An Experimental 9000-Watt Airborne Sound System*

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Summary—**An experimental 9000-watt speech announcing system AN/AIC-11(XA-1) was developed for installation in a B-26 aircraft. The system was used for studies of direct communication through the atmosphere to ground personnel from aircraft operating at relatively high altitude. The equipment consisted of a turbine generator type of auxiliary power unit; three 3000-w amplifiers, each driving a separate twin-horn loudspeaker; signal preparation, control, and monitoring units; a loudspeaker mounting frame which rotates the loudspeakers and supports two of the twin horns outboard from the fuselage; and magnetic tape recorders.**

Introduction

THE EXPERIMENTAL 9000-w airborne sound system to be described was intended for the investigation of direct acoustical transmission of information to listeners on the ground 6000 feet below an aircraft moving at a speed of 160 mph. There are various tactical, operational, psychological, and rescue purposes for which successful transmission under these conditions would be useful.

Small public address systems (having an audio power of about 100 w) have at times been used successfully in small civilian aircraft for advertising purposes. They have generally operated over small towns where the ambient noise level is low and the area of coverage is small, simply by circling over the target area at minimum engine power, airspeed, and altitude, and directing the loudspeaker horn toward the center of population.

When larger noisier areas are to be covered more rapidly, or when higher altitude and speed are operationally necessary, it becomes much more difficult to obtain satisfactory speech transmission through the medium and adequate sound level at the ground. The reasons for this difficulty have been given in a separate paper,[1] which reports data obtained from the system described here and earlier systems. Important factors include turbulence and gross inhomogeneity in the medium (resulting in random fluctuations in level), the effect of source motion in speeding up the fluctuations, multipath interference, normal attenuation with distance, high-frequency absorption in the medium and Doppler effect. The effect of combined controlled distortions of these types upon the intelligibility of speech was investigated experimentally in the laboratory and has been reported in detail elsewhere.[2]

Operating conditions for the equipment were influenced greatly by the type of aircraft which had been selected. The choice of B-26 had been made because of tactical reasons and the availability of the aircraft for the purpose at the time the project began. An announcing system of lower power, which had been used in a B-26 with limited success in Korea, was intended to be improved or superseded by this project. The end result sought was "good intelligibility for at least 30 seconds at any point within $\frac{1}{4}$ mile of the aircraft's flight path projection on the ground."

System Planning

Preliminary calculations and measurements on the earlier system indicated that its efficiency was good; that its intelligibility (as recorded several feet below the loudspeakers during flight) was high; that the intelligibility of speech received at ground level was submarginal; and that the power-delivering capacity needed to be increased by an order of magnitude if possible. By installing a new type of gas turbine, which had recently been developed by the Solar Aircraft Company, it appeared possible to obtain sufficient primary power for a 9000-w audio system.

The only high-power audio amplifiers available in the power range needed for this project were many times too heavy for airborne operation. Therefore it was necessary to develop a power amplifier specifically for the purpose. For reasons given later it was decided to divide the power amplification system into three parts.

Fig. 1 is a block diagram of the experimental system. Starting, monitoring, and stopping the Solar gas turbine was accomplished by the operator through the use of an engine control box. The starter motor for the turbine was used as a dc generator after the engine has reached operating speed. This change in function was accomplished by the dc power control box. Direct current was supplied to a 60-c inverter for 110-v alternating current for the tape recorder-playback unit and the test equipment power outlets in the control cabin. Twenty-eight volts dc from the turbine generator was also supplied through the amplifier power control panel to the filament circuits of the three 3000-w amplifiers. Small amounts of dc power were also used for the signal preparation unit and for the motors of the loudspeaker rotator.

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[1] D. W. Martin, R. K. Duncan, and R. L. Murphy, "Vertical Atmospheric Transmission from a Moving Sound Source," paper presented at the Second Int'l. Congress on Acoustics in conjunction with the 51st Meeting of the Acoust. Soc. of Am., Cambridge, Mass.; June, 1956.

[2] D. W. Martin, R. L. Murphy, and A. Meyer, "Articulation reduction by combined distortions of speech waves," *J. Acoust. Soc. Am.*, vol. 28, pp. 597–601; July, 1956.

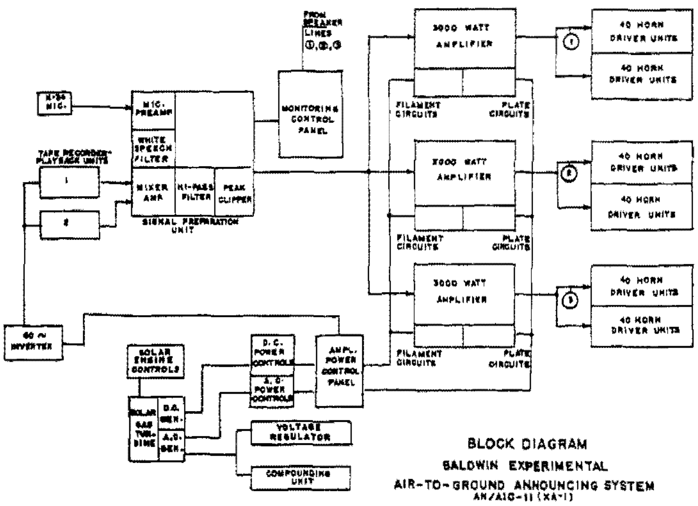


Fig. 1—Experimental 9000-watt airborne sound system —block diagram.

A 400-cycle 15-kva Eclipse-Pioneer alternator was mounted on the turbine. Associated with this was a voltage regulator and a compounding unit. AC power to the power amplifiers was controlled through relays actuated by switches on the amplifier control panel. The output from each amplifier was supplied to two banks of 40 horn-driver units each, each bank driving a separate half of a twin horn. The central twin horn was mounted in the under side of the bomb bay, and the other two twin horns were mounted outboard. Either prerecorded tape material or direct speech from the operator using an M-34/AIC microphone could be supplied to the power amplifiers through the signal preparation unit. A standard Air Force preamplifier and dynamotor of the AN/AIC-10 type were mounted in the signal preparation unit. The output of the preamplifier passed through a special electrical filter which equalized the transmitted speech spectrum to a uniform speech spectrum, so that the maximum practical pre-emphasis of high-frequency energy would be supplied to the power amplifiers. Both tape and microphone signals passed through a high-pass filter which protected the loudspeakers, and a peak clipper which protected the amplifiers. The monitoring control panel permitted the operator to check signal levels at all necessary points in the signal preparation unit, and to monitor the output of each power amplifier both by vu meter and by headset listening.

System Development

Auxiliary Power Unit

The aircraft engine generators could not supply the necessary primary power. Extra pads on each engine were not suitable for mounting alternators of the required capacity, and conventional auxiliary power units having sufficient capacity were much too heavy and large to be considered for installation in the B-26 aircraft. A Solar type M-1 gas turbine engine (Mars 50 hp) had recently become available. In addition to the 15 kva, 28-v dc generator included in the M-1 equipment, an Eclipse-Pioneer type 28EO3-3-B alternator was driven by the turbine engine. Fuel was obtained from the reserve fuel tank of the aircraft. The entire auxiliary power plant was installed in the rear turret compartment after the turret was removed. Installation of the turbine (and of other equipment described in this paper) was accomplished by WADC personnel. The Solar equipment performed well for the duration of the project with only normal maintenance and care.

Signal Preparation Unit

An obvious correction for the high-frequency losses in the atmosphere would be to pre-emphasize the speech spectrum at high frequencies. However, the atmospheric losses are far greater than the amount of droop in a normal speech spectrum. To pre-emphasize the high-frequency components beyond a uniform speech spectrum condition[3] would be a poor compromise, because it would necessitate lowering the amount of power transmitted by the system in the frequency range where transmission is best. In order to achieve the uniform speech spectrum condition, an equalizer filter was designed having a response characteristic inverse to the speech spectrum transmitted by the M-34/AIC microphone from a typical speaking voice. This equalizer was installed in the signal preparation unit, just following the microphone preamplifier, and ahead of the mixer amplifier stage, so that it would have no effect upon prerecorded tape material.

Both speech signal and tape playback pass through a 400-cps high-pass filter which prevents the loudspeakers from being driven excessively below horn cutoff frequency. This was followed by a symmetrical peak clipper. The threshold of peak clipping was adjusted to protect both the power amplifiers and the loudspeaker groups from peak overload.

Fig. 2 is a schematic diagram of the signal preparation unit. Not shown are the microphone (and its press-to-talk switch), the microphone-recorder selector switch, the monitoring headset, the vu meter, and the meter circuits and selector switch. The standard Air Force interphone components are indicated by blocks only. The speech spectrum equalizer is shown at upper right, the mixer amplifier at lower left, the high-pass filter at lower center, the peak clipper at lower right, and a separate monitoring headset amplifier at upper left (which amplifies from meter-circuit level up to a level suitable for listening in aircraft noise). Note that metering and monitoring connections are provided at various circuit points.

[3] D. W. Martin, "Uniform speech-peak clipping in a uniform signal-to-noise spectrum ratio," *J. Acoust. Soc. Am.*, vol. 22, pp. 614–621; September, 1950.

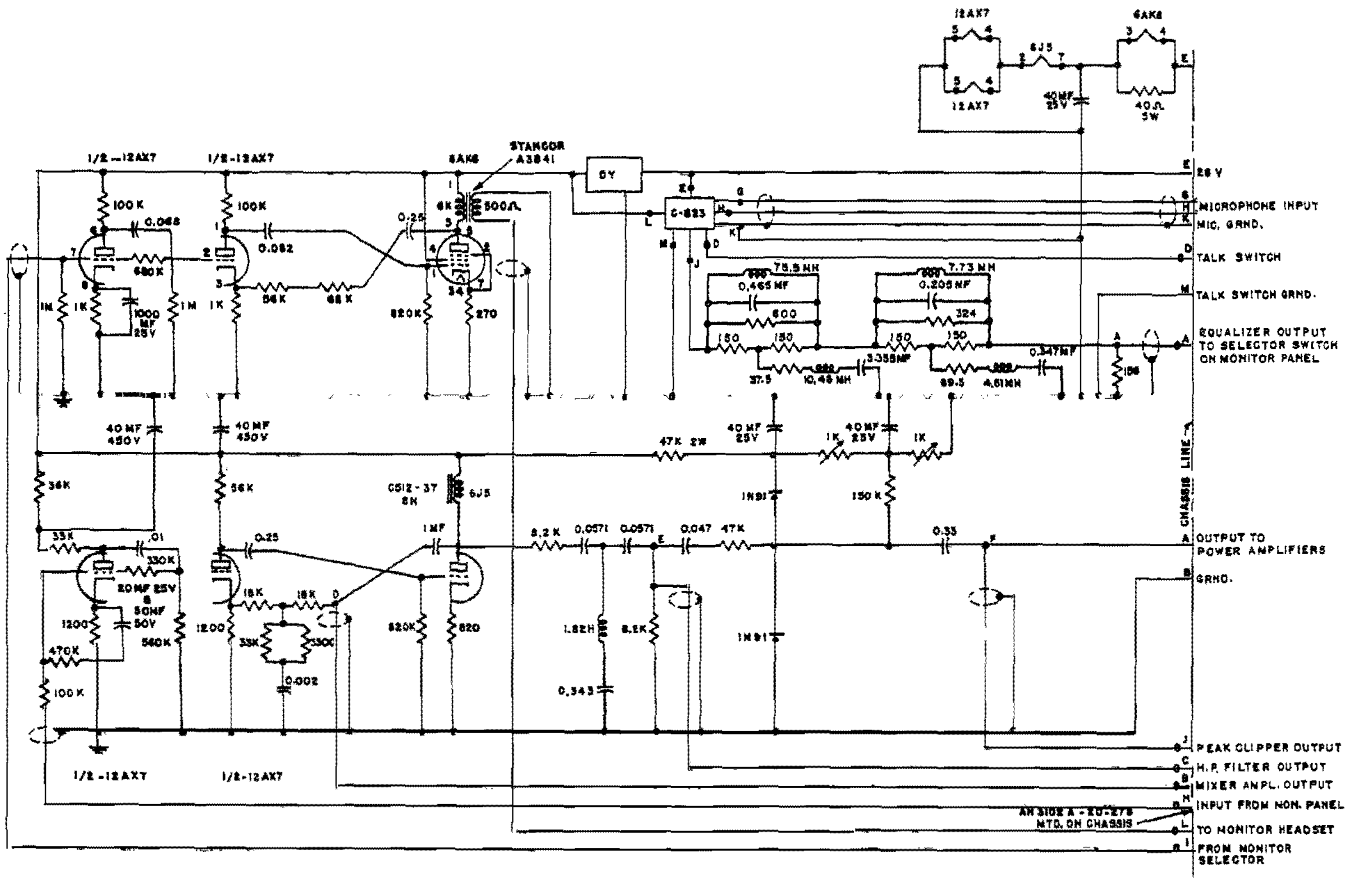


Fig. 2—Signal preparation unit—schematic diagram.

3000-Watt Amplifier Unit

For maximum electrical dependability, and because it appeared physically necessary to divide the loudspeaker system, it was decided to divide the 9000-w system into three equal parts. Initially, triodes were decided upon for the output stage of the power amplifier, because of the low source impedance obtainable even without feedback. It was not certain that feedback could be used at all without complicated correction to insure unconditionally stable operation. Tube type 304TH was chosen because of the very low driving power required (according to published characteristics), and the small number (four) of tubes required for 3000-w power-delivering capacity, minimizing parasitic problems. For the 3-kv plate supply for the output stage four 872A's in a bridge circuit with a swinging choke (5–25h) and a 2 mfd filter capacitance provided more than adequate supply and filtering. A schematic of the 3000-w amplifier is shown in Fig. 3 (opposite). Each half of the output transformer consisted of 295 turns of No. 20 heavy Formvar. Each half of the secondary consisted of twelve parallel turns of No. 12 heavy Formvar. The laminations consisted of a five-inch stack of No. 29 Allegheny Ludlum Audio A, EI-19. The over-all dimensions of the transformer were $10 \times 7 \times 5$ inches. After the first model was constructed it was found that considerably increased driving power would be required. This was confirmed by revised specifications from the manufacturer of the output tubes. A new type of audio power amplifier,[4] which had been developed at Baldwin for possible commercial use, was adapted to the needs of the driving amplifier stage. The commercial amplifier circuit developed by Bereskin was adapted to give 70 w, using two 807's with a specially designed bifilar output transformer for driving the 304TH's, and a special 400-cps power transformer.

After these changes it was determined that the power output capability was 3200 w, and the 400–4000 cps response at this level was flat to within plus or minus 0.1 db. However, there was considerable distortion caused by variation in the bias voltage moving the operation slightly into class C at full power. Installation of a triode bias control tube reduced this variation, so that at 3000 w the harmonic distortion was reduced to 10 per cent.

2.5 μh chokes were inserted into each plate lead in the output stage, and were completely effective in stabilizing the operation. Hum and noise voltage at the 8-ohm output was less than 0.25 v rms, compared to 155 v at 3000 watts output. In the actual installation the hum voltage was somewhat higher because of system wiring. Only a few db of feedback could be applied while maintaining stability, so none was used in the final model. All power switching was accomplished by relays installed on the underside of the chassis.

[4] A. B. Bereskin, "A high-efficiency, high-quality audio-frequency power amplifier," IRE Trans., vol. AU-2, pp. 49–60; March, 1954.

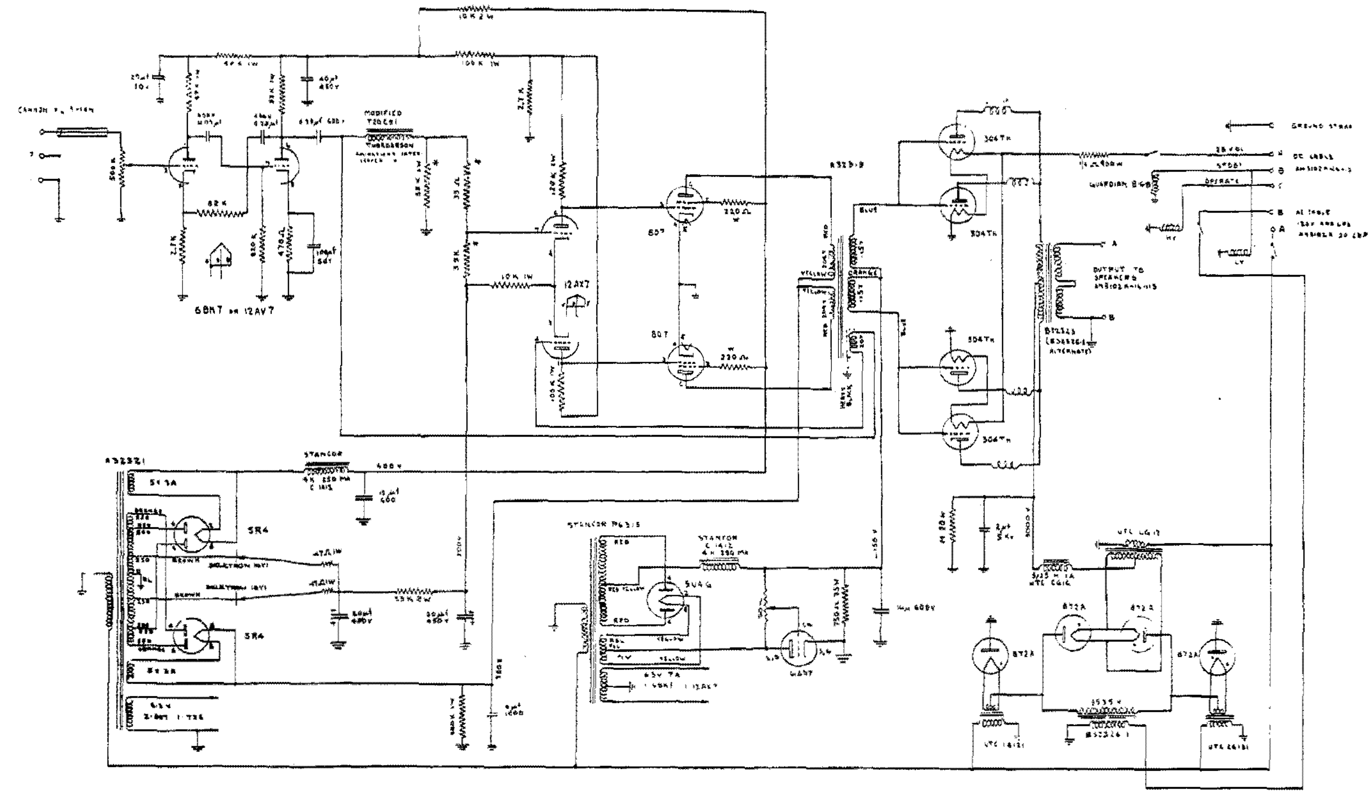


Fig. 3—3000-watt power amplifier—schematic diagram.

Twin-Horn Loudspeakers

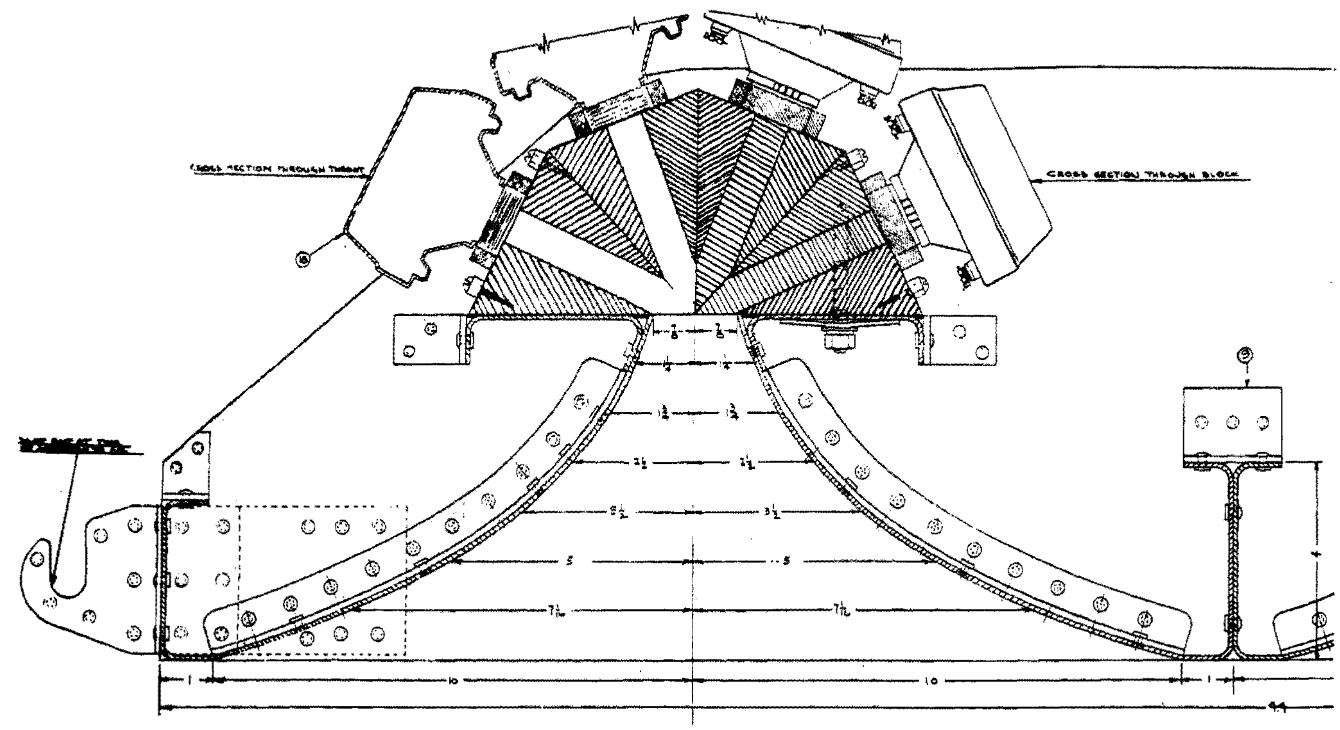
The large acoustical power-delivering requirement necessitated the use of horn-driver units having maximum power-delivering capacity per pound of driver mass. Random noise (500–4000 cps band) was used in a series of life tests on individual driver units at various input powers. The sound pressure output level was monitored to determine the relative power-delivering capacity of each type of driver which appeared to meet the requirements. The University LB-35 unit (having a metal voice coil form with serrated edges) was adopted on the basis of these tests. This type of unit (when suitably horn loaded) operated for periods of several hours at 37 w. Higher powers did not yield corresponding increases in sound pressure level. The units failed due to heating after one-half hour at 45 w. On the basis of these tests the number of driver units required per 3000-w amplifier was estimated to be 80, or a total of 240 units in the system. At this level, however, loudspeaker distortion is high at high frequencies.

The desirability of connecting all of these driver units to a single horn (to minimize interference effects in the directional pattern) was recognized from the beginning. However, a number of other considerations prevailed, both theoretical and practical. It was recognized that nonlinear distortion would be a factor, and it was desired to keep this to a reasonable percentage if possible. At high frequencies the distortion is a function of the low-frequency cutoff of the horn,[5] and it is advisable to place this cutoff (determined by the rate of horn flare) near the lowest frequency that it is planned to reproduce from the horn. The rate of flare chosen in the design corresponds to a low-frequency cutoff of 400 cps.

Mechanical layouts were made for the throat block on which horn units would be mounted. In the design, it was necessary to keep the driver units as close together as physical dimensions would practically permit; to make the horn length within the throat block short, so that the rapid rate of flare would not cause the horn cross-section dimensions to exceed a wavelength where the short horn sections join at the throat of the horn extension, at the highest frequency to be reproduced; and to keep the design practical from a maintenance standpoint.

Fig. 4 shows the throat block adopted, in two sectional views. Half of the block is shown in cross section through the horn throats, and the other half is shown through the solid block between throats. The horn sections were of uniform width inside the throat block, with the exponential expansion occurring in planes at right angles to this view. Rail construction, commonly used in the piano industry, was adopted in order to simplify the construction. Driver units were mounted on aluminum bars running lengthwise of the throat block. Terminals for loudspeaker wiring were mounted on brass strips running lengthwise between rows of driver units. The throat block itself was constructed of wood rails of triangular cross section, between which were glued curved pattern blocks with suitably tapered edges to give the desired horn contour. In a design for produc-

[5] M. Y. Rocard, "Sur la propagation des ondes sonores d'amplitude finie," *C. R. Acad. Sci., Paris*, pp. 161; January, 1933.



HALF TWIN HORN

Fig. 4—Half twin horn—cross section.

tion, the throat blocks would be cast, of course, but in a research model system the wooden construction was expedient, light in weight, and economical considering the small quantity required.

The associated horn was fabricated of riveted sheet aluminum in the aircraft assembly department of the Baldwin factory. Horn mouth width was only a little greater than the over-all width of the throat block and driver unit assembly, and provided each unit with a 45 to 1 ratio between the area of horn mouth and throat, entirely adequate for efficient loading.

It was the design intent to make the angle of coverage small in the plane at right angles to the flight path, because the width of the coverage area specified was only one-half mile. This dictated that the horn mouth should have its long dimension across the fuselage.[6] Yet the fuselage width was sufficient to permit a throat block only ten driver units in length. Extending the horn outboard on each side of the fuselage a similar distance, which was almost as far as space between the fuselage and the engines would permit, provided only an array of 30 by 4 or 120 driver units, one-half of the number needed for the power. Consequently, it was decided to use a twin horn in each of the three locations, each half of each horn to have the design shown in Fig. 4. The end pieces of the horn were common to both halves, and were designed with the loudspeaker mounting frame and rotator assembly in mind. A perspective view of a twin horn (Fig. 5) shows the driver units, end plates, and mechanical bracing. Fig. 6 is an on-axis view which shows the detail of the throat block pieces which provided the blend of the small horn sections into the larger horns.

The driver units were connected in a series-parallel arrangement giving each twin-horn loudspeaker a nominal impedance of 8 ohms. In an initial test one side of the system was grounded. The large potentials between the voice coil and the grounded case of the units connected farthest above ground potential caused arcing and failure of a number of units. A center-grounded arrangement was then adopted. This change plus a 300-v breakdown test on all driver and replacement units reduced loudspeaker unit failures to a minimum. Experience with this problem emphasized the importance of ease of unit replacement in a loudspeaker system of this size, and the development of a good maintenance check for loudspeaker units. The most practical method of checking driver units, with the aircraft on the flight line, was to measure the current into each driver unit with a "clip-on" ammeter, while a steady audio signal was supplied from each power amplifier. Ear plugs plus a headset are needed for aural protection in this test.

Response and impedance characteristics of one of the three twin-horn loudspeaker systems are shown in Fig. 7. The response-frequency characteristic is rather uniform throughout the frequency range of 400 to 3700 cps. The nominal impedance of 8 ohms is a fairly conservative value in this same frequency range.

[6] H. F. Olson, "Elements of Acoustical Engineering," D. Van Nostrand Co., Inc., New York, N. Y., p. 25; 1940.

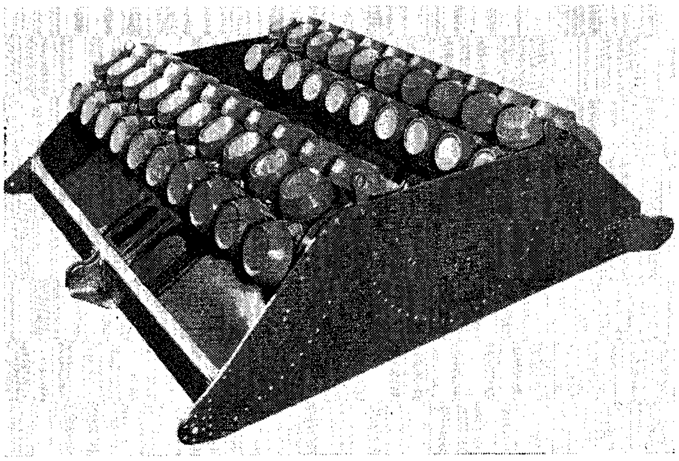


Fig. 5—Twin-horn loudspeaker—perspective view.

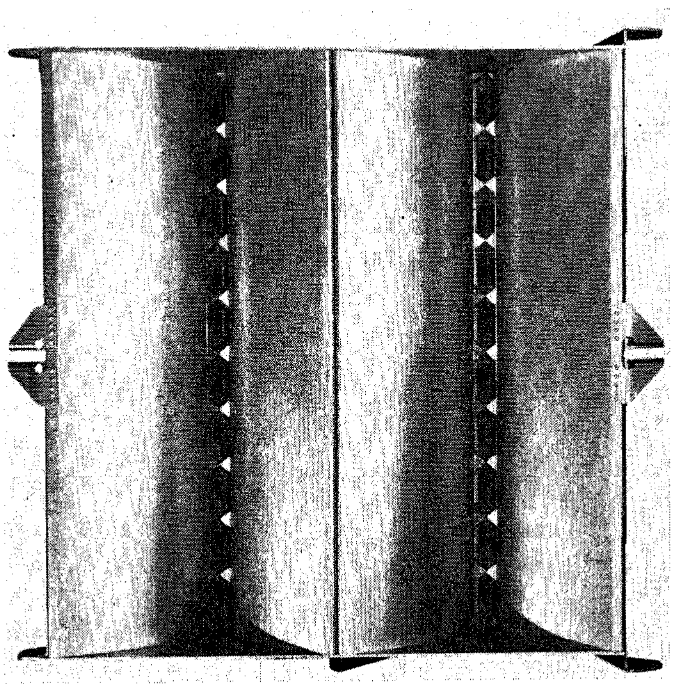


Fig. 6—Twin-horn loudspeaker—axial view.

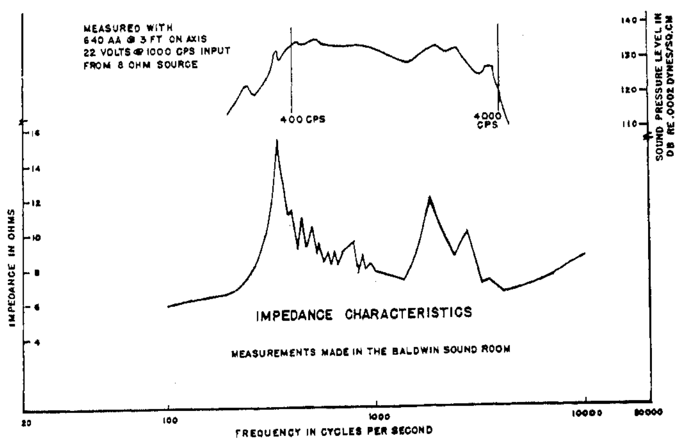


Fig. 7—Twin-horn loudspeaker—response and impedance frequency characteristics.

Fig. 8 (next page) shows the polar characteristics of a twin horn in its plane of symmetry (transverse to aircraft flight path) at a distance of 12 feet at frequencies of 400, 500, 1000, 2000, 3000, and 4000 cps. This is the plane in which a narrow beam had been the design aim. It is apparent that this aim was achieved. Addition of the other two twin horns in line in this same plane increases the directivity even further.

Fig. 9 shows the directional characteristics in the plane of the horn flare (containing the aircraft flight path). The directivity pattern in this plane is much broader, as desired, but because two horn sections were used, the directivity patterns are considerably more nonuniform than would be ideal for the purpose.

Loudspeaker Rotator

Time of effective transmission of speech had been recognized as one of the greatest factors limiting the usefulness of airborne announcing systems. Although the loudspeaker system was designed to have a fairly broad directivity pattern in the plane of the line of flight, it was decided to mount the loudspeaker system on a motor driven rotator, in order to extend the period of transmission. It was not feasible from an acoustical standpoint to rotate the horn system in such a manner that part of it would be inside the fuselage when rotation occurred. The relatively small clearance between the fuselage and the ground before take-off and after landing made it impractical to mount the loudspeaker system below the aircraft. Consequently a system was devised in which the rear edge of the loudspeaker system would be low at the beginning of the rotation cycle; then it would be raised until the loudspeaker was horizontal; finally the front edge would be lowered to complete the cycle. Screw jacks driven by dc motors controlled by relays were used to accomplish the rotation cycle. A control panel in the pilot's compartment provided visual indication of the status of the rotator. The automatic cycle was completed in approximately 30 seconds which doubled the effective period when highly directional loudspeakers were used.

System Installation and Evaluation

Fig. 10 is a greatly expanded view of the equipment operator's compartment. Actually this space was extremely crowded. The installation of all major equipment items is shown in phantom in Fig. 11. Also shown is a wind-screened monitoring microphone installed under the central loudspeaker, with which the output of the system could be measured or monitored at any time during flight. The aircraft is shown in Fig. 12 during an actual flight test, when the monitoring microphone was lowered on cables at various distances below the loudspeaker system, in order to evaluate the intelligibility of the speechwave as a function of distance. At the time of this test the outboard loudspeakers had not yet been

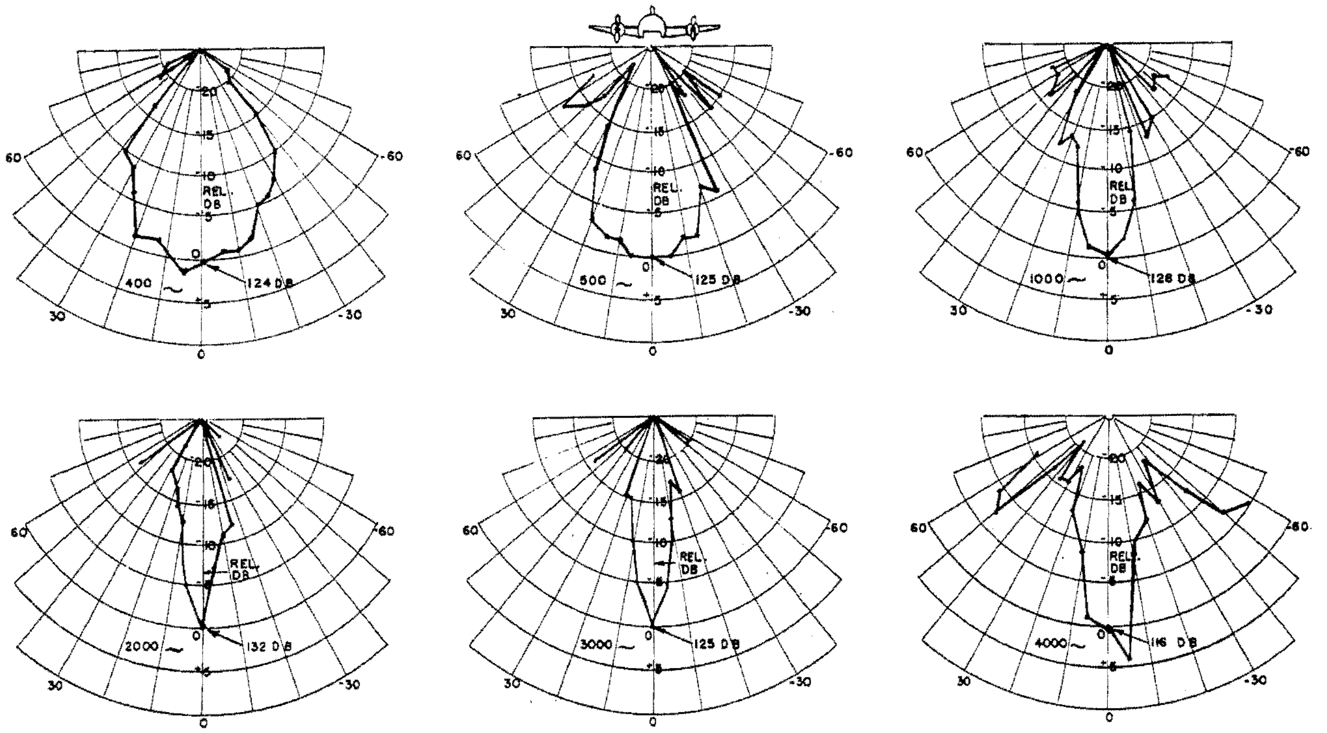


Fig. 8—Twin-horn loudspeaker—directional characteristics in the plane of symmetry.

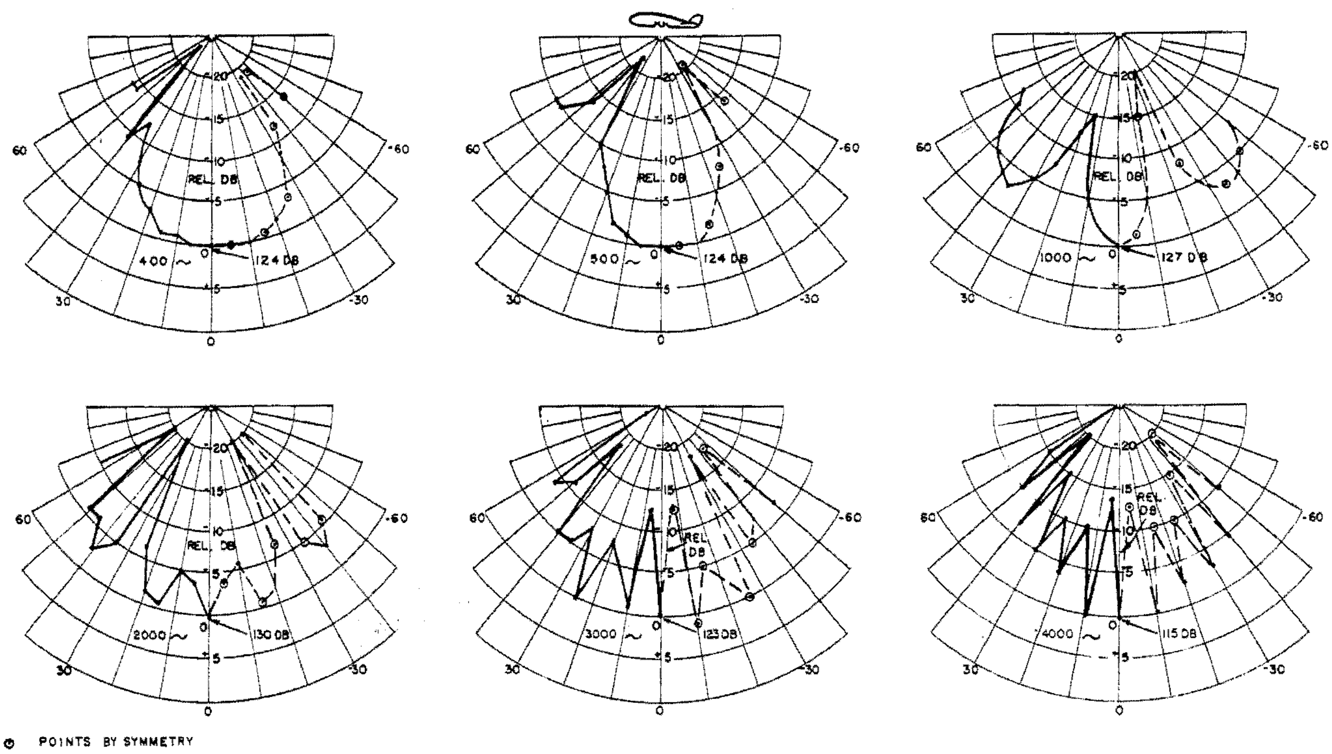


Fig. 9—Twin-horn loudspeaker—directional characteristics in the plane of flare.

installed (the nose wheel was not retracted because one of the supporting cables passed through this space).

Extensive flight tests over the rural Baldwin Ancor plant established that the increased power and improved directivity pattern increased the transmitted speech level by approximately 10 db. It was also found that reducing the air speed to 120 mph increased the average transmitted level approximately 6 db more. Width of coverage was measured and recorded by microphones located $\frac{1}{4}$ mile on each side of the projected flight path. Without the loudspeaker rotation the number of words per pass correctly received was increased nearly 3 to 1

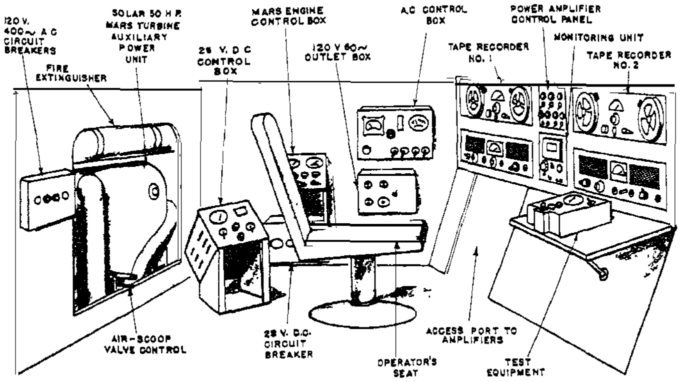


Fig. 10—Expanded view of equipment operator's compartment.

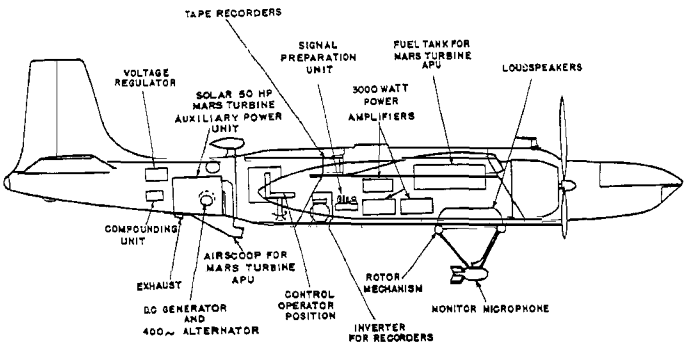


Fig. 11—Location of major equipment items in B-26 aircraft.

Fig. 12—B-26 in flight with sound system and suspended microphone.

over the earlier system, and more than 4 to 1 when the aircraft speed was slowed down.

Although the combination of improved equipment and lower airspeed greatly improved word reception, it should not be assumed that really good communication was finally achieved. Rather the performance was increased from "sub-marginal" to "marginal." Special techniques which were found useful in improving the performance further were the prerecording of messages on tape, directing the loudspeaker toward the target, the use of selected vocabulary and, particularly, redundancy. For listeners within $\frac{1}{4}$ mile off the projected flight path, reception of 15-second messages on basic topics repeated several times during the same pass, were received quite well. Air-to-ground transmission from high altitude at speeds in excess of 100 mph (but not greater than 200 mph) was concluded to be feasible under favorable weather conditions, although word intelligibility is low by conventional standards.

Proposed Improved System

During the latter part of the project two major equipment developments occurred which could be applied to a future design for production. Both of these developments reduce the space and weight requirements, and together they permit a revised installation layout which, according to preliminary estimates, would permit restoration of the armament complement of the aircraft.

Single-Flare Horn

Although it still seemed impractical, from a loudspeaker maintenance standpoint, to construct a throat block with more than four horizontal rows of driver units, the two throat blocks from a twin horn were adapted to be mounted on a wooden single-flare horn model shown in Fig. 13 (next page). A possible disadvantage of this arrangement is the necessity for a lower cutoff frequency (approximately 240 cps), tending to increase the distortion at high frequencies. The response characteristic of the single-flare horn was somewhat rougher on the axis than for the twin horn, but this irregularity was minor compared to that of the twin-horn response curves at angles off the axis.

As expected, the directional characteristics in the plane of symmetry resembled closely those for the twin horn. Fig. 14 demonstrates the expected improvement in smoothness and broadness of directivity pattern in the plane of horn flare. This type of horn is recommended for future systems if a practical means can be devised for servicing the loudspeaker units, such as by hinging the horn flares near the throat block to give access to the lower rows of units. With such a broad directional characteristic, the loudspeaker rotator could be eliminated.

Bereskin Amplifier

During the completion of the development and construction of the 3000-w power amplifiers previously described, it was decided to attempt circuit development for an improved type of amplifier which could be used to advantage in a follow-up design. This circuit development has been described elsewhere.[7] The laboratory model constructed, operating from separate laboratory power supplies, showed great promise in reducing the circuitry, the number of tubes, the size and weight. It also provided the nominal audio power at very low distortion. Possible disadvantages were the necessity for forced-air cooling (which might be provided easily in an airborne system) and increased plate potential for the

[7] A. B. Bereskin, "A 3000-watt audio power amplifier," IRE Trans., vol. AU-4, pp. 37–41; March, 1956.

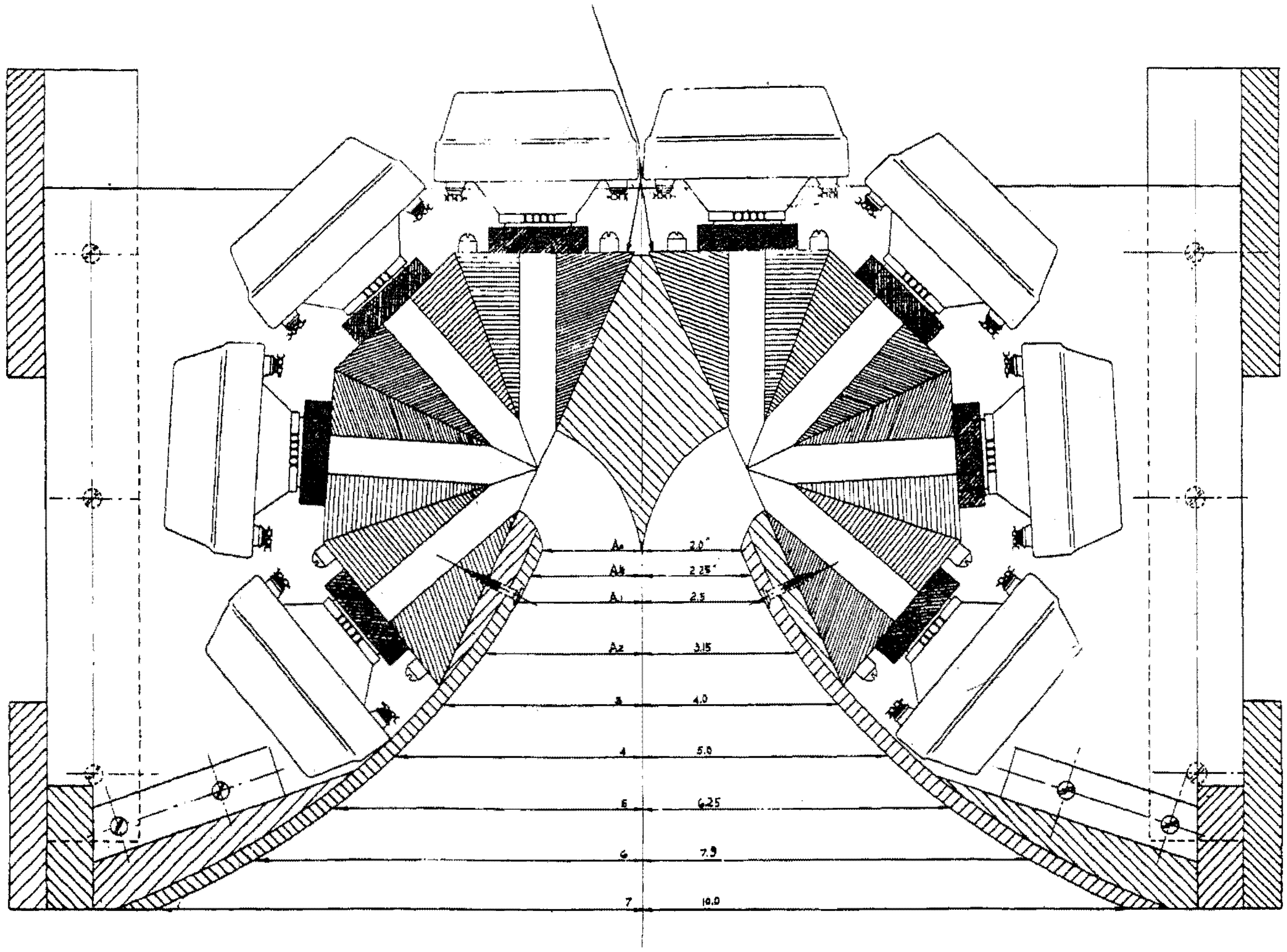


Fig. 13—Experimental single-flare horn cross section.

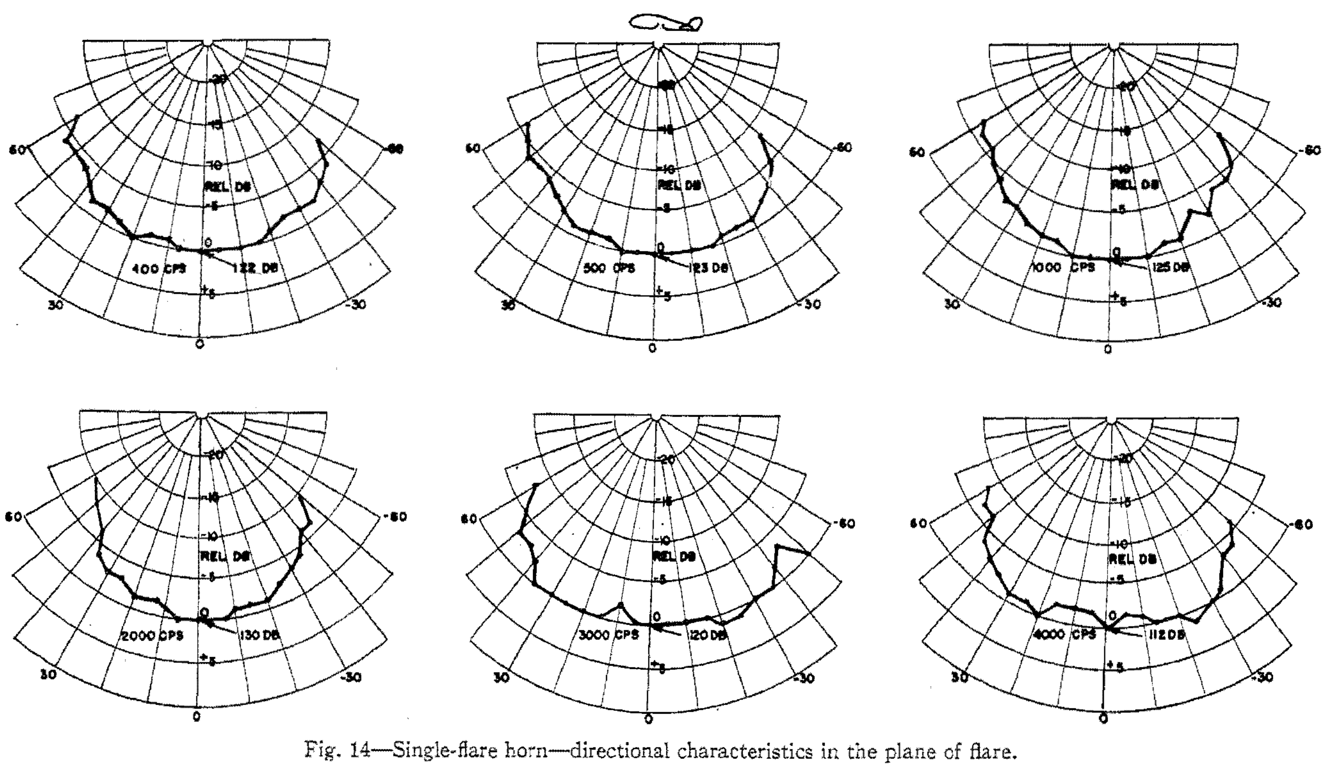


Fig. 14—Single-flare horn—directional characteristics in the plane of flare.

output tubes. The circuit employed a pair of 4-1000A tubes operating Class B_1 push-pull with a 5-kv plate supply voltage and 1.25-kv screen voltage. The driver stage used a pair of 6AU6 tubes. Consideration of this amplifier for any future airborne systems of this type is strongly recommended.

Proposed System

Through reduction of the front-to-back dimension of the loudspeaker system, using the single-flare horn, and reduction of the power amplifier dimensions, it would be possible to install the auxiliary power unit in the same compartment now occupied by the loudspeakers and amplifiers. Proximity of all the major equipment items would facilitate integration of controls into a more compact package. It would also permit restoration of aircraft armament to spaces now required for audio equipment and auxiliary power unit installation. Reduction in size and weight, and integration of the type proposed might also provide a system which could be installed more easily in aircraft of different types.

ACKNOWLEDGMENT

This development was done in close cooperation with WCLNE3, Communication and Navigation Laboratory, Wright Air Development Center. E. Lazur, L. N. Theroux, and Capt. Richard Biles were directly active in the government phases of the entire project. R. Murphy of Baldwin and W. L. Lane and others of WADC operated airborne equipment during flight tests.

On the Phasing of Microphones*

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Summary—**Correct phasing of microphones is most important when two or more microphones are connected simultaneously to a single transmission system. The phasing of all gradient and of some phase-shift microphones is reversed for rewardly arriving sound waves. In this paper the phase-frequency characteristics of most common microphones are described; methods for predicting or ascertaining the phasing of microphones are given; and a system is proposed for experimentally determining the absolute phasing of an unknown microphone.**

INTRODUCTION

THE PHASING of microphones is not of the same manifest importance to the user as some of the other performance characteristics, such as frequency-response or sensitivity. Anyone who has dealt with sound systems knows that reversal of leads of a balanced input amplifier does not alter the performance of the system; moreover it is known that the character of common musical tones is not patently altered by the phase relationship of the individual harmonic components. Why then is phasing of any interest at all? There are two principal reasons: 1) when two or more microphones which are used in proximity of each other and feed into a common system are out-of-phase there is evidence of cancellation of output, especially at low frequency; and 2) the waveform of speech sounds tends to be dissymmetrical and the phasing of a microphone may have a bearing upon the degree of modulation of certain amplitude-modulated transmitters.

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To explain the existence of effect 1) refer to Fig. 1(a): two microphones M_1 and M_2 are connected through two amplifiers to a common line L. If a sound source S_1 is placed in the plane of symmetry between the two microphones, and assuming that the microphones are connected so that the electrical impulses add, then the response-frequency of the system will be the same as that of either microphone; this response is represented by the straight line S_1 in Fig. 1(b). If the sound source is placed in position S_2, the phase of the sound pressure at both microphones is not significantly different at low frequency because the wavelength of sound is long. At high frequency, however, the difference in phase becomes significant, and cancellations will be encountered whenever the differences of path between S_2 and the microphones is equal to odd multiples of $\frac{1}{2}$ the wavelength of sound. The response-frequency characteristic of the system under this condition is indicated in Fig. 1(b) by the dash-line labeled S_2.

Consider next the system in which the microphones are connected "out of phase." This condition is shown diagrammatically in Fig. 1(c). For a source placed in position S_1 in a plane symmetrical with respect to the microphones, the voltages generated by both microphones (if they are identical) will be equal and opposite resulting in complete cancellation at all frequencies. For a source placed in position S_2 there will be cancellation at low frequency; at high frequency, each time the difference in path length between S_2 and the microphones equals an odd multiple of half the wavelength of sound there will be an additive condition, resulting in a re-