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DEVELOPMENT PLAN FOR A VOICE COMMUNICATION SYSTEM SIMULATOR

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DEVELOPMENT PLAN FOR A VOICE COMMUNICATION SYSTEM SIMULATOR

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INTRODUCTION

During development and test phases of the voice communication systems for the Apollo Missions, communication engineers recognized that the design of voice communication equipment for optimum performance depends on many factors. These are (1) voice operated relay performance, (2) voice signal spectrum shape, (3) signal dynamic range, (4) automatic gain control, and (5) signal clipping levels. All of the factors contribute to the optimum performance of a given system in their own special way. For example, the voice operated relay can disturb the speech if the turn-on and cut-off times are not properly designed. If the turn-on time is too long, the first word of the voice transmission will be lost. Likewise, if the cut-off time is too short, the voice message becomes very choppy. Although simple in statement, these(on- off), times play an important role in determining the overall intelligibility of a voice system. Similarly, all of the other four parameters play important roles toward determining voice intelligibility. So that the proper design of all of these parameters can be implemented into the development of future voice communication equipment, the material to follow presents the concept, design, and specification of a voice system simulator. This simulator is to be a design tool which will provide a means for simultaneous or individual analysis of all of these parameters.

Preliminary work on this project was begun in January 1969. However, completion of the project is not expected until late in 1969, or the early part of 1970. At the completion date, the unit shall be ready for use.

1.0 CONCEPT

The concept of the Voice Comm System Simulator (VCSS) is that it is a universal audio test unit which will encompass variable voice processing techniques. Such techniques are: (1) Voice-operated relay, (2) voice spectrum shaping, (3) voice dynamic range, (4) automatic gain control, and (5) voice clipping. Voice clipping, as well as the other four techniques, will provide the variable capability of the test unit in that each technique can be independently adjusted. Being independently adjustable, these techniques can be made to simulate an Audio System of many different characteristics. For example, changing the spectrum of the voice, while holding all other parameters constant, significantly affects the output signal, thus providing a signal different to the one which existed before the spectrum change was made. The same is true when either one of the other four parameters is changed. It is most important that this kind of variable capability exists in the Voice Comm System Simulator (VCSS) because it provides the versatility needed to effectively simulate and study an extremely large class of audio system characteristics. Therefore, the concept of the audio system simulator is a universal voice system composed of a number of subsystems, representing independently variable system parameters used as design tools for the development of voice communication equipment.

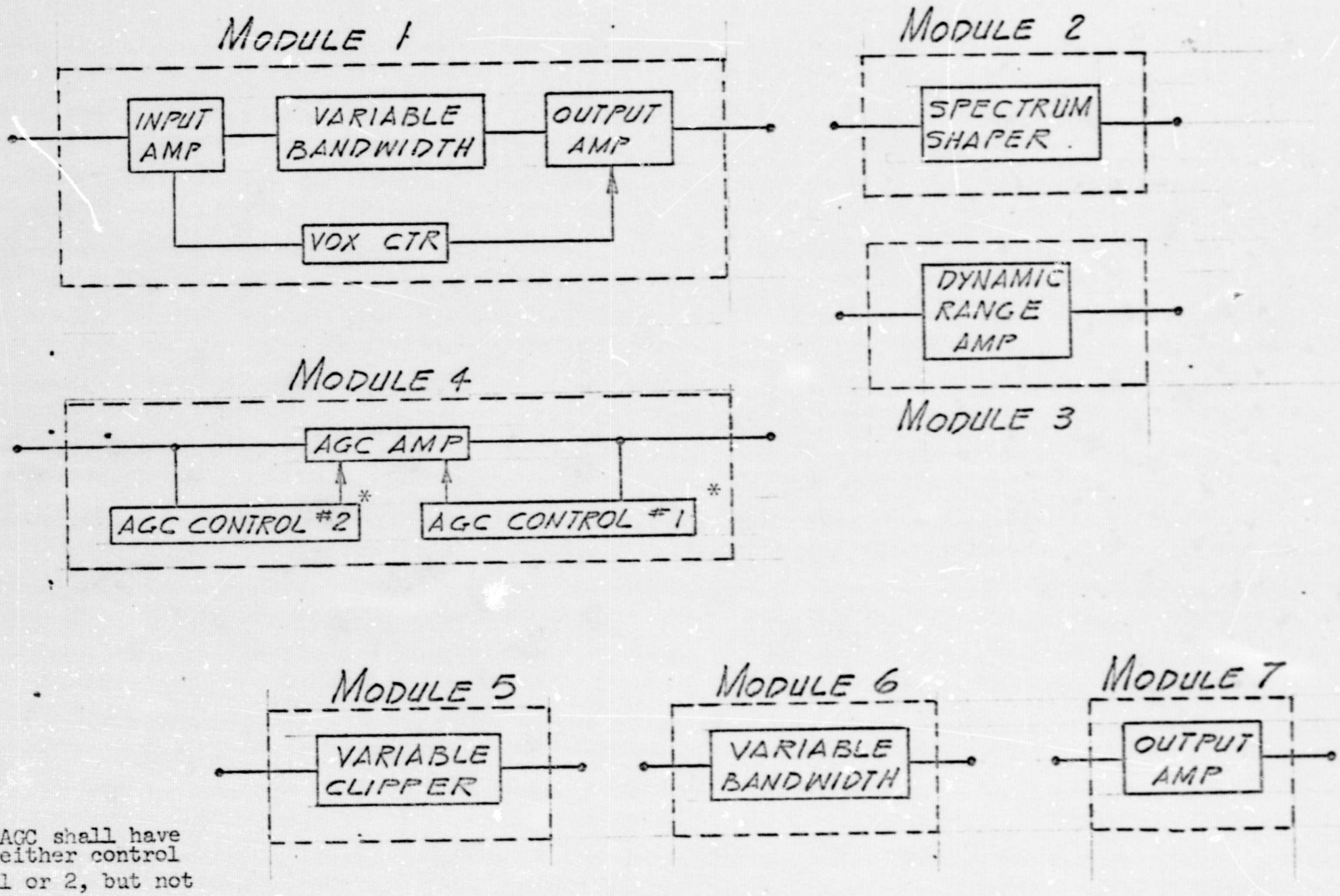
2.0 SYSTEM DESCRIPTION

The Voice Comm System Simulator (VCSS) is to consist of seven independent modules which can be arranged into any combination of 1 through 7. Six of these seven modules shall be different from each other and perform particular operation on a voice signal. Module one, for example, shall perform band-limiting and vox-operations; module two shall perform spectrum shaping; module three shall provide variable dynamic range; module four shall perform automatic gain control functions; module five shall perform voice clipping; module six shall perform output bandwidth limiting, and module seven shall perform the final output amplification.

Physically, the VCSS shall be contained within a three to four foot relay rack and be portable either by wheel or hand carrier. Each module shall be connected via a Trumpeter patch panel. This patch panel will allow either module to take the position of any one of the other module's position in the normal system sequence. The normal system sequence is shown in Figure 1 and a pictorial concept of the system is shown in Figure 2. The VCSS shall also provide external control for providing adjustments of parameters contained in each module.

3.0 SYSTEM DESIGN & SPECIFICATION

3.1 DESIGN



*AGC shall have either control 1 or 2, but not both.

FIG. 1 SYSTEM SEQUENCE

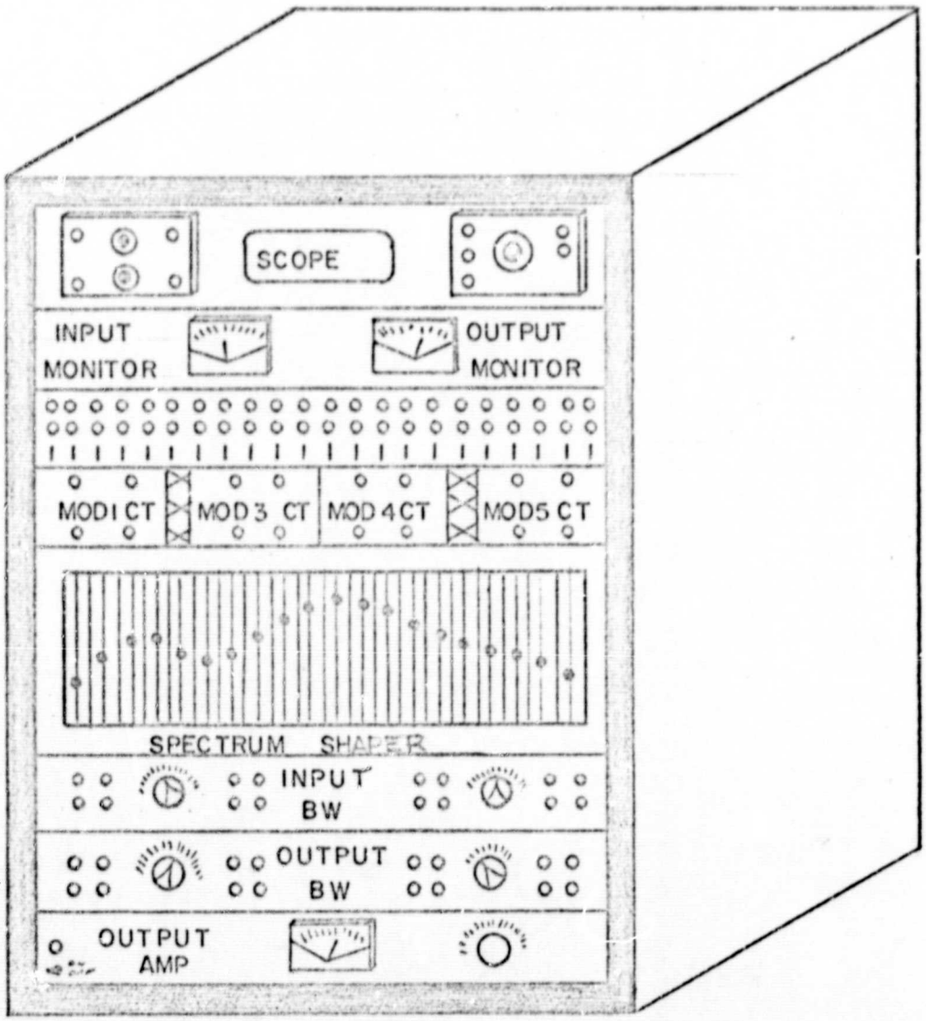


FIG. 2 PICTORIAL CONCEPT OF SYSTEM

The design of the VCSS shall use two approaches: they are (1) the utilization of off-the-shelf equipment and (2) in-house circuit development. Off-the-shelf equipment will be used to provide the band-limiting and spectrum-shaping functions which corresponds to modules 1, 6, and 2, respectively. All other modules shall be designed from in-house circuit development. Recommendations are provided in Section IV of this paper for circuit concepts.

Except in the off-the-shelf equipment, transformers will not be used. However, the input impedances for all modules shall be designed around 600 ohms and the output impedance is to be less than 60 ohms. More details of these requirements are found in the next section on specifications.

3.2 SPECIFICATION

The main specifications for the VCSS are given in Table 1, thus, this section provides only notes and comments to that set of specifications. Beginning with modules, the special notes and comments are as follows.

3.2.1 Module 1: VOX and Variable Bandwidth

3.2.1.1 Variable Bandwidth

The variable bandwidth shall be continuously variable from DC to 20 kHz, therefore, providing a signal with any bandwidth between DC and 20 kHz.

3.2.1.2 VOX: Voice Operated Relay

The voice operated relay shall have a variable threshold, a variable turn-on time, and a variable release time.

3.2.1.2.1 The threshold shall be adjustable over the entire input range, -78.2 to +2.2 dbm.

3.2.1.2.2 The turn-on time shall be adjustable from 0 to 10 ms.

3.2.1.2.3 The release time shall be adjustable from 100 ms to 2 sec. The normal time shall be 500 ms.

3.2.2 Module 2: Spectrum Shaper

This module shall have a resolution of one-third octave and the level in each one-third octave (represented by a 1/3 octave bandpass filter) shall be adjustable from 0 to 60 db.

3.2.3 Module 3: Dynamic Range (see also Sec. 3.2.8)

This module shall have a normal internal noise level, at the module input, of 5 uv. However, this noise level shall be adjustable by additive means, to a level of one volt rms.

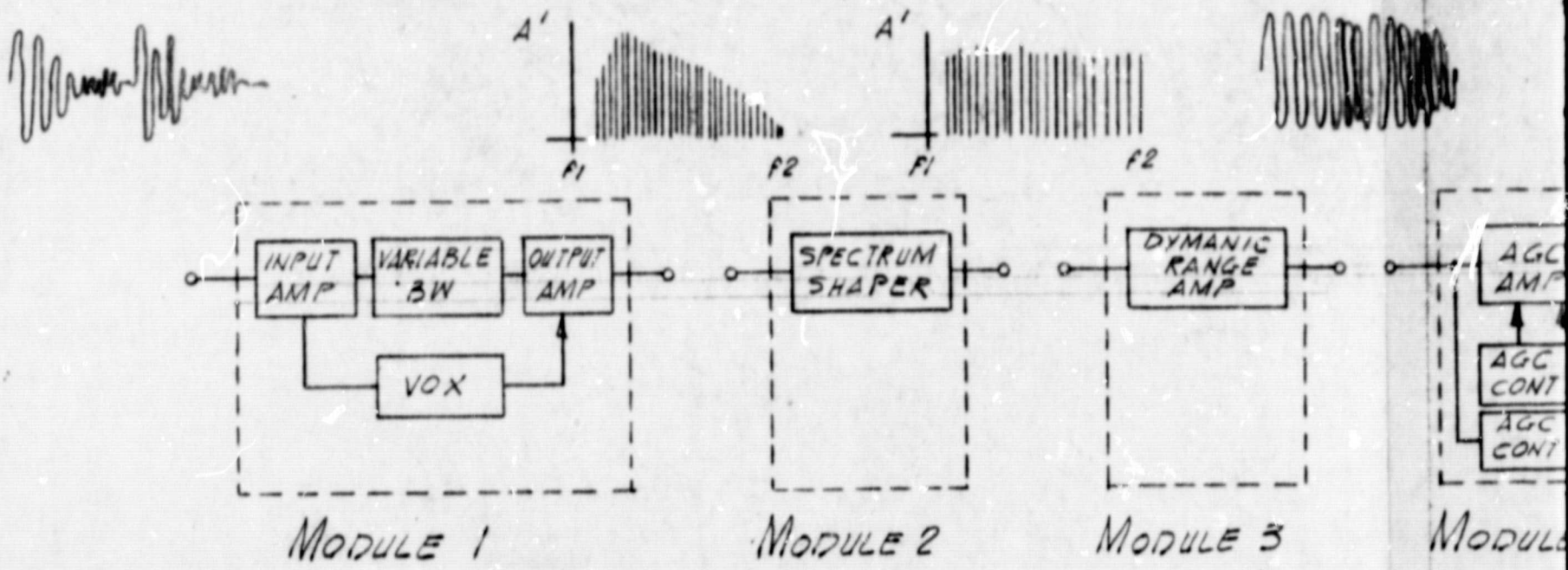
3.2.4 Module 4: Automatic Gain Control

This module shall have four variables. These are (1) attack time, (2) release time, (3) threshold, and (4) dynamic range.

3.2.4.1 The attack time shall have a normal value of 10 ms and be variable from 1 ms to 100 ms.

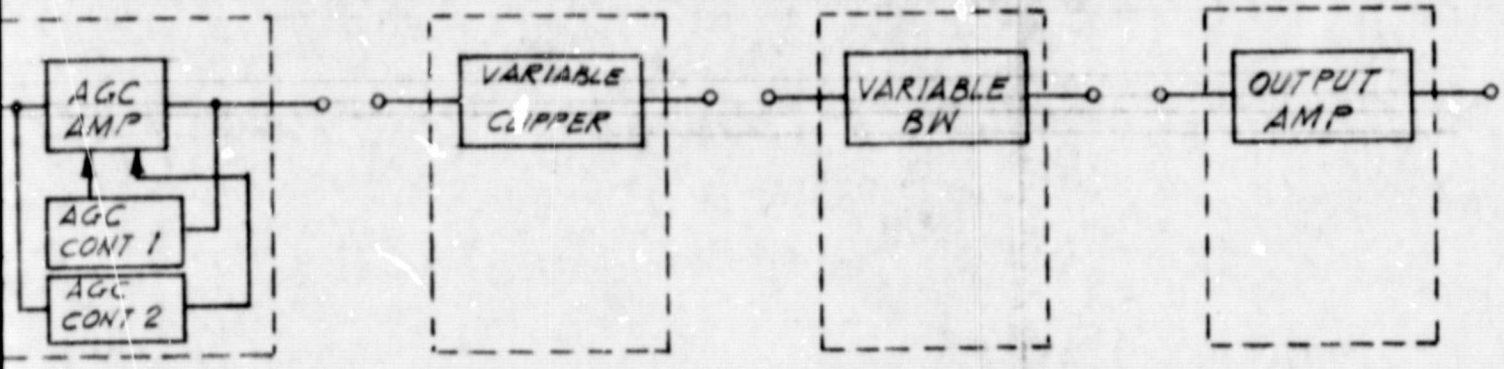
3.2.4.2 The release time shall have a normal value of 600 ms and be variable from 100 ms to 2000 ms.

3.2.4.3 The upper and lower threshold shall be separated by 30 db and a normal setting shall be 0 dbv to 30 dbv. However, the range



SPECIFICATION \ UNIT No.	MODULE 1	MODULE 2	MODULE 3
INPUT SIGNAL LEVEL	VARIABLE FROM -78.2 dbm TO 2.2 dbm	+ 2.2 dbm	+ 2.2 dbm
INPUT IMPEDANCE	600 Ω	600 Ω	600 Ω
BANDWIDTH	CONTINUOUSLY VAR OVER THE RANGE DC - 20 KHz	25 Hz TO 20 KHz	DC TO 20 KHz
GAIN	80 db	UNITY	UNITY
VOX TURN ON	DESIGNED TO OPERATE OVER THE COMPLETE RANGE -78.2 dbm + 2.2 dbm		
VOX ON TIME	500 MILLISEC		
ATTACK TIME AGC			
RELEASE TIME AGC			
OUTPUT LEVEL	+2.2 dbm	+ 2.2 dbm	+ 2.2 dbm
OUTPUT IMPEDANCE	< 60 OHMS	< 60 OHMS	< 60 OHMS

FOLDOUT FRAME 1



MODULE 4 MODULE 5 MODULE 6 MODULE 7

	MODULE 4	MODULE 5	MODULE 6	MODULE 7
	+ 2.2 dbm	+ 2.2 dbm	+ 2.2 dbm	+ 2.2 dbm
	600 Ω	600 Ω	600 Ω	600 Ω
Hz	DC TO 20 KHz	DC TO 20 KHz	CONTINUOUSLY VAR OVER THE RANGE DC - 20 KHz	DC TO 20 KHz
	42.2 db		UNITY	22 db
	10 MILLISECOND			
	600 MILLISECOND			
m	+2.2 dbm	+2.2 dbm	+2.2 dbm	VARIABLE FROM +2.2 dbm TO 24.2 dbm
HMS	< 60 OHMS	< 60 OHMS	< 60 OHMS	< 60 OHMS

FOLDOUT FRAME 2

UNIT No. / SPECIFICATION	MODULE 1	MODULE 2	MODULE 3	MOD
AMPLIFIER NOISE	5 μV MEASURED AT INPUT	< -50 dbv MOD OUTPUT	-160 dbv AT AMP OUTPUT	≤ 60d OUTPUT MA
AMPLIFIER SATURATION	LEVEL ≥ + 42.2 dbm	LEVEL > 38.2 dbm	LEVEL ≥ + 38.2 dbm	LEVEL
DYNAMIC RANGE	≥ 80 db	≥ 80 db	≥ 116 db	≥ 9
VOX THRESHOLD (VARIABLE)				
VOX OPERATING RANGE				
AGC THRESHOLD (VARIABLE)				
AGC OPERATING RANGE				
DYNAMIC RANGE VARIATIONS				
CLIPPING LEVEL VARIATIONS				
ROLL OFF CHARACTERISTIC	UPPER - 24 db/OCT LOWER - 24 db/OCT			
	FOLDOUT FRAME 1			

3	MODULE 4	MODULE 5	MODULE 6	MODULE 7
AMP	≤ 60dbV AT AMP OUTPUT FOR NO INPUT MAX GAIN	-80dbV AT MOD OUTPUT	-60db-m. AT AMP OUTPUT	-60db AT AMP OUTPUT WITH MAX GAIN
2db-m	LEVEL > 38.2db-m	VARIABLE FOR DIFF CLIPPING LEVEL	LEVEL > 38.2db-m	LEVEL > 32.2db-m
	≥ 90db	VARIABLE WITH CLIPPING LEVEL	≥ 90db	≥ 90db
		N.A.		
		0db TO INFINITE		
			UPPER - 24db/OCT. LOWER - 24db/OCT.	
			FOLDOUT FRAME 3	

from 0 db to 30 db shall be shiftable over the dynamic range. For example, if the lower limit of the threshold is -8 db. the upper limit shall be +22 db. See Figure 3.

3.2.4.4 This function shall be variable in that the minimum noise level, 5 microvolts, at the system input can be increased to any desired level up to 0.5 volts rms.

3.2.5.0 Module 5: Speech Clipper

This module shall have a speech clipper with a variable clipping level from 0 db to infinity. Where infinity is defined as the point where only the zero axis crossing of the signal remains.

3.2.6.0 Module 6: Variable Bandwidth

This module shall be the same as the bandwidth section of Module 1.

3.2.7.0 Module 7: Output Amplifier

This module shall have a variable gain from 0 db through +22 db.

3.2.8.0 In addition to the comments of section 3.2.3, the dynamic range module shall be designed such that the range can be symetrically variable. In other words, when varying the dynamic range, both the negative and positive amplitude halves of the signal shall be expanded or reduced simultaneously to/from saturation down to the noise level desired.

4.0 SYSTEM DEVELOPMENT PLAN

The development of the Voice System Simulator is to be conducted through two means. The first is the purchase of off-the-shelf equipment and the second is in-house development. Equipment purchases are to be initiated simultaneously with

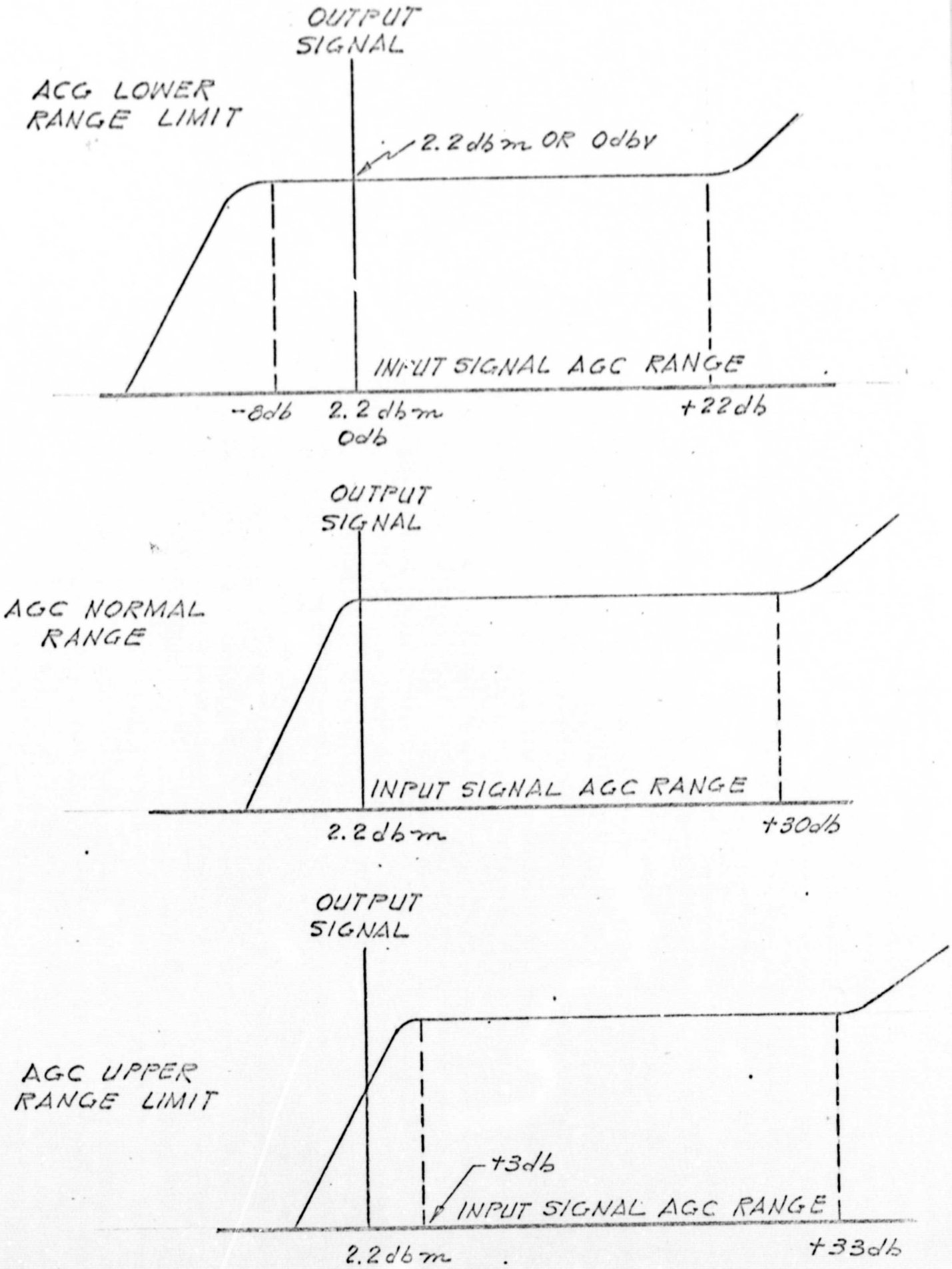


FIG. 3 AGC CHARACTERISTICS

the initiation of the in-house development. The in-house development will be broken down into the design and construction tasks. Each of these tasks will correspond to the design and implementation of particular modules. Since each module is to operate independently of the other, a parallel task effort will be performed. A special task will be implemented for the design and implementation of a common power supply to supply all modules except the off-the-shelf equipment. After all modular tasks have been completed, a system implementation and test task shall be performed to prove the functional operation of the system. To perform the system implementation task, test plans and procedures will be prepared. Following the system implementation task, a task shall be performed to prepare instruction and theory of operation manuals. Figure 4 provides a bar graph, outlining the System Development Plan discussed above.

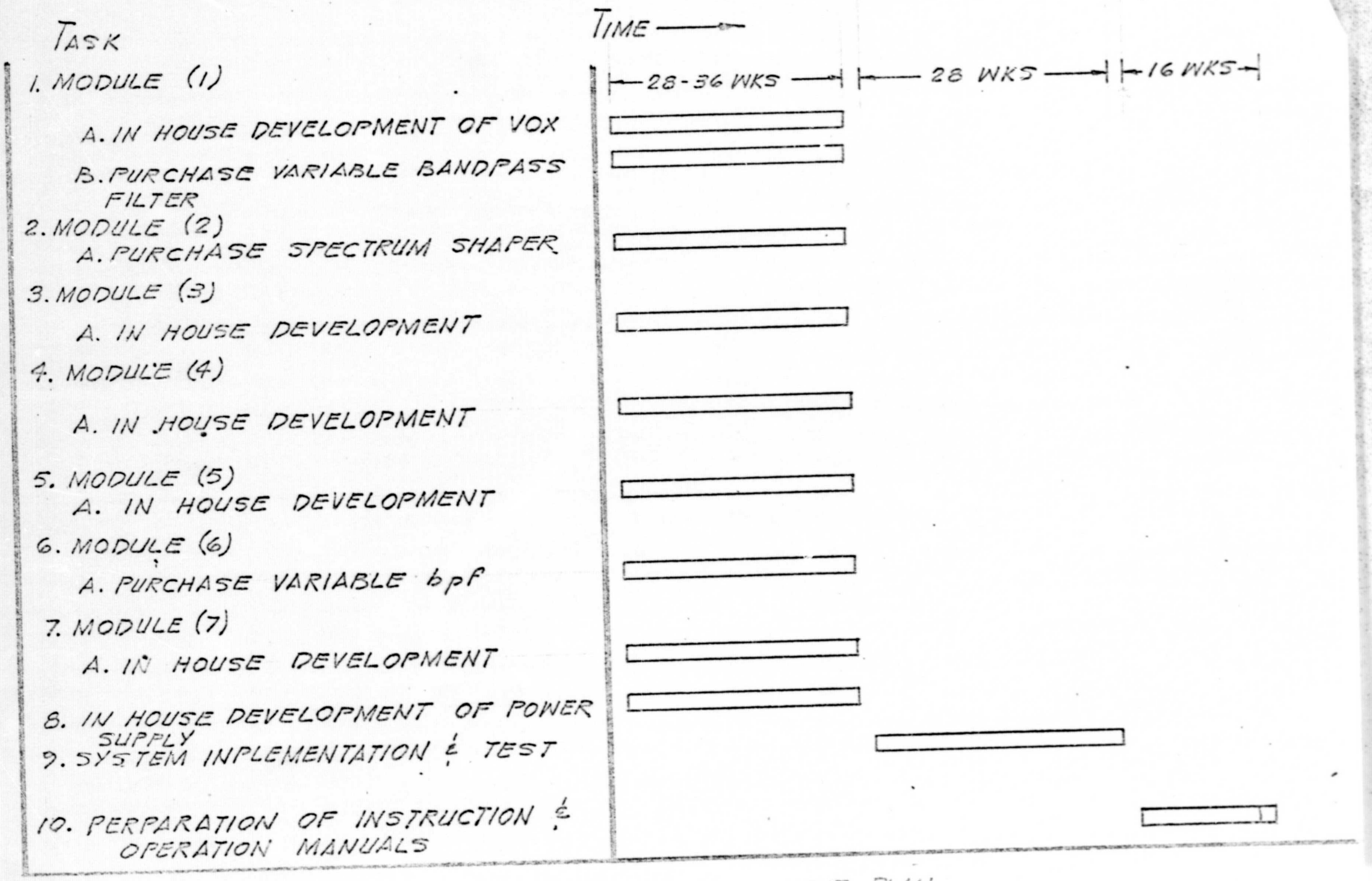


FIG. 4 SYSTEM DEVELOPMENT PLAN