Virtual AM Stereo and Surround Sound to setup AM/FM Radio Theatre

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ABSTRACT

Introduction of virtual surround sound and stereo to AM radio has been proposed in this study. This technology can be further applied to aid the construction of an AM radio theatre. Adding to the advantages of AM, the lower bandwidth, higher range and simpler circuitry, AM can now offer excellent sound effect with the post-transmission process. The motivation for the introduction of virtual surround sound is the poor quality of AM sound. In this study, the response by human ear has been thoroughly investigated and the methodology to create virtual surround sound has been developed. The elements essential to setup audio theatre such as the components of audio chain, multiple unit audio speaker, inner section of the ear, psychological effect of different ranges of frequency and radio theatre design have been extensively studied on the basis of Helmholtz audition theory. The vital changes include the different frequency division multiplexing of message at the transmitting end, three phases of the process, resulting in the vertical and horizontal digital connection, espresso program and the 3x12 speaker design theatre.

Keywords: Espresso, Surround Sound Theatre and Three Frequency Range.

1. INTRODUCTION

In a concert hall, or, a studio, the background noise sets the lower end of the dynamic range if it is a FM or non-remote-origin AM signal [1][2]. This is higher in a concert hall when compared to a studio which supposed have a sound level of around 10-20 dBA. The highest levels are likely to be found in music symphony orchestra, for instance, can manage momentary peaks of perhaps 0.1K dBA or even more [2]. With other types of music, such as a rock concert, the highest levels may be greater than that of a symphony orchestra but the background level may not be low. As far as the AM sound [2][3] from a radio is concerned, the sound from the music system, even with a surround sound for example, a digital Dolby system, its effect is really flat and the original effect (including stereo) of the signal to the listener is almost lost [1][2]. This paper attempts to adopt the most ideal methodology to make AM effective at the receiving end, and the required components [4][5] and techniques involved in the process are focused. By this methodology, an AM broadcasting system can be setup with a stereo effect at the receiving end.

2. HUMAN EAR RESPONSE

Human ear is an extremely sensitive organ and can detect sound intensity as low as 10^{-12} watt/m^2. When the intensity of sound increases be a factor of 10, the increase in intensity is 1 bel. Thus, the dynamic range of human ear's audibility is 120 decibels [25][26]. When the intensity increases by a factor $10^{0.1}$, the increase in intensity is 0.1bel. If the loudness is α_i for intensity I and $\alpha_{i\,o}$ for intensity I_o,

$$\alpha_i = \beta_i \log_{10} I \qquad \text{Eq. (1)}$$
$$\alpha_{i\,o} = \beta_i \log_{10} I_o \qquad \text{Eq. (2)}$$

The intensity level L is the difference in loudness. So,

$$L = \alpha_i - \alpha_{i\,o} \qquad \text{Eq. (3)}$$

$$L = \beta_i \log_{10} I - \beta_i \log_{10} I_o \qquad \text{Eq. (4)}$$
$$L = \beta_i \log_{10} (I/I_o) \qquad \text{Eq. (5)}$$

When the value of β_i is unity,
$$L = \log_{10} (I/I_o) \text{ dB} \qquad \text{Eq. (6)}$$

If the intensity level changes by 1 decibel, then

$$1 = 10 \log_{10} (I/I_o) \qquad \text{Eq. (7)}$$
$$\text{Or } I/I_o = 1.26 \qquad \text{Eq. (8)}$$

Thus, the intensity level alters by 1 decibel when the intensity of sound changes by 26%. The lowest change in intensity level, which can be detected by the human ear, is 1 decibel.

3. THEORY OF AUDITION

According to Helmholtz Theory of Audition, both forced and resonant vibrations play a vital part in exciting the sensation of hearing. When the sound waves, collected by external ear and sent down the auditory canal, as shown is figure 1, strike the tympanum, because of which forced vibrations are setup [25][26][27]. These vibrations are transmitted through the ossicles, anvil and stirrup to the oval window communicating with the labyrinth. On reaching the cochlea the vibrations are handled on by incompressible fluid in its two chambers to the organ of corti and the basilar membrane. They will ser into resonant vibrations and those fibers of the membrane are tuned to respond to the motions of that particular frequency and thus, the corresponding nerve filaments will be stimulated and transmit the excitation to the brain. The arrangement is that, sounds of different pitches entering the ear simultaneously will produce sympathetic vibrations in different parts of the basilar membrane, stimulate corresponding nerve filaments and give raise to different sensations in pitch, loudness and duration.

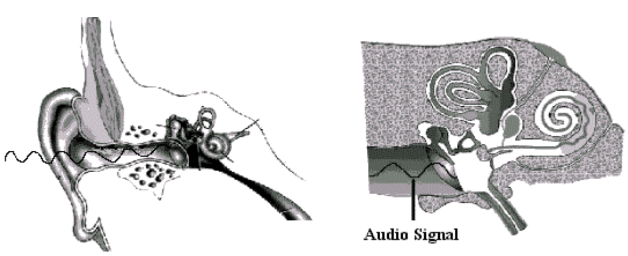

Fig 1 Inner Section Of Ear

The connection also largely depends upon the psychoacoustics. The medium frequency is the one, which really determines the direction of sound. So great care and accuracy is maintained through out the cycle horizontally and vertically, such that it does not cause amplitude variation and distortion on hearing. The tweeter is given lesser importance and the circuit logic is of not much variation to the listener. Moreover, it greatly increases the clarity of the audio signal. Special amplification is given to the low frequency range audio signal, as stereo effect has to match on with the variation vertically and horizontally.

If the non linear response of the human ear is expressed as $\eta=\beta_1\lambda+\beta_2\lambda^2$ where $\lambda=\lambda_1 \cos \varphi_1 t + \lambda_2 \cos \varphi_2 t$, is the sum of two harmonic sound waves, then the non-linear response is given as

$$\eta = \beta_1(\lambda_1 \cos \varphi_1 t + \lambda_2 \cos \varphi_2 t) + \beta_2(\lambda_1 \cos \varphi_1 t + \lambda_2 \cos \varphi_2 t)^2 \qquad \text{Eq. (9)}$$

$$= \beta_1 \lambda_1 \cos \varphi_1 t + \beta_1 \lambda_2 \cos \varphi_2 t + \beta_2(\lambda_1^2 \cos^2\varphi_1 t + \lambda_2^2 \cos^2\varphi_2 t + 2\lambda_1\lambda_2 \cos \varphi_1 t \cos \varphi_2 t \qquad \text{Eq. (10)}$$

Now employing trigonometric identities

$$\text{Cos}^2\varphi_1 t = ½ + ½ \cos 2\varphi_1 t, \qquad \text{Eq. (11)}$$

$$2 \cos \varphi_1 t \cos \varphi_2 t = \cos(\varphi_1+\varphi_2)t + \cos(\varphi_1 - \varphi_2)t \qquad \text{Eq. (12)}$$

The response can be expressed as

$$\eta = ½ (\lambda_1^2 + \lambda_2^2) \beta_2 + \beta_1\lambda_1 \cos \varphi_1 t + \beta_1\lambda_2 \cos \varphi_2 t + ½ \beta_2 \lambda_1^2 \cos 2\varphi_1 t + ½ \beta_2\lambda_2^2 \cos 2\varphi_2 t + \beta_2\lambda_1\lambda_2 \cos(\varphi_1-\varphi_2)t + \beta_2\lambda_1\lambda_2 \cos(\varphi_1+\varphi_2)t \qquad \text{Eq. (13)}$$

4. SOUND PROCESS

At the transmitting end, the AM sound which has to be transmitted is passed to a cross over unit and is divided into low, medium and high frequency. Then the three signals are modulated and multiplexing process (frequency division) is carried out and is transmitted. In the receiving end, the reverse process is carried out, where the signal is demultiplexed and required noise reduction process is carried out. Though, setting the most ideal condition with respect to the response of the human ear is a research domain, because of the non-availability of complete data of how the brain processes the sound signal, it is still feasible to set the ideal surround condition by practically testing and interviewing people of different ages, as the hearing ability differs from one age to the other. With the results of such experiments, the ideal case may be justified.

Now the logical connections are made with input as the digital AM audio signal. There are two phases, which are common to all the three ranges of frequency, low, mid and high. Since the mid-frequency play a vital role in determining the direction of sound, a different logic is implemented. Hence there is logical variation in the phase three between the three ranges of frequency. Suitable CMOS [4][5] can be used from the given table a.1 as per the requirement with respect to propagation delay for the circuit connection.

CMOS Family	Typical Propagation Delay* (nanoseconds)	
	Typical	Maximum
CD 4000 series	50.0	100.0
High-speed CMOS	19.0	024.0
High-speed, TTL compatible	19.0	025.0
Advanced CMOS	04.2	006.7
Advanced, TTL compatible	07.6	013.0

* 5 volts

Tab a.1 CMOS Integrated Circuit Families

Dynamic Propagation delay is a factor, which may intervene, but can be rectified by making all the outputs with some delay such that all the delays equate to same numerical value.

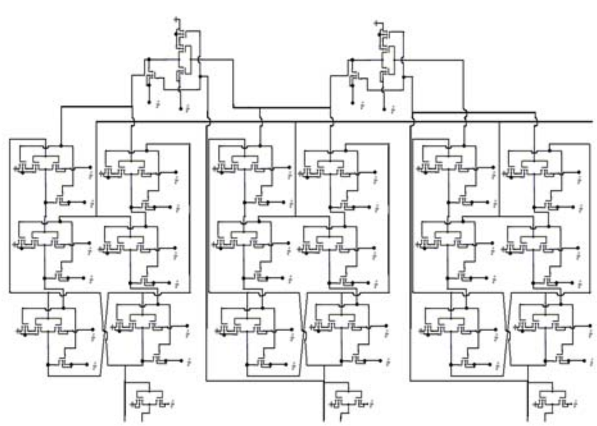

Fig 2 Phase (I)

The above figure 2 shows the first phase in the AM sound processing. The received signal is first digitalized. Then it is sent as input to the above circuit. Here the signal plays the role of a clock pulse. As per the variation in the input, the output varies. Each of the six outputs is feed as input into the second phase, figure 2, where the input is decoded using the Programmable Logic Array methodology. Corresponding to the variations in the output of the first phase, output of the second phase differs. Either should opt for parallel input using delay logic or go for an amplifier. Considering the importance for low frequency ranges and the dynamic signal changes, opting the former option would be better as there is minimal loss of frequencies. Hence the output depends on the parallel output of the designed logic.

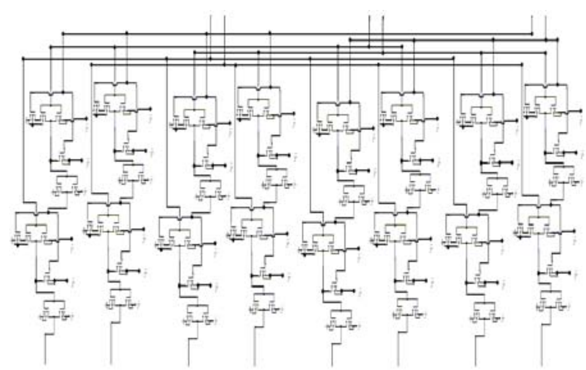

Fig 3 Phase (II)

The phase three is divided into three sections, first, for the low frequency, second, for the mid frequency and third, for the high frequency range. The logical output between the three speakers is as per the table a.2, which is similar to one that of the Espresso program [20]. The output '1' denotes that one of the 3 ranges is given to one of the speakers. For example, if the output is 001 for high frequency, means the high frequency is fed into the third speaker. The second espresso similar connection is made by a single right shift for the medium frequency and the remaining is for the low frequency. Each speaker transmits all the three range of frequencies at different instances of time with respect to the output. For the same, two espresso similar connections are made for the two speakers (low and medium frequency) and the third holds the remaining low frequency in the vertical column of the speakers. Therefore the final output is in such a way that all the three speakers transmit the one of the ranges of frequency each at a time. This sets the vertical connection for the theatre.

1	-00010	010	8	0010-0	001
2	0-0010	010	9	0001-0	001
3	00-010	010	10	0000-1	001
4	000-10	010	11	-11000	010
5	00001-	010	12	1-0100	010
6	1000-0	001	13	01-100	010
7	0100-0	001	14	101-00	010

15	0-1001	010	33	11-111	010
16	10-001	010	34	111-11	010
17	-00101	010	35	11111-	010
18	010-01	010	36	1111-1	001
19	11000-	010	37	-111-1	100
20	00110-	010	38	1-11-1	100
21	1110-0	001	39	11-1-1	100
22	1101-0	001	40	111--1	100
23	1011-0	001	41	1111--	100
24	0111-0	001	42	--- 111	100
25	1100-1	001	43	--1 -11	100
26	1010-1	001	44	-1- -11	100
27	0110-1	001	45	1-- -11	100
28	1001-1	001	46	--1 11-	100
29	0101-1	001	47	-1- 11-	100
30	0011-1	001	48	1-- 11-	100
31	-11111	010	49	-11 -1-	100
32	1-1111	010	50	1-1 -1-	100

Tab a.2 Vertical Logic

5. ELECTRICAL METHOD

The potential divider is called a pan pot, or panoramic potential divider. If the slider is in the center of its travel the microphone's output is split equally between the C and D channels. But if it is nearer one end than the other, at X say, then the signal in the D channel will be greater than in the C channel and we have the required amplitude difference.

Let $\kappa_1=\alpha_1 \sin 2\pi(\mu t-\gamma)/\lambda$ Eq. (14)
And $\kappa_2=\alpha_2 \sin [2\pi(\mu t-\gamma)/\lambda+\phi]$ Eq. (15)

represent the displacement equations of two wave trains of the same frequency range but with different amplitudes and phases, 'ϕ' being the phase difference. The equation of the resultant wave train is given by

$$\kappa=\kappa_1+\kappa_2 \quad \text{Eq. (16)}$$

$$=\alpha_1 \sin 2\pi(\mu t-\gamma)/\lambda + \alpha_2 \sin [2\pi(\mu t-\gamma)/\lambda+\phi] \quad \text{Eq. (17)}$$

$$=\alpha_1 \sin 2\pi(\mu t-\gamma)/\lambda + \alpha_2 \sin (2\pi(\mu t-\gamma)/\lambda) \cos \phi + \alpha_2 \cos (2\pi(\mu t-\gamma)/\lambda) \sin \phi \quad \text{Eq. (18)}$$

$$= \sin (2\pi(\mu t-\gamma)/\lambda) [\alpha_1 + \alpha_2 \cos \phi] + \cos (2\pi(\mu t-\gamma)/\lambda) [\alpha_2 \sin \phi] \quad \text{Eq. (19)}$$

Now, let 'θ' be an angle such that

$$\alpha_1 + \alpha_2 \cos \phi = r \cos \theta \quad \text{Eq. (20)}$$

And $\alpha_2 \sin \phi = r \sin \theta$ Eq. (21)

Then $\kappa=\beta \sin (2\pi(\mu t-\gamma)/\lambda) \cos \theta + \beta \cos (2\pi(\mu t-\gamma)/\lambda) \sin \theta$ Eq. (22)

$$= \beta [\sin (2\pi(\mu t-\gamma)/\lambda) \cos \theta + \cos (2\pi(\mu t-\gamma)/\lambda) \sin \theta] \quad \text{Eq. (23)}$$

Or $\kappa=\beta \sin[(2\pi(\mu t-\gamma)/\lambda)+\theta]$ Eq. (24)

Where,
$\beta=\sqrt{(\alpha_1^2+\alpha_2^2+2\alpha_1\alpha_2 \cos \phi)}$ Eq. (25)

And from Eq. (14) Eq. (15),
$\tan\theta = \alpha_2\sin\phi/(\alpha_1+\alpha_2\cos\phi)$ Eq. (26)

Thus, the resultant wave train is of the same frequency but different amplitude and phase [25][26][27]. If the amplitude are equal i.e., if $\alpha_1=\alpha_2$ and the phases are same i.e., $\phi=0$, superposition gives a vibration of double amplitude, 2α. This conclusion is arrived on the supposition that the interfering wave trains are plane and do not spread out laterally. But for the lateral spreading out of the waves, their amplitudes gradually diminish, and the intensity of the wave will fall off inversely as the square of the distance from the source. Since the structure is a regular hexagon, the sum of amplitudes from the sources at almost all points would be a value close to each other.

6. INTENSITY AND AUDIBILITY

In the figure 4, the Wegel's curves represent the threshold of audibility and feeling, where the frequency is drawn against the intensity [25][27]. When the two curves are extrapolated, enclose an area on the audiogram, called 'The audio sensation area', which represents the auditory sensation area for a normal human ear.

Suppose, the pitch and the intensity of any sound are beyond the limits set by the area, sound is not heard. The ear may feel the sound wave as a pressure but not as an audible sound. The peak sensitiveness of the ear ranges from 2000 Hz to 2500 Hz, and the frequency in music ranges from 40 Hz to 4000 Hz. The intensity that causes maximum painful sensation in the ear is at a frequency of about 800 Hz.

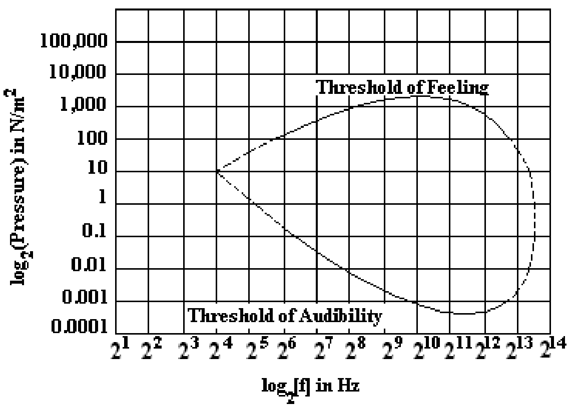

Fig 4 Audibility and Feeling

The figure 5 shows the most precise design of a Radio theatre with 36 speakers with 3 speakers in each vertical column and 2 vertical columns at each corner, by which a better surround sound can be obtained, with respect to form equilateral triangles. As shown in the figure 5, the amplification is alternative, aiding the stereo effect. The hexagonal shape forms a number of equilateral triangles.

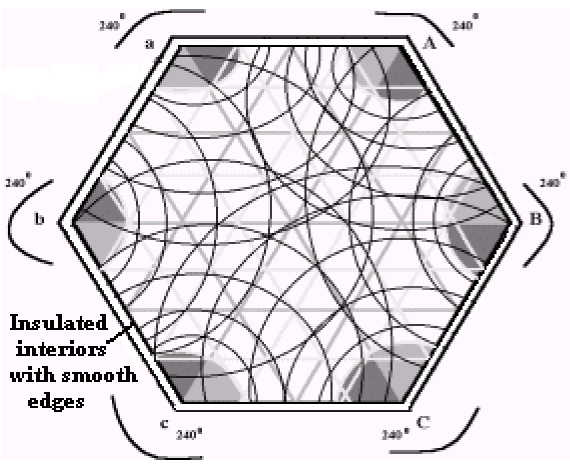

Fig 5 Theatre Design

Because of that, the listener can feel the surround sound from each angle of 60^0, which consists a pair of unequally amplified speakers. In the low frequency and high frequency speakers, the logical connection is given such that alternatively the extra-amplified sound is heard.

But for the medium frequency the logical connection is made using the extended Phase II connection where the extra-amplified sound is made in a cyclic fashion in steps of 2. This sets the horizontal connections for the theatre.

In AM Stereo, the sum and difference channels take different paths through the transmitter, as opposed to FM, where both go into a composite base band, which goes through the transmitter in a single path. In FM, we limit, compress and clip the left and right channels individually in order to create the maximum RMS. In AM Stereo, 50% left plus 50% right equals 100% L+R, and the L+R is the mono sum that most of the listeners hear. Lose the left or lose the right and there goes half the modulation and half the apparent loudness. Since one also process L-R, one can increase the RMS of the difference and increase the apparent separation by several dB.

7. DISCUSSIONS AND CONCLUSION

The ideal listening position is at an apex of an equilateral triangle, the loudspeakers being at the other two points. Curious imaging is apt to occur if the listener is closer than this. Satisfactory listening occurs further back, provided the listener stays on the centerline. The listening room should be acoustically symmetrical about an axis formed by the centerline between the loudspeakers. Audio transmission and reception are the two factors that really concerns whenever one is trying to setup a long distance audio communication system. It is AM that plays a major role for the above challenging research domain, since the effect is not the same in AM as in the case of FM.

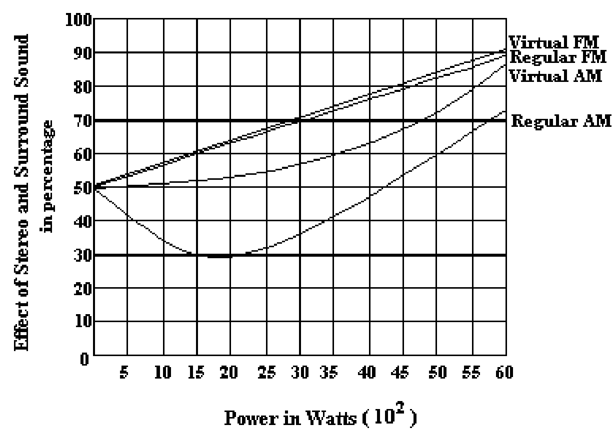

Fig 6 Effect of each Signal against Volume

As shown in the figure 6, this new concept of Virtual Surround Sound would give almost 95% of original effect to the listener. Thus, this concept, methodology and architecture may help in achieving the almost effect that could be felt with any other surround sound system, and may even give a better effect for specific sound inputs.

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