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APOLLO EXPERIENCE REPORT - VOICE COMMUNICATIONS TECHNIQUES AND PERFORMANCE

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16. Abstract

The primary performance requirement of the spaceborne Apollo voice communications system is percent word intelligibility, which is defined in this report and is related to other link/channel parameters. The effect of percent word intelligibility on voice channel design and a description of the verification procedures are included. Development and testing performance problems and the techniques used to solve the problems also are discussed. Voice communications performance requirements should be comprehensive and verified easily; the total system must be considered in component design, and the necessity of voice processing and the associated effect on noise, distortion, and cross talk should be examined carefully.

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SUMMARY

The Apollo voice communications system is composed of many individual links that are interfaced to meet overall requirements. These links can be grouped into two basic types: ground to ground and spaceborne. The ground-to-ground links constitute a sizable part of the total system; however, because these links are basically commercial designs, they are not discussed in this report. Instead, the emphasis is on spaceborne links and the interfaces of these links with the ground. The system-performance requirement, design aspects, and performance testing are discussed. Of particular interest is the use of percent word intelligibility as the primary performance requirement, the precise definition of word intelligibility, the effect of the requirement on voice channel design, the relationship of word intelligibility to other link/channel parameters, and the verification that the requirement has been met. Many aspects of voice channel performance have been defined during the Apollo Program; however, a complete list of design criteria and an expeditious method of verifying the performance still are unavailable.

INTRODUCTION

The success of the Apollo Program required that a voice-communications capability be established across the 389 000 kilometers between the earth and the moon. The capability had to include not only direct communications but also a full conference that included the two crewmen on the lunar surface, the pilot in the orbiting command module, and the flight controllers in the Mission Control Center (MCC). These voice channels were implemented using links in the very-high-frequency (vhf) band (250 to 300 megahertz) and channels in the unified S-band (USB) system. The USB also provided for telemetry, command, ranging, and television functions.

Early in the Apollo Program, the performance requirement for voice communications was established as 90-percent word intelligibility for the prime links and 70 percent for the backup links. This performance requirement, combined with the range and functional requirements, formed the basis for the system design. The requirements were met, but not without many design, analysis, and testing iterations.

Results of the Apollo voice communications developmental program are discussed along with brief discussions of the voice channel requirements, the configurations, and the radio-frequency (rf) links concerned.

Unlike commercial ground-to-ground communications systems, the Apollo voice-communications links were constrained by the extreme range between transmitters and receivers and by the limitations on spacecraft transmitter power, antenna gains, and obtainable receiver noise figures. These constraints caused considerable difficulty in defining minimum acceptable signal-to-noise ratio (SNR) performance and in determining optimum modulation. Because voice is a complex signal, the evaluation of which is largely subjective, specific minimum design criteria do not exist, and link calculations based on a sinusoidal modulating signal cannot be related directly to word intelligibility. However, considerable work has been done on defining general relationships among word intelligibility and channel bandwidths, signal power, noise power, and so forth. On the basis of this previous work, the NASA and the prime contractors mutually agreed on a signal-to-noise standard for the Apollo voice channels of 14 decibels for 90-percent word intelligibility and 4 decibels for 70-percent word intelligibility when both the signal and the noise powers are expressed in root-mean-square (rms) values. This standard served the purpose for which it was intended; but, as was expected and later proved, specifying a single value of SNR for a specific percent word intelligibility is not valid. The fact also was known that preemphasis followed by peak clipping would increase the rms value of the modulating signal, thereby increasing the output rms SNR and the resulting intelligibility. Therefore, for the links that had insufficient rf margins, these voice-processing techniques were used to obtain the required performance. However, many pitfalls exist in the use of voice processing; these pitfalls must be understood fully if the required improvement is to be realized.

DISCUSSION

In this section, the performance requirement for the Apollo voice communications, the functional requirements, the design considerations, the systems testing, and the conclusions are presented.

Performance Requirement

The use of percent word intelligibility as the performance requirement was one of the more controversial aspects of the Apollo voice communications system. The meaning of the requirement in terms of end performance was not understood. Also, no standardized, rapid, and economical method existed for verifying that the performance had been met. No clear-cut guidelines existed on how a system should be designed to meet the requirement, and, likewise, no direct relationship existed with other link/channel parameters from which circuit margins could be calculated. Many questions remain; however, some have been answered and data pertaining to others have been derived.

The use of percent word intelligibility is an attempt to place a quantitative value on voice channel performance. Word intelligibility is defined as the percent of monosyllabic words correctly understood after transmission by means of the communications channel. The quality of the channel is not considered, and, under certain conditions, the channel can contain high noise levels, interfering tones, and crosstalk, yet still score high on a word-intelligibility test. Such interfering signals can be annoying, but, if the signals do not mask the speech, the effect on word intelligibility is small. In addition, people usually communicate using sentences or phrases, not monosyllabic words. Because of these factors, percent word intelligibility can be misleading, and, used alone, is an incomplete system performance criterion. However, the use of SNR measurements, spectral analyses, crosstalk measurements, and subjective listening, along with the word-intelligibility measurements, provided sufficient performance data to develop and verify the Apollo voice systems. Generally, the 90-percent intelligibility requirement was met or exceeded, and the significance of this figure and the limitations involved are now well understood.

The Apollo specifications state that the percent word intelligibility shall be based on the "American Standard Method for Measurement of Monosyllabic Word Intelligibility, Test Guide." However, this standard merely establishes general guidelines, not specific test standards. After considerable research on industrial testing methods and after several attempts to perform in-house word intelligibility tests, an agreement was established with the U.S. Army to score tapes at the Fort Huachuca Testing Center, Arizona. At Fort Huachuca, the standard phonetically balanced monosyllabic-word-evaluation technique, highly trained listeners, and an automatic grading system are used. This technique produces repeatable results; however, even with the automation, the procedure is time consuming. A minimum of 15 minutes of system time per data point is needed to produce the tapes, and as long as 4 weeks is required to score the tapes from a test series. A more rapid method of verifying voice channel performance is desirable, and efforts to develop such a method are recommended. A standard method, available to all organizations involved in the design and testing of voice channel components and systems, should be adopted early in any future programs.

The performance requirement has been discussed only in terms of verification capabilities. Of equal importance is the ability to relate the performance requirement to channel design criteria. At the beginning of the Apollo Program, no specific channel design criterion was known from which a channel could be designed that would provide 90-percent word intelligibility under all conditions. However, previous (usually empirical) efforts had produced general relationships among word intelligibility and bandwidths, distortion, signal power, noise power, and so forth. This previous work established a base for the determination of the required channel parameters. From these relationships, it was determined that the SNR had the predominant effect on intelligibility. Also, because SNR is a simple measurement and is included easily in link circuit margin calculations, a conversion between word intelligibility and signal-to-noise ratios was chosen. The conversion, as agreed upon by the NASA and the Apollo prime contractors, was 14 decibels rms/rms SNR for 90-percent word intelligibility and was 4 decibels for 70-percent word intelligibility. This relationship served the purpose of an established standard for use in design, analysis, and testing; however, the relationship was not valid for every channel (fig. 1) because only a mean channel bandwidth of 3 kilohertz and white noise were considered. The effects of specific bandwidths, non-Gaussian noise, distortion, spectral distributions, and so forth were not considered in this conversion. In a few instances, less than optimum design resulted (either underdesign or, equally as undesirable, overdesign), and additional work is needed to define minimum channel-design criteria for future programs. However, with this standard, the Apollo voice system design was simplified because optimum total-link

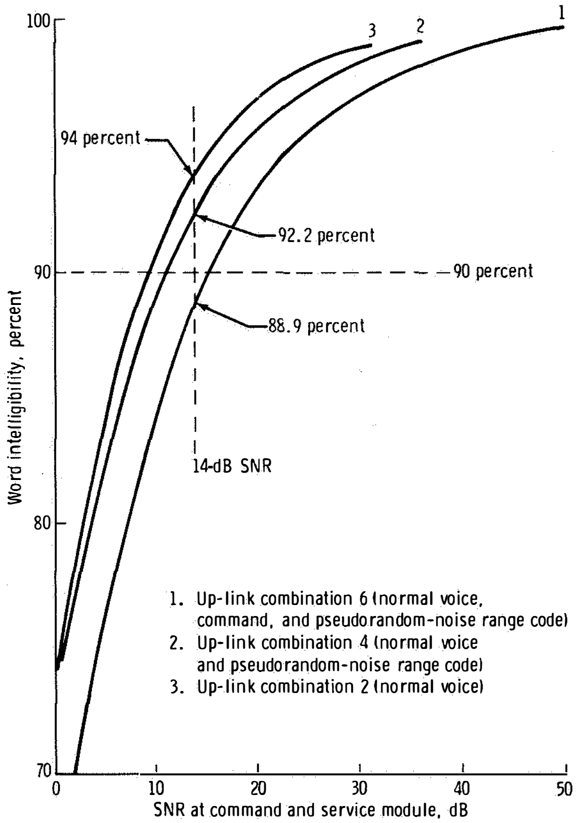


Figure 1. - Word intelligibility as a function of SNR for up-link combinations 2, 4, and 6.

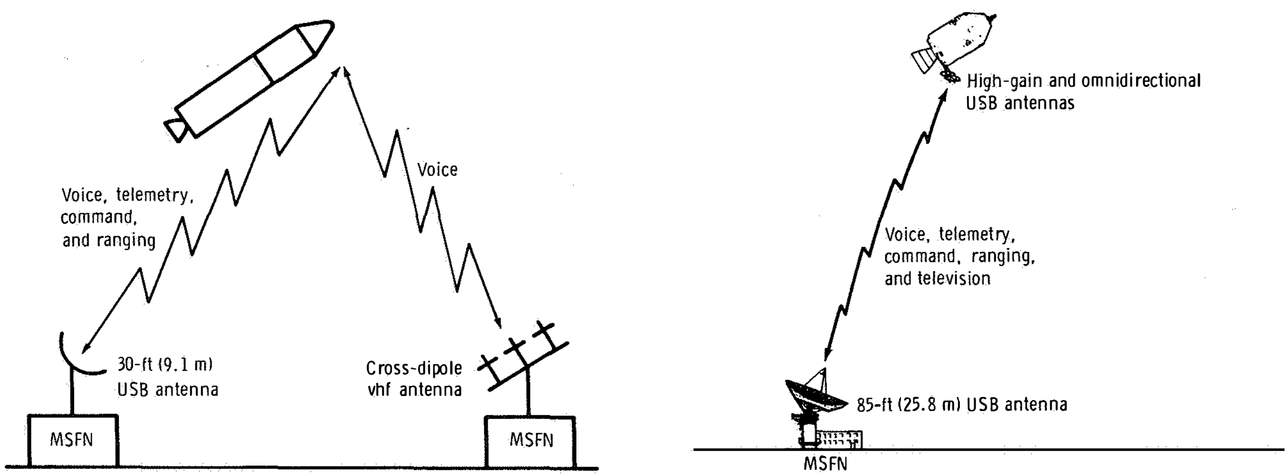
(voice, telemetry, command, ranging, and television) design was possible (by optimum selection of transmission powers, modulation, cable losses, receiver-noise figures, and so forth). Furthermore, preemphasis and peak clipping could be added when necessary to obtain the 14-decibel rms/rms or 4-decibel SNR requirements.

Functional Requirements

The basic functional requirements were established early in the Apollo Program. As the program evolved, so did the requirements. For the first lunar-landing mission, the voice requirements were as follows.

1. Launch through translunar injection: Command and service module (CSM) duplex voice with the Manned Space Flight Network (MSFN) using vhf/amplitude modulation (AM) or the USB system (fig. 2(a))

2. Translunar and transearth coasts: CSM duplex voice with the MSFN using the USB system (fig. 2(b))



(a) Launch through translunar injection. (b) Translunar and transearth coast.

Figure 2. - Functional voice requirements.

3. Lunar-orbital operations:

a. Command and service module and lunar module (LM) duplex voice with the MSFN using the USB systems (fig. 2(c))

b. Command and service module duplex or simplex voice with the LM using vhf/AM (fig. 2(c))

c. Command and service module/ MSFN/LM voice conference using the USB systems and MSFN relay (fig. 2(c))

4. Lunar-surface operations: Same as lunar-orbital operations with the addition of duplex voice conference among the two extravehicular astronauts (EVA-1 and EVA-2) on the surface of the moon, the MSFN, and the command module (CM) pilot using the following systems.

a. Very high frequency/frequency modulation (FM) from EVA-2 to EVA-1 with vhf/AM relay from EVA-1 to the LM (fig. 2(d))

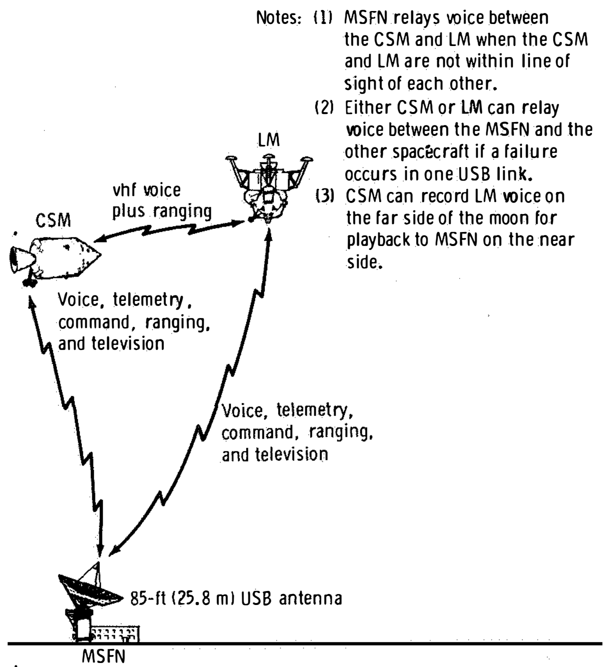
b. Very high frequency/AM from EVA-1 to both EVA-2 and the LM and vhf/AM from the LM to both EVA-1 and EVA-2 (fig. 2(d))

c. Unified S-band between the LM and the MSFN with LM relay between the MSFN and EVA-1 and EVA-2 (fig. 2(d))

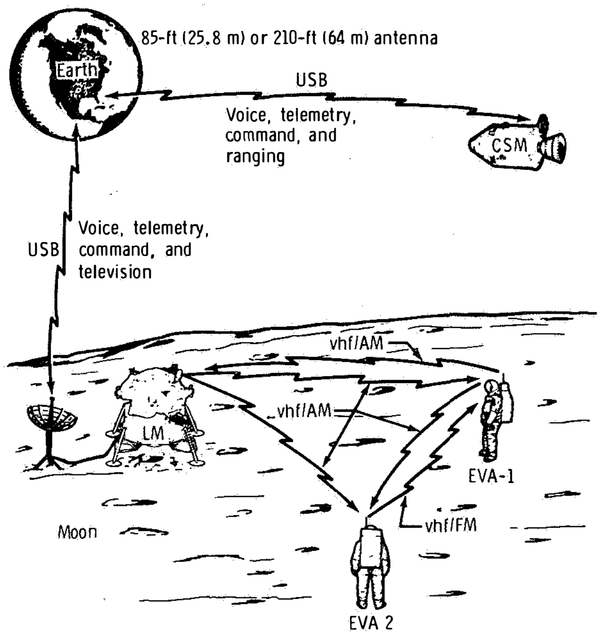
d. Unified S-band between the CSM and the MSFN with MSFN relay between the CSM and LM (fig. 2(d))

5. Recovery operations: CM simplex voice with recovery aircraft using vhf/AM (fig. 2(e))

In addition to these requirements, requirements existed for communications among the crewmen in each spacecraft; the CSM recording of CSM communications and the LM vhf voice with subsequent playback by way of the USB to the MSFN; the prelaunch voice with the launch complex; and,



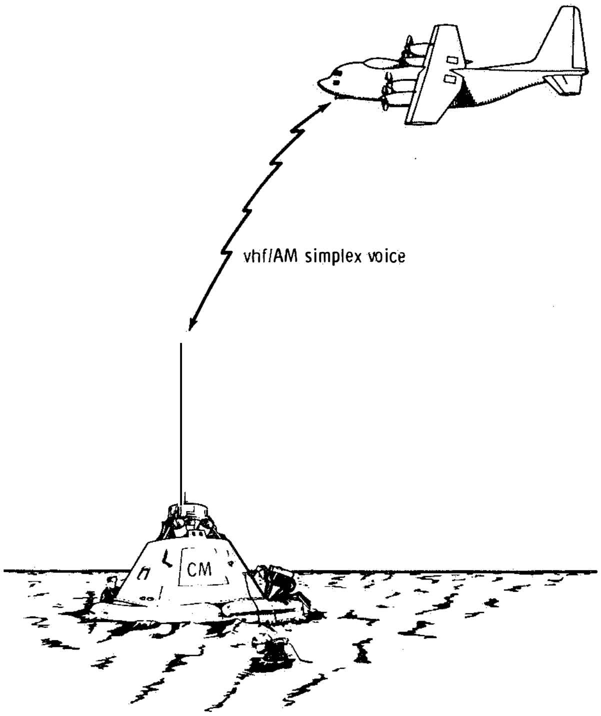
(c) Lunar-orbital operations.



Notes: (1) MSFN relays voice between CSM and LM.
(2) LM relays voice between MSFN and EVA-1 and EVA-2.
(3) EVA-1 relays voice from EVA-2 to LM.

(d) Lunar-surface operations.

Figure 2. - Continued.



(e) Recovery operations.

Figure 2. - Concluded.

for contingencies, the CSM relay of voice between the LM or EVA-1 and EVA-2 and the MSFN and LM relay of voice between the CSM and the MSFN. These requirements were met using different configurations of the CSM, LM, extravehicular communications system (EVCS), and the MSFN communications links. In addition to the voice requirements, these links had to accommodate requirements for telemetry, command, ranging, and television.

The CSM communications equipment consisted of three major groups: audio centers, USB, and vhf. The audio centers (one for each crewman) provided audio-channel selection, audio channel level control and isolation, voice operated relay (VOX), and push-to-talk (PTT) functions. The USB equipment (premodulation processor, transponders, power amplifiers, and so forth) provided modulation, transmission, reception, and demodulation of S-band signals between the CSM and the MSFN. The USB down-link voice was transmitted in three basic modes: the normal voice on the 1.25-megahertz FM subcarrier that phase modulated the carrier after being frequency multiplexed with a 1.024-megahertz pulse code modulation (PCM) subcarrier and the turned-around pseudorandom noise (PRN) range code (fig. 3); the backup voice, which was phase modulated directly onto the carrier after being multiplexed with the PCM subcarrier and PRN (fig. 4); and the dump voice, which was frequency modulated directly onto a different carrier after being combined with PCM and analog telemetry subcarriers. The USB down-link normal voice was preemphasized 6 dB/octave and was clipped 12 decibels before subcarrier modulation, and the backup voice was preemphasized 6 dB/octave and was clipped 24 decibels. The CSM USB also could relay received LM vhf voice or EVCS voice/data (fig. 5) to the MSFN by combining these signals with the CSM normal voice before subcarrier modulation. When relaying EVCS voice/data, the CSM voice was filtered 300 to 2300 hertz to prevent interference with the EVCS data subcarriers. The USB up-link voice was received by two basic modes — the normal voice by means of a phase-modulated carrier and a 30-kilohertz FM subcarrier and the backup voice by means of the 70-kilohertz FM command subcarrier. A squelch was provided for the normal up-link voice subcarrier. The CSM vhf equipment consisted of two transceivers on different frequencies and provided for transmissions and reception of vhf signals among the CSM and the LM, the EVCS, the MSFN, and the recovery aircraft. The vhf could operate simplex on either frequency or duplex using both frequencies. During lunar landing and ascent, a ranging tone (approximately 30 kilohertz) was transmitted along with the duplex voice between the CSM and the LM. The vhf transmitted voice was preemphasized 6 dB/octave and clipped approximately 24 decibels, and the resulting waveform keyed the transmitter on and off to produce maximum rf efficiency. Between words, a 30-kilohertz noise suppression oscillator or the

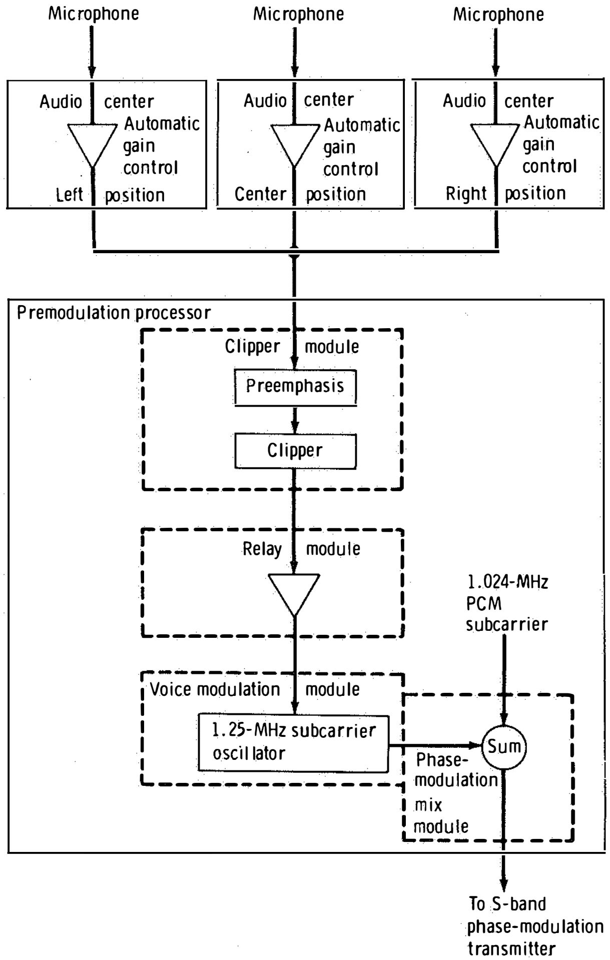


Figure 3. - Command module normal-voice signal flow.

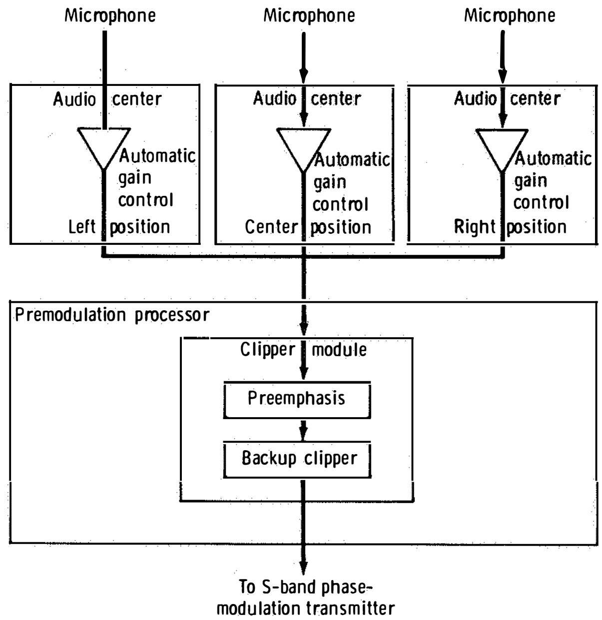


Figure 4. - Command module backup-voice signal flow.

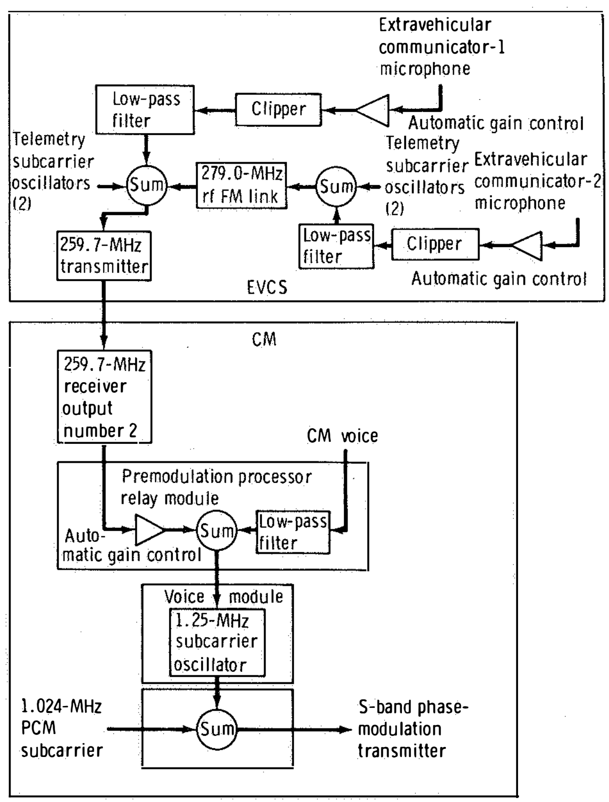


Figure 5. - Command module relay of EVCS signal flow.

ranging tone (depending on mode selection) modulated the vhf transmitter and provided an rf signal that quieted the receiver at the LM, the EVCS, or the MSFN.

The LM communications equipment was similar to that of the CSM. Functionally, this equipment was grouped as in the CSM: audio centers, USB, and vhf. However, in the LM, the audio centers and the premodulation processor were in one box referred to as the signal processor assembly. The major functional differences were in the USB transmission techniques. The LM down-link voice was transmitted in three basic modes: the normal voice (fig. 6), the baseband voice (fig. 7), and the backup voice (fig. 7). The normal

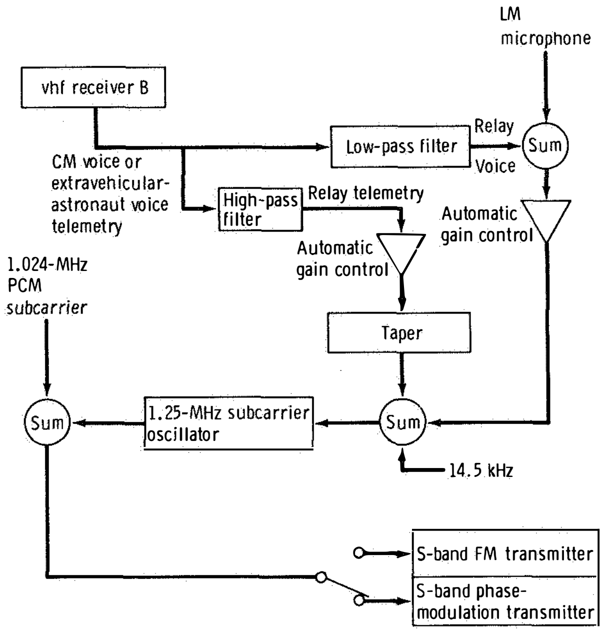


Figure 6. - Lunar module normal-voice signal flow.

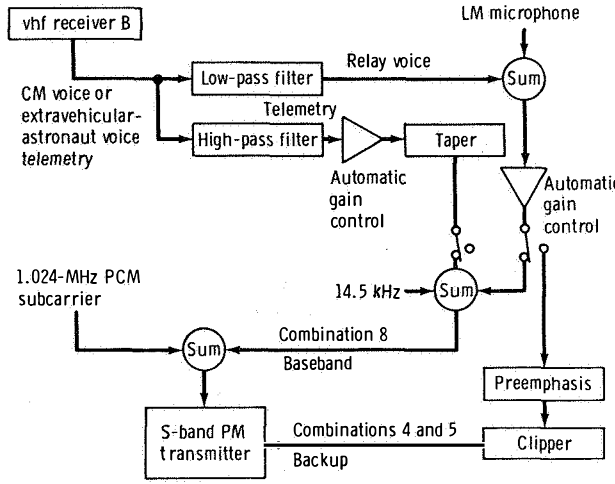


Figure 7. - Lunar module baseband- and backup-voice signal flow.

voice was frequency multiplexed with a 14. 5-kilohertz biomedical subcarrier; the composite signal frequency modulated a 1. 25-megahertz subcarrier. The 1. 25-megahertz subcarrier either phase modulated the carrier after being multiplexed with a 1. 024-megahertz PCM subcarrier and a PRN range code or frequency modulated the carrier after being multiplexed with the PCM subcarrier and the television. The baseband voice was frequency multiplexed with the biomedical and PCM subcarrier, and the composite signal phase modulated the carrier. The backup voice was similar to baseband voice, except that the biomedical subcarrier was not transmitted and additional voice processing was used. The backup voice was preemphasized 6 dB/octave, clipped 24 decibels, and transmitted in a "hot mike" mode. No processing was used for the normal and baseband voice other than automatic volume control (AVC), and the voice was transmitted in either a VOX or PTT mode. Also, in the normal and baseband modes, the LM could relay CSM voice or EVCS voice/data to the MSFN.

The EVCS provided communications between EVA-1 and EVA-2 and between them and the LM or CSM. The EVCS was a replacement for the original space-suit communicator, which could support only one extravehicular astronaut. The EVCS consisted of two extravehicular communicators (EVC-1 and EVC-2) and could operate in three modes: the dual mode, with data from each extravehicular astronaut to be relayed by the LM or CSM to the MSFN and with voice between EVA-1 and EVA-2 and between the extravehicular astronauts and the MSFN by way of the LM or CSM; the primary mode, with voice and data from one extravehicular astronaut; and the secondary mode, with voice only between one extravehicular astronaut and the LM or CSM. In the dual mode, EVC-1 relayed EVC-2 voice and data to the LM or CSM, thus permitting the use of only one vhf receiver on the spacecraft (fig. 5), which minimized LM/CSM changes when the EVCS was implemented. The EVCS transmitted voice was clipped approximately 12 decibels before modulation.

The MSFN equipment applicable to voice communications usually consisted of a dual USB system for simultaneous support of two spacecraft and vhf/AM transmitters and receivers for earth orbital mission phases. No up-link voice processing was used other than compression amplifiers, which were used to prevent overmodulation. Transmitter or channel keying was under positive control of the capsule communicator at the MCC, Houston, Texas, by the use of an in-band phase-shift-keying tone-burst technique. On the down links, a 2.5-kilohertz low-pass filter (LPF) was installed on all USB channels after it was determined that most of the Apollo down-link voice was less than 2.5 kilohertz and that the elimination of interference (noise and EVCS data subcarriers) above this frequency improved the quality without detectable word intelligibility degradation. Squelch was used on the vhf/AM receivers, but not on the USB, because of the requirement to use all received USB voice regardless of the degree of noise and because of the ability of the ground crew to disconnect the channel output from the lines upon complete loss of signal. However, under carefully controlled conditions, a voice-operated gain-adjusting amplifier (VOGAA) was used, with less than completely satisfactory results, to limit noise between spacecraft transmissions.

Channel Design Considerations

The significant design considerations were in the areas of voice processing (VOX, preemphasis, and clipping), filtering, and squelch. Generally, the premodulation voice processing in the spacecraft consisted of VOX or PTT circuits to eliminate background noise and to key certain transmitters; AVC amplifiers to ensure sufficient modulating levels and to prevent circuit overloads that could cause uncontrolled distortion; preemphasis networks and peak clippers; and, in some instances, low-pass filters to prevent interference with data channels. After this processing, the voice was applied to the modulators and transmitters. The modulation techniques used were FM/phase modulation (PM); FM/FM; and baseband FM, PM, and AM. For the CSM and LM vhf transmitters, the resulting clipped voice keyed the carrier on and off so that the output resembled a square wave. The amount of voice processing used, especially peak clipping, depended on the individual links and the need to maximize their performance. On certain links, no clipping was used; others used 12 to 24 decibels.

The purpose of the peak clipping was to reduce the peak-to-rms ratio of the voice signal and thereby increase the rms modulation level. The objective was to meet the 14-decibel rms/rms and 4-decibel SNR requirements for 90- and 70-percent word intelligibility at the maximum required ranges. Clipping does increase the intelligibility under marginal rf signal conditions. The improvement for the AM links is significant, but for FM links the improvement generally is much less because of the FM thresholding effects. Tests conducted in January 1966 on the Apollo USB up link (FM/PM) with preemphasis and 0-, 12-, and 18-decibel clipping indicated an improvement of 1 to 3 decibels at subcarrier threshold for 12-decibel clipping and no additional improvement with 18-decibel clipping. If only the change in peak/rms ratios is considered, the improvement should have been approximately 7 and 9 decibels, respectively. In AM links, the improvement also does not approach the expected value, presumably because of the added distortion products and increase in transmitted interword noise (caused by reducing the peak voice amplitude to a value closer to the peak background noise amplitudes). The original clipping levels for the CSM and LM vhf transmitters were greater than 40 decibels. With this heavy clipping and the on/off carrier modulation technique, cabin noise as much as 40 decibels less than the normal audio levels could modulate the carrier 100 percent, resulting in a 0-decibel signal-to-interword noise ratio. The primary solution to this problem was to reduce the clipping level to approximately 24 decibels. However, before this solution was found, considerable effort was expended on improving the noise-cancellation characteristics of the microphones and on exploring means of suppressing the cabin noise on the audio-center input lines. The noise-cancellation characteristics were improved by the incorporation of improved directional pickup, but attempts at developing a noise suppressor were abandoned. The suppressor that gave the most promise was a modified center clipper that consisted of a pair of back-to-back diodes shunted by a resistor. The performance of this device was satisfactory under normal conditions; however, the performance was highly sensitive to the input audio level and presented a reliability problem.

Another problem related to the on/off carrier modulation technique was high distortion (unintelligible speech) when the CSM was first tested with the MSFN vhf receivers before the Apollo 7 mission. This distortion was traced to a misalinement of the receiver and was corrected by improving the MSFN alinement procedures. Because of the square wave modulation, the required intermediate frequency bandwidth was almost identical to the actual bandwidths when the Doppler effect and the effects of transmitter and receiver stability were considered. Therefore, precise receiver alinement was necessary.

The following factors should be considered in future programs if clipping is considered.

1. Radio frequency link performance and the resulting need for clipping

2. Increase in rms modulation resulting from clipping

3. Background (cabin) noise

4. Distortion caused by clipping

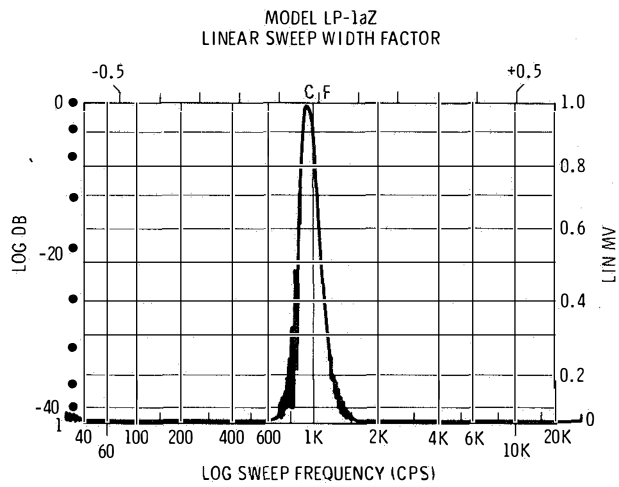
5. Type of modulation

6. Receiver bandwidths

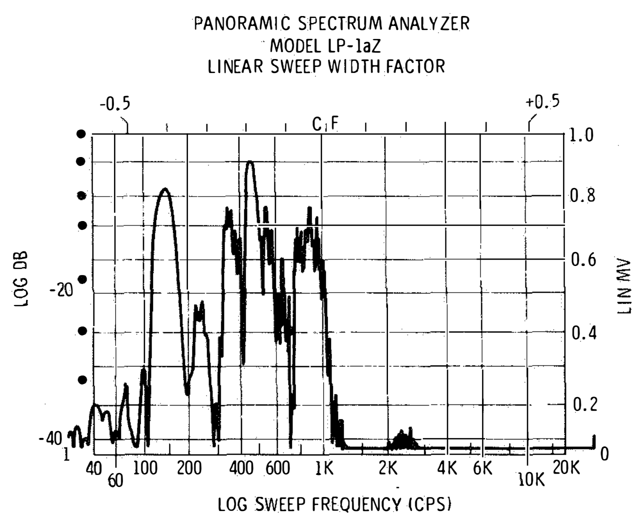
The AVC amplifiers that were used to compensate for speaking level variations and thus to maintain constant clipping/modulation levels also had some drawbacks. These amplifiers also could emphasize background noise and, when cascaded (as was often done in the Apollo systems for the relay and conference configurations), could produce a very high gain system that could raise normally negligible crosstalk to objectionable levels. (This was one of the factors that allowed the echo to be heard during the Apollo 11 lunar-surface operations.) For these reasons, the use of compression in place of AVC amplifiers in future programs is being considered at the NASA Manned Spacecraft Center (MSC). Also, in the early Apollo design, some problems were experienced because of the combined VOX and AVC attack times; with the VOX preceding the AVC, it took the combined attack times for the channel to stabilize. This problem was resolved by putting the AVC and VOX detector circuits in parallel. Other VOX related problems such as switching transients (banging) and first-syllable chopping were resolved by similar design refinements.

Low-pass filters were used in some premodulation voice-processing circuits to prevent interference in data channels (for example, EVCS 3.9-, 5.4-, 7.35-, and 10.5-kilohertz subcarriers) that are transmitted along with the voice. The filters that were used had cut-off frequencies as low as 2.3 kilohertz. The use of these low-pass filters did not degrade intelligibility significantly, because the speech from the Apollo microphones usually was limited (because of the spectrum of the male voice) to less than 2.5 kilohertz and most of the power was below 2.0 kilohertz (fig. 8). Although this speech normally would not interfere with the data channels, the distortion produced by the clipping could produce interference.

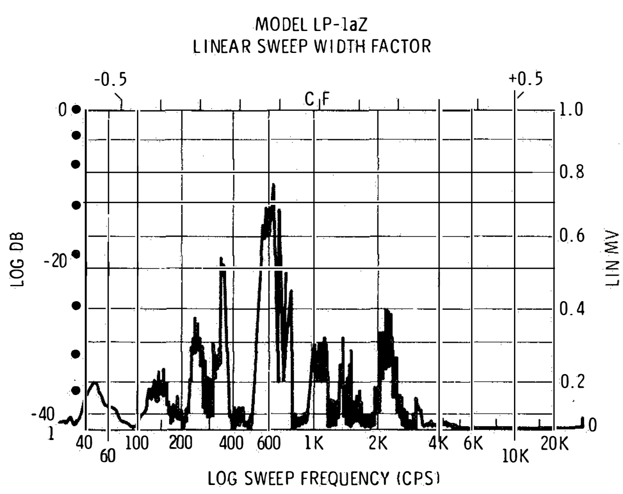
On the MSFN, low-pass filters were used on the USB down-link voice channels to prevent the data channels from interfering with the voice and to minimize out-of-band noise. The original MSFN signal data-demodulator system (SDDS) voice filters had an upper cut-off at approximately 3.5 kilohertz (noise bandwidth of 3.65 kilohertz). During tests in November 1967, the MSC Audio Techniques and Evaluation Laboratory (ATEL) of the Telemetry and Communications Systems Division (TCSD) determined that the SDDS filters permitted excessive data channel and system noise interference to the voice. Therefore, a subsequent effort was initiated in the ATEL to develop an optimum SDDS voice-channel LPF. The results of this effort produced the recommendation that a 0- to 2.5-kilohertz LPF with a rolloff of 36 dB/octave for frequencies above 2.5 kilohertz be installed at the MSFN



(a) Recorded 1-kHz sine wave.



(b) Subject saying "board."



(c) Subject saying "necessary."

Figure 8.- A spectrum display of Apollo microphone output.

sites. Extensive tests were indicative that the 2.5-kilohertz LPF improved the speech quality by eliminating the annoying noise and data channel tones above the speech spectrum without detectable effects in speaker recognition or intelligibility. Subsequently, this filter was installed at all MSFN sites. However, although the 2.5-kilohertz LPF was optimum for the Apollo down links, it may not be optimum for future programs. The sex of the speaker, microphone quality, premodulation processing, link noise characteristics, and adjacent channel interference all must be considered when selecting a postdetection filter.

In the original Apollo designs, voice channel squelch was used only on the vhf receivers. No squelch was implemented on the USB in either the spacecraft or the MSFN. However, in system tests it was demonstrated that the loss of the voice subcarrier or carrier on one channel could disrupt voice communications on other channels and be annoying because of high noise levels produced in the earphones. This problem was acute on the up-link relay modes, and squelch circuits were installed in both the CSM and the LM before the first lunar landing. Subcarrier detection was used instead of carrier detection because of the different subcarriers and demodulators used for normal and backup voice and because the voice subcarriers were not present for all up-link transmission combinations. Because of these factors, a carrier-operated squelch was unsatisfactory. Because of the up-link signal levels, the sensitivity (or operating point) of the up-link squelch circuits was not critical; usually, the up-link subcarrier SNR was sufficiently high for good discriminator operation, or the signal level was so low as to be almost nonexistent. Thus, the operating point was usually set 1 or 2 decibels below discriminator threshold even though word intelligibility was usually greater than 80 percent at this level. However, squelch "disable" was provided to extend communications capability to lower rf levels and to give the crewmen the option of operating with or without squelch. At times, certain spacecraft crewmen preferred to operate without squelch because the level of channel noise provided an indication of the channel availability.

Squelch on the USB down links constituted a problem that could not be solved entirely by either a carrier or subcarrier detector. The down-link signal levels usually were not as strong as the up-link levels, and a requirement existed to use all down-link voice regardless of the degree of noise. The ground-station operators could handle a complete loss of subcarrier or carrier by disconnecting the objectionable channel output from the lines. The real problem occurred during mission phases when both the CSM and LM USB links were active and the down-link audio signals were combined at an MSFN station for transmission through one landline to the MCC, and, in the MSFN relay mode, when the audio from each spacecraft was transmitted on the up links to the other spacecraft. During these mission phases, a noisy but usable down link could seriously degrade the otherwise good voice from the other spacecraft or, in the MSFN relay mode, could seriously degrade the MCC voice transmissions received in the other spacecraft. This problem could not be solved with a carrier or subcarrier squelch because the squelch would mute the output (noise plus usable voice) of the objectionable channel. In an attempt to solve the down-link squelch problem, voice-operated gain-adjusting amplifiers were installed at the MSFN sites. These devices were intended to pass the "voice plus noise" and mute the "noise only," theoretically, a good solution. However, the VOGAA had some bad characteristics: it dropped syllables and words and produced objectionable banging sounds. Because of these characteristics, tests were performed in the ATEL to determine the precise operating characteristics of the VOGAA. As a result of these tests, detailed procedures were developed and implemented by the MSC Flight Support and Flight Control Divisions for the use of these amplifiers. In the procedures, specific mission phases were defined for the use of the amplifiers, requirements to maintain correct input levels, requirements for the MSFN operators to monitor the input/output, and procedures for bypassing the amplifiers. A follow-on task also was undertaken to optimize the VOGAA speech detection circuits for Apollo speech. An analysis of the Apollo speech using power-spectral-density measurement equipment was indicative that the greatest concentration of power for each spoken word was in the frequency band of 400 to 800 hertz. Because this band of frequencies would have the highest speech-to-noise power ratio, it was decided to apply only this part of the speech spectrum to the VOGAA speech-detection circuits. A telemetry filter with a 540-hertz center frequency and an 80-hertz bandpass was chosen. The incorporation of this filter and the extension of the VOGAA "hold" time after speech was detected resulted in fewer speech dropouts and less impulse noise. However, any amount of voice dropout is unsatisfactory, and the development of an ideal syllabic detector, perhaps using digital techniques, is desirable.

The voice processing (AVC and clipping) and modulation indexes were optimized to obtain the required circuit margins for each down-link transmission combination. One of the adverse byproducts of this link optimization was changes in received rms audio levels at the MSFN when different down-link voice modes were selected. This phenomenon was first noticed in an Electronic Systems Test Laboratory (ESTL) system demonstration conducted in July 1967 and was subsequently evaluated and verified in MSC document EB68-3709, "Evaluation of Apollo USB Voice Downlink Audio Level Variation," dated December 1968. The variation was large (approximately 35 decibels); however, most of the variation (approximately 19 decibels) could be fixed by a one-time MSFN adjustment. The only satisfactory solution to the remaining variation was real-time level control by the ground station operators. In future programs, the effect on audio levels of audio processing and changes in modulation caused by mode changes should be a design consideration.

Audio Test Techniques

The major Apollo total systems compatibility testing was performed in the ESTL using spacecraft hardware (simulated hardware, developmental models, and flight models) and a configuration controlled MSFN USB station. For the voice channels, basically this testing consisted of establishing the up-link and down-link signal combinations, simulating the rf ranges by means of adjustment of rf path loss, measuring voice-channel SNR (predetection and postdetection), making word-intelligibility tapes for subsequent scoring, determining VOX/squelch operating characteristics, determining system crosstalk, and conducting functional tests to determine operational problems.

The ATEL was established at MSC in 1966. This laboratory provides a capability for detailed evaluation of audio characteristics and development of voice communications and test techniques. For the Apollo Program, the major ATEL functions were as follows.

1. Perform or coordinate intelligibility tests

2. Maintain an audio-tape library of master source tapes and test tapes

3. Perform detailed analyses of selected ESTL and flight audio tapes to resolve anomalies

4. Develop audio-test techniques and standards to be used in the ATEL and ESTL

5. Develop recommended system modifications to improve total system performance

Initially, word-intelligibility source tapes were prepared in the ATEL soundroom using standard auditory test word lists, and the resulting data tapes (made by playing the source tapes through the ESTL systems) were played back to subjects in the soundroom for intelligibility scoring. Because of the limited availability of test subjects, the results were not reliable. To improve the test results, arrangements were made to perform a word intelligibility test in the MCC using flight controllers as subjects. The results of this test were good; a sufficient number of trained and motivated subjects were available. However, the results of a subsequent test in the MCC were not as good. Because of inconsistent results, efforts were intensified to obtain an outside organization to score word-intelligibility tapes for the NASA. The result was that an interagency agreement was made with the U.S. Army to score the tapes at the Fort Huachuca Test Center. The source tapes for these tests were made using trained speakers and Apollo microphones in Apollo suits. These source tapes were played through the ESTL system according to a procedure developed for this purpose. The resulting data tapes were given a cursory examination in the ATEL and were forwarded to Fort Huachuca for scoring. The master source tapes and the returned data tapes were filed in the ATEL tape library. On specific problems, ATEL personnel performed a more detailed evaluation of the tapes. This more detailed evaluation included SNR measurements using test tones that were recorded on the tapes, speech-to-noise measurements using a technique developed by the TCSD, spectral analyses using cathode-ray tube (CRT) spectral analyzers or a power spectral-density X-Y plotter, and time-domain analyses using oscilloscopes and a high-speed beam oscillograph.

The speech-to-noise measurement technique was developed because the word intelligibility results did not agree closely with the SNR (using a sinusoid as the signal) measurements. It was believed that signal-to-noise ratios based on the actual speech would be more representative, and, therefore, more relatable to intelligibility. Unfortunately, the use of available meters to measure rms speech levels was too time consuming and inaccurate because the results depended on visual calibrations. Therefore, an effort was initiated to develop a speech-to-noise measurement technique that would produce reliable and repeatable results.

A digital computer was programed to detect speech in the presence of noise. Measurements of speech-plus-noise power and between-word-noise power were made and used to calculate the speech-to-noise ratio for 1 second intervals. The speech-to-noise ratios for the 1 second intervals and the long term speech-to-noise ratio were printed by the computer. Analog test equipment was used to develop and verify the speech detection and speech measurement techniques applied in the digital computer program. A strip-chart recorder and a printer were added to the analog system to allow simulation of the digital system for comparison purposes. Both systems provided reliable data. The analog system was retained in the ATEL. Subsequent use of this technique proved the invalidity of specifying that all links (or all link configurations) will provide a specific word intelligibility percent score for a given SNR. However, a reliable, repeatable method of determining actual rms speech-to-noise ratios does now exist (ref. 1).

A method also was developed to establish and measure voice modulation indexes accurately. Previously, such setups and measurements depended on visual estimates (from an oscilloscope) of the voice levels. Because speech is a dynamic signal, human judgment was required in making the determination. Also, the actual voice modulation levels of a system can be greatly different from the design levels if sine waves are used in the modulation-index (MI) setup. To overcome this problem, a static representation of the voice signal was produced by performing a spectral analysis of the voice output of the Apollo microphones and by simulating the spectrum with a noise generator and variable filters. The simulated spectrum can be recorded on tape and used as the system input in making MI adjustments or measurements. The need for this technique was demonstrated by the problems experienced with the LM backup down-link voice. On this channel, the modulation design and setup were based on sine waves; however, when voice was used, the actual peak modulation was much greater. This excessive voice modulation caused serious degradation in the PCM telemetry channel.

One of the major tasks in the postflight evaluation of Apollo voice performance was the determination of the actual spacecraft/MSFN configuration: backup voice as opposed to normal voice, CSM or LM or both remoted to MCC, up link to CSM or LM or both, LM or CSM relay of EVCS, and so forth. The MSFN recordings were the major source of data available for making this determination. The 14-track recorders provided a record of all site voice channels: CSM USB air-to-ground up and down link, LM USB air-to-ground up and down link, vhf air-to-ground up and down link, CSM USB normal-voice down link, LM USB normal-voice down link, CSM USB backup-voice down link, LM USB backup-voice down link, CSM USB verification receiver up link, LM USB verification receiver up link, and "Net 1" (up and down links remoted to MCC). Because of the amount of data and the complexity of possible configurations, the approach used in the ATEL to determine the actual configuration was to use a multichannel beam oscillograph to make a time-correlated record of the recorder tracks, enabling more accurate determination of the configuration with more economical use of manpower.

The use of power-spectral-density plots and CRT spectral analyzers was useful in identifying interfering tones (data subcarrier intermodulation products, power supply electromagnetic interference, and so forth) and in determining optimum low-pass filtering.

CONCLUDING REMARKS

Experience gained in the development of the Apollo voice-communications system has produced some conclusions, which are listed here as an aid to those involved in defining or developing voice systems for future programs.

The performance requirement should be comprehensive and should be stated in such a manner that it can be verified economically by means of standard techniques. Although percent word intelligibility defines a major aspect of channel performance, it is not concerned directly with quality, crosstalk, and certain types of interfering tones and noise. Also, the techniques used in measuring percent word intelligibility are numerous and usually expensive; and, if percent word intelligibility is specified, the precise method of measuring it also should be specified. Additional work in defining an optimum performance requirement specification, including applicable design criteria and related test specifications, would be beneficial.

The total system, talker to listener, should be considered in the design of any component of a voice communications system. The effects of background noise and cascading of automatic-volume-control amplifiers, clippers, filters, and voice-operated relays should be well understood. It is not valid to design two independent links to meet the link-performance requirement and then cascade them and expect that the system requirement also will be met.

The use of voice processing should be examined carefully as to the need for it and its effect on noise, distortion, and crosstalk. If needed because of radio-frequency margin limitations, approximately 12-decibel peak clipping is a suitable level; most of the system improvement occurs with the first 12 decibels, and 12 decibels generally are not enough to produce severe distortion or to emphasize cabin noise. Automatic-volume-control amplifiers should be avoided whenever possible because (depending on dynamic range) the amplifiers too can emphasize noise and raise negligible crosstalk to objectionable levels.

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National Aeronautics and Space Administration
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