooms, or airports. In fact, in one the papers by Shroeder, the experiments were conducted and used in a large meeting room successfully.

As discussed earlier, incorporating a Digital Direct Frequency Synthesizer (DDS) will give a great stability and fine tuning as for example, the Qualcomm Q2334. However, since the SSB frequency shifter depends on the quality of the BPF filter, a Digital Signal Processor (DSP) such as TMS32025C would allow a better design and one can adjust the filter accordingly for better performance. Yet, the whole SSB filtering techniques can be performed by using an analog interface to the DSP board, and using a personal computer as a controller to download the algorithm. The Quadrature modulation is preferable for this digital process rather

than the filtering technique. The Weaver technique offers a better solution than the Hilbert technique, because of the simplicity in implementing a Lowpas-Filter over a 90^0 shifter. Furthermore, because of the changing acoustical characteristics from one room to another and the time varying characteristic of the acoustic feedback, an adaptive filtering technique would be most preferable. The adaptive system will be able to adjust itself automatically by changing different weight elements according to different room's response, to suppress the whistling noise of public address-room acoustic feedback. For further research, the adaptive system can be implemented using the TMS32025C or any other member of the DSP family.

As a final note, while the idea turned out to be not original; in fact, it was described in the early sixties, the time for it it may be now, because integrated circuts made it feasible in terms of cost, size and portability. Due to this, we feel the reintroducing the idea at this time may be fortuitous.

APPENDIX A

A.1 Theory of Additional Gain

Reference is made to equation 3.1 which is derived in reference [5]. The frequency shifting system of Figure 3.1, starting with sinusoidal signal input of frequency f_0, it will undergo during its first trip around the feedback loop an amplification (or attenuation) $g(f_0)$ dB. During the second round trip its frequency will have been changed from f_0 to $f_0 + \Delta f$ and its gain will correspond to that frequency, $g(f_0 + \Delta f)$ dB. After N trips around the feedback loop the cummulative gain of the signal will be:

$$cummulative\, gain = g(f_0) + g(f_0 + \Delta f) + ... + g(f_0 + N\Delta f) \qquad (A.1)$$

Wether this expression will diverge towards $+\infty$(instability) or toward $-\infty$ (stability) depends on the average gain of the frequency response. If the average loop gain remains below 0 dB, the system will be stable. Otherwise instability will occur. Thus, frequency shifting permits one to increase the gain by the amount equal to the difference between the maximum gain and the average gain given by equation 3.1.

The sound pressure in the reverberant field of a room above a certain critical frequency is Rayleigh distributed:

$$R(p) = 2p * e^{-p^2} \qquad (A.2)$$

Here the mean square sound pressure, $\bar{p^2}$, has been put equal to unity to simplify the notation.

The sound pressure level in neper, g= ln p, is shown to be distributed as follows:

$$Q(g) = 2e^{[2g - e^{2g}]} \tag{A.3}$$

The mean of this distribution is:

$$g_{av} = -0.2886 Neper = -2.5 dB \tag{A.4}$$

Assuming that points spaced 1/T c/s apart on the frequency response are independent samples, the largest of T*W samples of p^2 is distributed as follows:

$$P(p_{max}{}^2) = T * W * e^{[p_{max}^2]} * \left(1 - e^{[p_{max}^2]}\right)^{T*W-1} \tag{A.5}$$

This follows from the distribution of p.

The most likely value of p_{max}^2 follows from the equation above by diffrentiation. It equals ln(TW). Since the distribution of p_{max}^2 for large TW is relatively sharp, the most likely value of the peak gain approximately:

$$g_{max} = 10 log(ln(T * W)) \tag{A.6}$$

Substracting equation A.4 from equation A.6 yields the desired difference between the maximum gain and the average gain(equation 3.1):

$$g_{max} - g_{avg} = 10 * log(ln(T * W) + 2.5) \, dB \tag{A.7}$$

Reverberation time

The reverberation time T is the time in seconds for a sound to fall to one millionth of it orignal intensity(-60 dB) after stopping the source. Radiotron Designer's Handbook distributed by RCA gives the following equation for T:

$$T = \frac{0.005 * V}{-S * ln(1 - \alpha)} \tag{A.8}$$

Where:

V= volume of room in cubic feet,

S=surface area of walls, ceiling and floor in square feet,

α=average absorption coefficient per square feet, in sabins(values less than unity). The sabin is an absorption unit representing a surface capable of absorbing sound at the same rate as does 1 sq.ft. of perfectly absorbing surface, such as an open window.

In a typical living room, the reverberation time is about 0.5 sec. at 500 Hz and the absorption coefficient is about 0.25.

A.2   Qualcom DDS model-Q2334

Introduction

The Qualcomm Q2334 family of Dual Direct Digital Synthesizers(DDS) generates a high resolution digitized sine wave signals using phase accumalation techniques combined with a patented on-chip sine wave lookup and Noise Reduction-Circuit(NRC). The Q2334 contains two independent DDS functions controlled from a single microprocessor interface. This interface controls bothe the phase and the frequency of the generated sine waves as well as the device's operating mode. Synchronus inputs are also provided to allow for phase and frequency modulation.

The Q2334 provides greater than 76 dB rejection of phase truncation spurs and 72 dB amplitude quantization signal-to-noise ratio. Three pin compatible vesions of the Q2334 DDS are availabe: 20, 30 and 50 MHz maximum clock speeds. This synthesizer is ideally suited for applications requiring high resolution sine wave generation, fast phase and frequency switching, and excellent phase and frequency stability.

The two independent on-chip DDS functions provide an efficient technique for implementation of full-duplex systems, quadrature oscillators, and spread spectrum.(reference QUALCOMM Q2334 Dual DDS Technical Data Sheet).

General Description

The Q2334 device consists of two independent DDS functions each controlled by a comon microprocessor interface, as illustrated in figure A.1 block diagram. Each DDS contains the following:

● Two Phase Increment Registers (PIR), A and B.

● External Multiplex(Phase Increment) Control.

● 32-bit Wide Phase Accumalator.

● Patented Sine Lookup Algorithm.
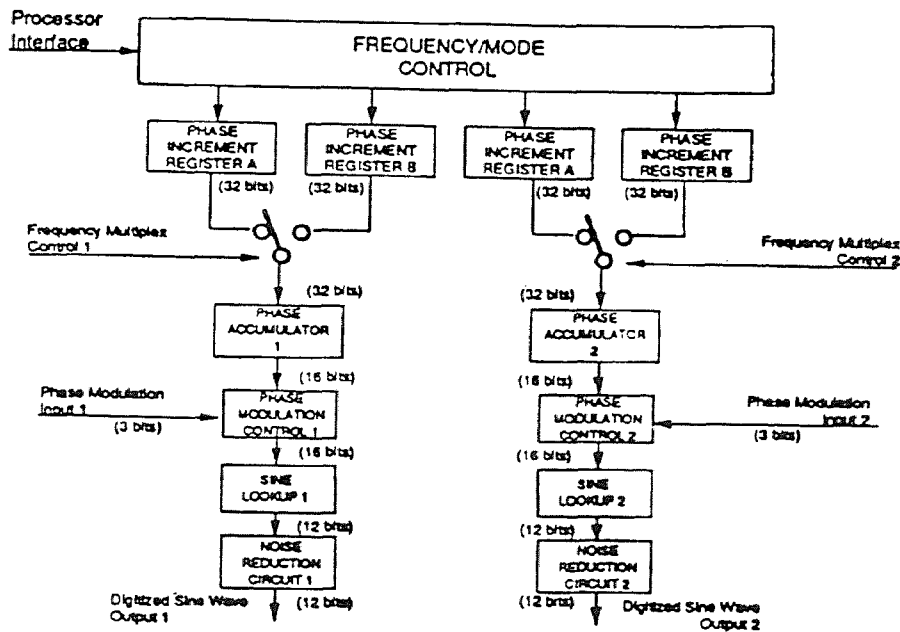
● Patented Noise Reduction Circuit(NRC).

Figure A-1: Q2334 Dual DDS block diagram.

The processor interface controls the phase and frequency of the Q2334 DDS. The specific mode of the Control(SMC) register and the Asynchrnous Mode Control (AMC) register. The SMC register is double buffered. That is, values loaded

into the SMC registers are synchronously enabled when the HOP CLK signal is asserted. The value loaded into the AMC registers does not require activation by any other signal.

Features

● Two Complete Direct Digital Synthesizer Funcions On-chip.

● Custom VLSI Design.

● Processor Interface for Control of Phase and Frequency.

● Patented Algorithm Sine Lookup Table.

● Patented Noise reduction Circuit.

● Synchronus PSK and FSK Modulation Inputs.

● 3 Maximum Clock Speed Versions: 20, 30 and 50 MHz.

● Frequency resolution: 0.005 Hz at 20 MHz Clock Rate.

● Double Buffered Registers Allow Synchronous, Phase Coherent Frequency Change.

● Can be Parralleled for Higher Clock Rate Operation.

● Low Power: 267 mW maximum at 20 MHz Clock Frequency per DDS.

Q2334 Evaluation Kit

A complete Direct Digital Synthesizer(DDS) system based on the Qualcomm Q2334 Dual DDS IC can be implemented using the Q2334 Evaluation Kit. The system includes a pre-programmed microcontroller, a Q2334 Dual DDS chip, a Sony CX2020202A bit Digital-to-Analog(DAC), three slectable analog output ports, an RS-232 terminal port with switch slectable baud rates and switches to allow a basic operating modes to be performed in a Stand-Alone Mode. A ±10 volts power supply and spectrum analyzer are the only equipement required to perform an evaluation of the basic capability of the DDS. The onboard 30 MHz oscillator or an external reference input provides the DDS and DAC frequency reference. Frequencies from D.C. to 12 MHz can be generated when using the on-board anti-alias filtered port. The synthesized waveform can be output in one of thee format: 1.) a direct output

from the DAC, 2.) a low-pass filtered output, and 3.) a low-passed output which is processed by a zero crossing detector to pproduce a TTL logic output, reference is made to figure A-2. This is very useful for applications of the DDS requiring variable logic clock frequencies.
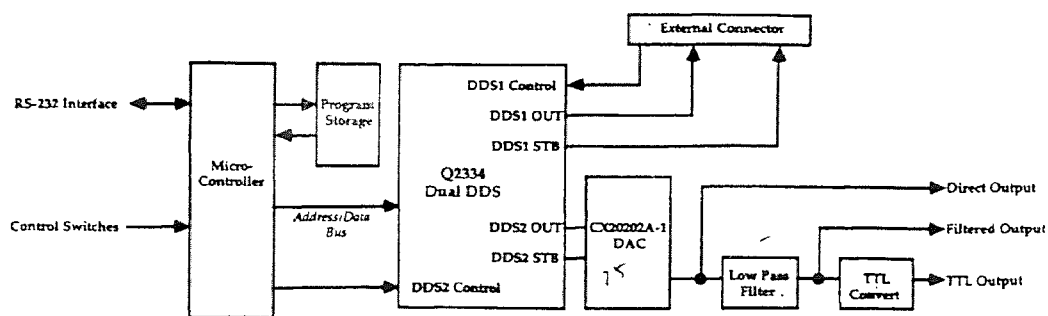
Figure A-2: Q2334 DDS Evaluation Kit block diagram.

The DDS Evaluation Kit comes with menu-driven monitor program which can be used with any RS-232 compatible terminal. Both synthesizers in the Q2334 can be controlled from the terminal keyboard.

The second DDS function on the Q2334 device is connected to an external connector to an external connector, allowing evaluation with user-supplied DAC and analog circuitry. The Kit come complete with schematics and software listings as well as complete intructions for operations.

For a more detailed specifications and applications of either the Q2334 Dual DDS IC or Kit, one should refer to Qualcomm Q2334 Dual DDS technical data sheet, and the User's Guide for Q2334 DDS Evaluation Kit from Qualcomm incorporation in San Diego, California.

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