



CCSDS

The Consultative Committee for Space Data Systems

Recommendation for Space Data System Standards

**VOICE AND AUDIO
COMMUNICATIONS**

RECOMMENDED STANDARD

CCSDS 766.2-B-1

BLUE BOOK

November 2017

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CCSDS Secretariat
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E-mail: secretariat@mailman.ccsds.org

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FOREWORD

This document is a technical Recommendation for use in determining which voice and audio industry standards should be used for interoperability between spacecraft, astronaut-to-spacecraft, spacecraft-to-ground, ground-to-spacecraft, and ground-to-ground applications. The voice and audio communications described herein are intended for missions that are cross-supported between Agencies of the CCSDS.

The international voice and audio industries have many standards and interfaces for acquiring, recording, and distributing live and recorded audio. That flexibility can lead to complexity when attempting to share or monitor voice or audio from acquisition to monitoring or recording locations. This Recommendation provides system designers with a subset of the larger industry set of standards from which to choose, depending on the application and purpose of the voice system.

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1 INTRODUCTION

1.1 PURPOSE

The purpose of this Voice and Audio Communications Recommended Standard is to specify the technologies, services, and service interfaces for real-time or near real-time voice and audio communications among terrestrial facilities and space systems in support of the mission operations of space flight.

This CCSDS Recommended Standard is intended for use by experts in voice and audio communications. A general background in the subject is provided in the corresponding Voice Communications Green Book (reference [E1]).

1.2 SCOPE

This Recommended Standard presents recommendations for voice and audio data communications in terms of:

- a) the exchange of voice and audio data between terrestrial facilities;
- b) the exchange of voice and audio data between terrestrial and space-borne facilities;
- c) emergency voice communications;
- d) rendezvous, proximity, and docking operations voice communications;
- e) search and rescue (SAR) voice communications.

Audio embedded in video feeds and special payload audio are considered a part of cases a) and b).

This Recommended Standard does not specify:

- individual implementations;
- the underlying mechanisms to perform the communication;
- the management activities required to configure and control the voice systems.

This Recommended Standard also does not address a range of practices that can be characterized as voice etiquette and protocol (roles and positions to talk) and are essential to the successful operation of any voice system, especially a system related to human space flight. Although these practices are out of scope in this Recommended Standard, implementers of voice systems are advised to recognize the need for user training and are encouraged to seek normative voice etiquette and protocol requirements in the mandatory training material for voice systems users in their space agency.

1.3 APPLICABILITY

This Recommended Standard applies to any terrestrial and space voice and audio communications systems claiming to be interoperable through adherence to CCSDS Recommended Standards and for use in mission operations.

This book is heavily oriented to the voice communications used for human space flights. Non-crewed space missions are less complex than crewed ones, and requirements for audio in non-crewed space missions can be seen as a subset of the audio requirements for human space flight.

Real-time or near-real-time voice communication is applicable for relatively short distances, spanning at most a planet and its moon(s). For longer distances, audio file exchange is generally recommended.

1.4 RATIONALE

The first goal of this Recommended Standard is to promote interoperability and cross support among cooperating space agencies and to reduce operation costs by allowing the sharing of facilities. Sharing facilities typically takes place when Mission Control Centers with several space agencies operating different voice systems need to communicate with one another in one facility, like the Moscow Mission Control Center (MCC-M) or DLR/ESA's Columbus Control Center (COL-CC). Interoperability is also important when a single Mission Control Center uses several voice systems to support different aspects of the same mission, as happens at NASA's Huntsville Operations Support Center (HOSC).

The second goal is to reduce the cost to the various agencies of performing common data functions by eliminating unjustified project-unique design and development. That is the case with the common development of the voice system used now in all the NASA and European Space centers related to the ISS program.

By having interoperable and compatible voice systems, intensive testing and voice communications issues can be avoided, thus reducing costs and improving reliability.

1.5 DOCUMENT STRUCTURE

This document consists of seven sections and five annexes:

- section 1 presents the purpose, scope, applicability, and rationale of this document and lists the definitions and references used throughout the document;
- section 2 provides an overview of voice and audio communications;
- section 3 provides requirements for recommended voice and audio data in terrestrial voice systems (ground communication systems);

- section 4 provides requirements for recommended voice and audio between spacecraft and ground systems;
- section 5 provides requirements for recommended voice in emergency voice communication;
- section 6 provides requirements for recommended voice communications in rendezvous, proximity, and docking operations;
- section 7 provides requirements for recommended voice and audio in SAR voice communications;
- annexes A–E include a Protocol Implementation Conformance Statement (PICS) proforma, security considerations, a list of acronyms and abbreviations, a discussion of DTN Bundle Protocol for voice and audio transmission, and a list of informative references.

1.6 DEFINITIONS

1.6.1 DEFINITIONS FROM REFERENCE [1]

This Recommended Standard makes use of the following terms defined in reference [1]:

A-Law: Companding logarithmic algorithm used throughout the world except North America and Japan. Encodes 13-bit linear Pulse Code Modulation (PCM) samples to logarithmic 8-bit samples. The G.711 encoder will create a 64 kb/s bitstream for a signal sampled at 8 KHz. The European A-Law system provides more quantization levels at lower signal levels.

μ -Law: Companding logarithmic algorithm used in North America and Japan. Encodes 14-bit linear PCM samples to logarithmic 8-bit samples. The G.711 encoder will create a 64 kb/s bitstream for a signal sampled at 8 KHz. The North American system tends to give more resolution to higher range signals.

1.6.2 TERMS

A list of abbreviations used within this document is provided in annex C. For the purposes of this document, the following definitions apply:

audio: Sonic information capable of being recorded or transmitted (typically 20 Hz–20 kHz). In addition to human voice, audio may contain experiments' noises, sounds, or music, all of which need more quality and higher data rates than voice to be transmitted.

audio files: Digital files that may contain recorded human voice or any kind of audio. Because they are files, they are transmitted like any other files, using file transfer standards.

bandwidth, BW: The rate of data transfer or throughput, measured in bits per second (bit/s) or multiple of bit/s (kb/s, Mb/s, Gb/s).

encoding: Analog-to-digital conversion or compression of digitally represented data.

NOTE – This definition of encoding is specific to audio data and differs from definitions of encoding found in other CCSDS documents, e.g., an algorithmic process that computes and adds check bits to a series of information bits to create a codeblock.

five by five: Of an audio signal, having sufficient strength and clarity to be subjectively considered to be of good quality.

mission control center, MCC: A facility that manages human space flights.

NOTE – This book uses MCC because it is the term used for human space flight. Mission Operations Center (MOC) is the term used primarily for satellite missions and is thus used in many CCSDS books.

Mission Operations Voice Enhancement, MOVE: A NASA project designed to replace existing voice systems utilizing a standardized commercial off-the-shelf approach.

plesiochronous digital hierarchy, PDH: Network technology for moving large amounts of data between systems that have nearly but not completely synchronized clocks.

packet switched network, PSN: A network technology where data is segmented into packets that are multiplexed onto links that may simultaneously support transmission of packets from many different sessions.

pseudowire, PW: A mechanism for emulating a point-to-point connection over a packet switched network.

quality of experience, QoE or QX: The subjective measure of a user experience with a service, in this case a space mission voice system.

quality of service, QoS: The ability of a communication system to provide predictable and differentiated services.

Quindar tones: The ‘beeps’ heard intermittently during space-to-ground voice communications in space missions. These were originally used by the Apollo missions to indicate the in-band signaling used to simulate Push-To-Talk (PTT) buttons.

summation: The mixing of multiple voice sources into a single conference loop.

voice: Audio produced by human vocal organs (typically 300–3000 Hz), capable of being transmitted using low-data-rate resources.

voice format: A collection of voice loops grouped together to meet the requirements of a particular situation or mode of operation. Voice formats are typically negotiated between two MCCs to prioritize loop connectivity to best match communication needs with the available channels.

voice loop; conference loop: The result of summation, a single conference containing the voice of all participating talkers.

NOTE – Because in each MCC there are many more internal voice loops than circuits available to transport the voice loops between the MCCs, different operations modes must be defined by different voice formats. Each format contains a limited number of voice loops matched to the available channels for a specific operational aspect between MCCs. These formats are interchangeable and dynamic; they need to be clearly defined between MCCs. Examples of voice formats are Joint Simulation and System Test Format. It is extremely important to reserve a number of channels in each format to maintain normal operations during a test or simulation.

1.6.3 NOMENCLATURE

1.6.3.1 Normative Text

The following conventions apply for the normative specifications in this document:

- a) the words ‘shall’ and ‘must’ imply a binding and verifiable specification;
- b) the word ‘should’ implies an optional, but desirable, specification;
- c) the word ‘may’ implies an optional specification;
- d) the words ‘is’, ‘are’, and ‘will’ imply statements of fact.

NOTE – These conventions do not imply constraints on diction in text that is clearly informative in nature.

1.6.3.2 Informative Text

In the normative sections of this document, informative text is set off from the normative specifications either in notes or under one of the following subsection headings:

- Overview;
- Background;
- Rationale;
- Discussion.

1.7 REFERENCES

The following publications contain provisions which, through reference in this text, constitute provisions of this document. At the time of publication, the editions indicated were valid. All publications are subject to revision, and users of this Recommended Standard are encouraged to investigate the possibility of applying the most recent editions of the publications indicated below. The CCSDS Secretariat maintains a register of currently valid CCSDS publications.

- [1] *Pulse Code Modulation (PCM) of Voice Frequencies*. ITU-T Recommendation G.711. Geneva: ITU, 1988.
- [2] *Lossless Compression of G.711 Pulse Code Modulation*. ITU-T Recommendation G.711.0. Geneva: ITU, 2009.
- [3] *Carrier-to-Customer Installation—DSI Metallic Interface*. ANSI T1.403-1989. New York: ANSI, 1989.
- [4] *Synchronous Frame Structures Used at 1544, 6312, 2048, 8448 and 44 736 kbit/s Hierarchical Levels*. ITU-T Recommendation G.704. Geneva: ITU, 1998.
- [5] *Characteristics of Primary PCM Multiplex Equipment Operating at 2048 kbit/s*. ITU-T Recommendation G.732. Geneva: ITU, 1988.
- [6] *CCSDS File Delivery Protocol (CFDP)*. Issue 4. Recommendation for Space Data System Standards (Blue Book), CCSDS 727.0-B-4. Washington, D.C.: CCSDS, January 2007.
- [7] *Licklider Transmission Protocol (LTP) for CCSDS*. Issue 1. Recommendation for Space Data System Standards (Blue Book), CCSDS 734.1-B-1. Washington, D.C.: CCSDS, May 2015.
- [8] *CCSDS Bundle Protocol Specification*. Issue 1. Recommendation for Space Data System Standards (Blue Book), CCSDS 734.2-B-1. Washington, D.C.: CCSDS, September 2015.
- [9] *Coding of Speech at 8 kbit/s Using Conjugate-Structure Algebraic-Code-Excited Linear Prediction (CS-ACELP)*. ITU-T Recommendation G.729. Geneva: ITU, 2012.
- [10] J. Postel. *User Datagram Protocol*. STD 6. Reston, Virginia: ISOC, August 1980.
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- [21] *Information Technology—Coding of Audio-Visual Objects—Part 3: Audio*. 4th ed. International Standard, ISO/IEC 14496-3:2009. Geneva: ISO, 2009.
- [22] *IP over CCSDS Space Links*. Issue 1. Recommendation for Space Data System Standards (Blue Book), CCSDS 702.1-B-1. Washington, D.C.: CCSDS, September 2012.
- [23] *TM Space Data Link Protocol*. Issue 2. Recommendation for Space Data System Standards (Blue Book), CCSDS 132.0-B-2. Washington, D.C.: CCSDS, September 2015.
- [24] *TC Space Data Link Protocol*. Issue 3. Recommendation for Space Data System Standards (Blue Book), CCSDS 232.0-B-3. Washington, D.C.: CCSDS, September 2015.
- [25] *Proximity-1 Space Link Protocol—Physical Layer*. Issue 4. Recommendation for Space Data System Standards (Blue Book), CCSDS 211.1-B-4. Washington, D.C.: CCSDS, December 2013.
- [26] *Proximity-1 Space Link Protocol—Data Link Layer*. Issue 5. Recommendation for Space Data System Standards (Blue Book), CCSDS 211.0-B-5. Washington, D.C.: CCSDS, December 2013.
- [27] *Proximity-1 Space Link Protocol—Coding and Synchronization Sublayer*. Issue 2. Recommendation for Space Data System Standards (Blue Book), CCSDS 211.2-B-2. Washington, D.C.: CCSDS, December 2013.

- [28] *Information Technology—Open Systems Interconnection—The Directory—Part 8: Public-Key and Attribute Certificate Frameworks*. 7th ed. International Standard, ISO/IEC 9594-8:2014. Geneva: ISO, 2014.
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2 OVERVIEW

2.1 INTRODUCTION

Voice and audio communications in space missions serve all the supporting personnel as well as the astronauts. The astronauts live and work in various space facilities developed and launched by different countries. The supporting personnel belong to different agencies and private companies as well. For mission safety and success, interoperability of voice communications is crucial.

Voice communication and audio data exchange are essential services for supporting space mission operations. They are especially important in a cooperative international environment in human space flight programs like the International Space Station (ISS). Between humans, one of the most common information exchanges is voice communication. In human space flight, the primary function of voice communication is to support the mission executed by astronauts, cosmonauts, taikonauts, and other space travelers. For purposes of this document, the following descriptions apply:

- *voice* is defined as the transport of human voice (typically 300–3000 Hz) using low-data-rate resources for transmission;
- *audio* (typically 20 Hz–20 kHz) may contain human voice, experiments' noises, sounds, or music, all of which need more quality and higher data rates than voice to be transmitted;
- *audio files* may contain either recorded human voice or any kind of audio. Because they are files, they are transmitted like any other files, using file transfer standards.

2.2 VOICE DATA EXCHANGE

2.2.1 GENERAL

Voice communications was once the realm of analog technology; now voice is encoded and exchanged in digital form. This Recommended Standard discusses exchange of voice data in five challenging domains, presented here as cases, that are applicable to internationally interoperable mission voice communications. The five cases defined in this book are the exchange of voice and audio data in the following situations:

- between different terrestrial facilities;
- between terrestrial and space-borne facilities;
- for emergency voice communications;
- for rendezvous, proximity, and docking operations;
- for SAR voice communications.

The exchange of voice data between ground-based personnel working in space missions can be achieved by commercial telecommunications services such as leased circuits like PDH (E1/T1) (references [3], [4], [5]) or Voice over Internet Protocol (VoIP). Both technologies can be combined inside and outside of an MCC, or they can be used for terrestrial or other kinds of communications as described in the five cases presented in this book. Carriers on the ground can use Multi-Protocol Label Switching (MPLS), Asynchronous Transfer Mode (ATM), or Integrated Services Digital Network (ISDN); however, current international communications are increasingly migrating to MPLS.

When VoIP is used, it is important that the originating voice codec be maintained from source to destination to prevent degradation resulting from transcoding. Examples would be to use IP/UDP/RTP/G.711 or IP/UDP/RTP/G.729.

If real-time voice communications are either impossible or not needed, it is preferable that audio files be exchanged using file transfer such as File Transfer Protocol (FTP), CCSDS File Delivery Protocol (reference [6]) over Delay-Tolerant Networking (DTN), or other approved standards.

2.2.2 CFDP

CFDP currently supports four classes of file transfer. There are proposed CFDP revisions that could lead to deprecation of Classes 3 and 4 in favor of using CFDP over DTN Bundle Protocol or LTP. Going forward, SIS recommends using one of the following two classes, each of which can be supported over the DTN Bundle Protocol (reference [8]) and DTN Licklider Transport Protocol (LTP) (reference [7]):

- Class 1—Unreliable CFDP Transfer over Bundle Protocol or directly over LTP-Red;
- Class 2—Reliable CFDP Transfer over Bundle Protocol or directly over LTP-Red or LTP-Green.

If Class-1 (Unreliable) CFDP is used over Bundle Protocol, then the expectation is that CFDP will invoke the Bundle Protocol reliable delivery option.

2.2.3 DTN

There are currently three mechanisms envisioned for voice and audio transmission using DTN:

- Mode 1 – Real Time Delivery, Best Effort;
- Mode 2 – Delivery of a sequence of bundles in transmission order and without duplicates or omissions;
- Mode 3 – Best-effort real-time delivery with (possibly delayed) complete stream reassembly.

Annex D presents a narrative of the classes of real-time voice transmission still under development by the DTN working group.

2.2.4 TRANSPORT TECHNOLOGIES

Digital voice is transported using packets (VoIP) or frames (E1/T1). For ground communications there are currently two major possibilities: using Time Division Multiplexing (TDM) with E1/T1 interfaces or VoIP (over E1/T1 or Internet). These technology options are described here:

- With TDM, each voice sample time slot in the TDM frame is called a ‘channel’. In European systems, TDM frames contain 30 digital voice channels (E1); in American/Japanese systems, the frames contain 24 channels (T1). Both standards (E1/T1) also contain extra bits (or bit time slots) for signaling and synchronization bits.
- TDM using PDH is still the more commonly used technology to transport voice with E1/T1 carriers (references [3], [4], [5]).
- TDM over Internet Protocol (TDMoIP) is the emulation of TDM over a PSN. TDMoIP is a type of pseudowire.
- VoIP is the emerging technology for new voice systems. VoIP can run over Internet or over dedicated networks (see next subsection).
- Space-to-Ground (S/G) voice communication primarily uses Radio Frequency (RF) links with different frequencies (S-band, Ku-band, and Ultra High Frequency [UHF]).

2.2.5 PROTOCOLS USED FOR VOICE TRANSMISSION

Depending on which kind of voice system will be implemented in the control center and the system’s functionality, one or a combination of the following protocols shown in table 2-1 will typically be used. These protocols are referenced from the normative sections of this document.

Table 2-1: Protocols Used for Voice Transmission

Protocol		Notes	Where Defined	Ref #
UDP (User Datagram Protocol)		None.	STD 6	[10]
RTP (Real-time Transport Protocol)		RTP is used for MCC and space communications.	RFC 3550 (STD 64)	[11]
RTCP (Real-time Control Protocol)		RTCP's primary function is to provide feedback on the QoS.	RFC 3550 (STD 64)	[11]
SRTP (Secure Real-time Transport Protocol)		SRTP has a sister protocol called Secure RTCP (SRTCP)	ECP 3711	[12]
TCP/IP (Transmission Control Protocol/Internet Protocol)		TCP/IP is appropriate for exchange of audio files. TCP/IP is NOT recommended for real-time voice communications.	STD 5, STD 7	[13], [14]
E1/T1		E1/T1 protocols support 30 (E1) or 24 (T1) digital channels simultaneously using TDM.	ANSI T1 ITU-T G.704/G.732	[3], [4], [5]
VoIP Signaling Protocols				
Session Control Protocols	Session Initiation Protocol (SIP)	SIP is designed to serve VoIP services over Internet Protocol (IP). SIP is an Application Layer protocol that can operate over UDP or TCP. SIP is currently (2016) at version 2.	RFC 3261	[15]
	H.323	H.323 is an ITU-T recommendation for call signaling and control, multimedia transport and control, and data-rate control for point-to-point and multi-point conferences for any packet network.	ITU-T H.323	[16]
Media Control Protocols	Media Gateway Control Protocol (MGCP)	MGCP is one implementation for controlling media gateways on IP networks and the Public Switched Telephone Network (PSTN).	RFC 3435	[17]
	Megaco (H.248)	H.248 follows the guidelines of the Application Programming Interface (API) MGCP architecture and requirements described in RFC 2805.	RFC 2805	[18]

2.2.6 CODECS

2.2.6.1 G.711

G.711 (reference [1]) is also known as Pulse Code Modulation (PCM). It is an ITU-T standard that can use lossless data compression to reduce data-rate requirements (reference [2]). G.711 maintains full audio quality at a relatively high data rate compared to other codecs. G.711 uses μ -law in North America and Japan, and A-law in other countries. It uses 64 kb/s bitrate (8 KHz sampling frequency \times 8 bits per sample).

NOTE – If there are no data-rate limitations, G.711 is the preferred codec for communications between MCCs or for S/G communications.

2.2.6.2 G.728

G.728 (reference [19]) is an ITU-T standard operating at 16 kb/s. It is a low-delay codec with linear prediction and low-data-rate requirements and is used commonly for communication between spacecraft, or S/G with low-data-rate (e.g., S-band) links.

2.2.6.3 G.729

G.729 (reference [9]) is an ITU-T standard providing an algorithm that compresses digital voice in packets of 10 ms duration. It is mostly used in VoIP and operates at a bit rate of 8 kb/s.

2.3 VOICE AND AUDIO COMMUNICATION REQUIREMENTS

Voice communication is the exchange of voice and audio data between different users. During a human space flight mission, a voice communication system should provide the following:

- point-to-point voice communication;
- voice loop or voice conference (these are implemented by voice summation, which is the mixing of multiple voice sources into a single conference [voice] loop);
- voice switch (or voice matrix);
- voice recording and playback of audio files in a standard format such as Advanced Audio Coding (AAC) (reference [20]; see also reference [E2]);
- voice/text conversion;
- keysets.

A keyset provides capabilities for selection of loops, as well as for audio input and output. Interaction with the keyset for loop selection is usually a physical interaction by pressing buttons. The interaction components of the device may be dedicated hardware buttons for each loop, or a software user interface shown as a touchscreen. Physical interaction is preferred for higher user awareness. For actual audio input, the physical interaction is done by actively pressing a PTT button to activate the microphone of the keyset for transmission of the voice signals to the selected loop. The keyset may be either an integrated or multipiece separated unit, or it can be a software application (soft keyset). The complete device is called a keyset and provides all necessary functionality for end-user interaction with the voice conferencing core.

2.4 VOICE AND AUDIO COMMUNICATION SERVICES

2.4.1 LIP SYNCHRONIZATION (LIP SYNC) FOR PUBLIC AFFAIRS

For public-relations events, an external facility communicates with the astronauts in space. The voice path normally uses S-band, and video uses Ku-band. The paths are different and have different latencies and need to be synchronized.

The final synchronization is done manually. The astronauts and the MCC test the whole path a few minutes before the event starts. The video personnel on the ground synchronize the video with the audio using the predefined values of the encoders. The personnel then either add or subtract steps of 10 milliseconds at a time to the encoder configuration until the lips are synchronized with the audio.

An automatic lip sync can be made possible if timestamps are used in the video and voice systems; however, this requires special equipment. Ideally, the synchronization of the video and audio should be done onboard and sent embedded from space.

Some of the new cameras are already using embedded audio (HD-SDI or SD-SDI) on the ISS.

2.4.2 VOICE RECORDING AND PLAYBACK

2.4.2.1 General

Voice recording and playback are an essential part of space missions. Classic uses of recording are to store voice data for post analysis during failure investigation, for reconstruction of issues, for public relations activities, and for training purposes. A common use is short-term playback of unclear communication to avoid misunderstandings and frequent repetition over the channels.

Based on the use cases, three different types of recording are used: user-specific recording, loop-specific recording, and interface-specific recording.

2.4.2.2 User-Specific Recording (Keyset Recording)

User-specific/keyset recording is the sum of all audio traffic for a specific user and is dependent on the user's loop selection. User-specific recording represents the presence of all audio signals at a user position.

Keyset recording and playback of a user position is used for a short time span only. In this way a user is able to replay the last minutes of all audio arriving at the keyset. It is mainly used for replays of unclear communication.

2.4.2.3 Loop-Specific Recording (Loop Recording)

Loop recording is defined as the sum of all audio traffic for a specific loop. It is user independent. Loop recording represents the presence of an audio signal within a loop.

Loop recording is the most common recording for space mission operations. All audio of a loop is stored separately within the recording system. This way, loop-specific investigations are possible (e.g., dialog between different positions).

Loop recording is very demanding in terms of storage capacity and correct time-tagging to allow replays of specific events based on a given time span.

2.4.2.4 Interface-Specific Recording (Interface Recording)

Interface recording represents the presence of an audio signal at an interface and is able to identify the direction of the signal (outgoing/incoming).

Interface recording is a technical support recording to verify audio sources within loops. It is aligned to loop assignments of channels.

Interface recording is mainly used to identify failure sources such as noise bursts within a loop, open microphone conditions, or other audible misalignments.

2.4.3 VOICE-TO-TEXT CONVERSION

Voice-to-text, also known as Automatic Speech Recognition (ASR) or text-to-speech (TTS), has become a standard application for many smartphones, browsers, cars, and dictation devices.

When long transmission time becomes an issue, voice-to-text software should be considered; voice should be converted to text and sent to the MCC as a text file. The reply could be sent in the same way and could optionally be converted back to voice.

NOTE – There are several voice-to-text and TTS software products available. Product selection is beyond the scope of this book.

2.5 VOICE AND AUDIO COMMUNICATION SECURITY

Security concerns in voice and audio communication are voice privacy, audio data integrity, and source authentication:

- *Voice privacy* means that the voice loop should not be heard by any other than the intended recipient(s). It can be achieved by employing protected data transport at lower layers, restricting distribution, or using cryptographic techniques.
- *Audio data integrity* means that the content of the audio/voice should not be altered between source and destination.
- *Source authentication* is the requirement that the audio should be attributable to a known origin. It can be achieved together with the privacy implementation.

A detailed description is found in annex B.

2.6 VOICE SYSTEM: PARAMETER CONSIDERATIONS

This document describes the setup of voice communication in a space mission context, points out special requirements and operational approaches, and defines the transmission, coding, interface, and quality parameters needed for space mission support.

Different parameters are required for each of the five voice communication cases defined in this book. Each of the five cases presented in this document identifies the parameters applicable to the case being presented. In some cases only some parameters are applicable. Best-practice recommendations are presented in normative text.

The following parameters need to be taken into account in the definition and implementation of a voice system:

- packet-switched versus circuit-switched systems (packet-switched systems are steadily replacing circuit-switched systems);
- users and permissions associated with the login (e.g., the roles Talk/Listen/Monitor [T/L/M]);

NOTE – Monitoring is used in many console positions. Especially in the external ones (e.g., payload operator) just to listen in a voice loop but without permission to talk. For the positions that have talk/listen/monitoring permissions, the monitoring button is often used to not disturb the voice loop when talking in another loop simultaneously or when talking in a side conversation.

- voice formats and activation procedures, a common case when more voice loops are needed than available circuits to transmit them;

NOTE – Voice formats need to be clearly defined and agreed to between the MCCs before use. Changing voice loops within a format is a dynamic task; good coordination is crucial. Voice Formats are normally specified in interagency Interface Control Documents (ICDs).

- PTT;
- codec selection and settings (including lossless, lossy, and eventually effects of packet drops);
- minimization of codec conversions;
- synchronization;
- latency requirements (e.g., 10–200 ms for VoIP systems inside of one control room);

NOTE – The latency requirements for a system running inside a control room need to be as low as possible. For systems communicating with external locations or different rooms inside of a facility, it is not a problem to have a higher latency.

- frequencies and channelization;
- signaling (Channel Associated Signaling [CAS] or Common Channel Signaling [CCS]);
- voice encryption and voice private-channel setup;
- quality monitoring (depending on the case, QoS and/or QoE will be needed; if the Internet is used for the voice transmission, quality cannot be guaranteed—see references [E7] and [E8]);

NOTE – Quality of service is related to the network. Quality of Experience is related to the human perception; should be five by five.

- alternative methods (secondary paths, such as IP telephony, e.g., Skype, VSee in the ISS for voice conference, which use different protocols and etiquette).

3 CASE 1: TERRESTRIAL (GROUND) COMMUNICATION SYSTEMS

3.1 OVERVIEW

3.1.1 BACKGROUND

Terrestrial communication systems can be divided into three cases:

- within an MCC;
- between MCCs;
- between an MCC and other external entities.

Correct functioning of the end-to-end voice system requires some support from the environment, particularly to assure availability and reliability of the networks, but also to ensure that the people operating the system do so according to established rules. That means console operators must to be certified in the use of the voice system, as with the etiquette and protocol as well.

3.1.2 TERRESTRIAL COMMUNICATION SYSTEMS WITHIN AN MCC

The voice system in an MCC typically uses a dedicated system, with dedicated network lines, and high availability. This assures a high voice quality, very high reliability of the communications, and good overview and control over the system functionality. All of these features make the MCC voice system a unique application.

MCCs employ centralized, high-performance, fault-tolerant, high-capacity voice conference systems to provide voice communications for flight operations and support personnel. The required level of performance is typically achieved through dedicated TDM-based systems or loop-based systems connected to keysets located in flight-control and support rooms. E1/T1 provides the necessary interface to other MCCs for voice communications.

The following context diagram (figure 3-1) illustrates E1/T1, Private Automated Branch Exchange (PABX), VoIP, and analog connections between different MCCs. Keysets are connected internally via T1 or VoIP interfaces.

As mentioned in 2.2.4, current technology is moving toward VoIP. In many systems there is a combination of VoIP (for internal distribution in an MCC) and E1/T1 interfaces for external or internal distribution, or a combination of VoIP for external keysets as well.

The advantages of VoIP are the low infrastructure and network costs. The disadvantage is the quality of service.

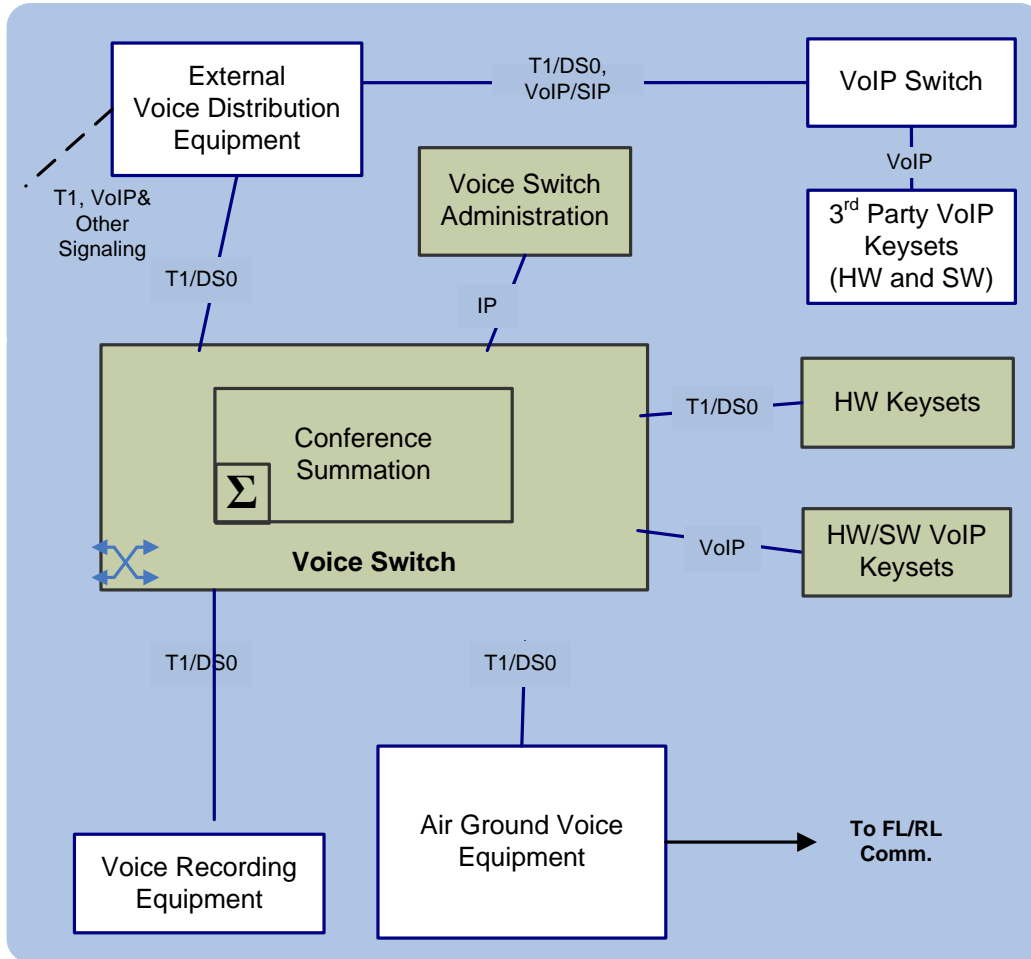


Figure 3-1: Example of Voice Connections inside of an MCC

Figure 3-1 shows a voice system inside an MCC having a PDH switch and a VoIP one, with the connections to the external world.

3.2 TERRESTRIAL VOICE SYSTEMS WITHIN AN MCC: RECOMMENDED STANDARDS

NOTE – There are MCCs with different voice systems inside of a facility that support the same mission. These different voice systems sometimes belong to one space agency (e.g., HOSC) or to different space agencies (e.g., MCC-M, COL-CC) and need to communicate with each other faultlessly. All of the requirements are testable, some of them in the ‘traditional CCSDS way’, others as is common in the voice communications world, but all are applicable for an MCC.

3.2.1 The voice system shall use a reliable network (at least 99 percent availability).

3.2.2 It shall be possible to define roles (positions) with the related permissions (T/L/M).

3.2.3 It shall be possible to define access permissions to the users to the associated roles (T/L/M) for all voice loops. X.500 (reference [28]) (e.g., Lightweight Directory Access Protocol [LDAP]) authentication should be used.

NOTE – Maintaining appropriate access controls can be quite labor intensive since the user information and role assignment can be very dynamic.

3.2.4 It shall be possible to define and store voice formats.

3.2.5 Voice formats should be stored with unique human readable names.

3.2.6 Voice formats shall be defined according to the specifications in the interagency ICD.

3.2.7 Changes in the voice formats shall be specified in an Engineering Change Request (ECR).

3.2.8 It shall be possible to dynamically exchange the voice formats with an interruption of less than 3 seconds.

3.2.9 It shall be possible to edit the voice formats (offline) and apply them as required in 3.2.4 and 3.2.6.

3.2.10 PTT shall be implemented in all kinds of keysets (manual or automatic).

NOTE – PTT can be implemented physically via a button or pedal, or via Voice Operated eXchange (VOX—aka voice operated switch, voice level detection, or voice activity level), where the voice level is monitored and the PTT is activated automatically.

3.2.11 VOX may be implemented on keysets.

3.2.12 If VOX is implemented:

- a) The VOX threshold must be adjustable between -26 dBFS and -94 dBFS.
- b) The VOX system should also have a hang system (VOX timeout) from 1 to 3 seconds to remain engaged during brief speech pauses.
- c) It must be possible to save and restore the VOX threshold settings for each individual user in a console position.

3.2.13 G.711 (reference [1]) should be used for communications in a MCC.

3.2.14 There shall be no more than 10 ms latency for communications inside of a control room and no more than 200 ms for connections to other rooms or facilities.

3.2.15 It shall be possible to apply signaling to different loops. The signaling should be compatible to the specifications in reference [3]. CAS and CCS shall be selectable.

NOTE – In analog systems signaling is always present. In digital systems an open line can be used without signaling, but for important loops like S/G, signaling is required. This includes CAS signaling (often called ‘robbed bit’ in the Americas) on the T1s and CCS signaling for the PABX.

3.2.16 It shall be possible to have channelization (association of voice loops to physical channels).

3.2.17 It should be possible to adjust the output delay for a given channel in 1 ms increments.

NOTE – This functionality could be external to the voice system.

3.2.18 Voice recording shall be possible for all loops in an MCC.

NOTE – Voice recording is extremely important in human space missions.

3.2.19 High-performance voice conferencing systems using TDM with G.711 μ -law or A-law (reference [1]) voice encoding shall be used in an MCC.

3.2.20 All systems designed after 2015 shall use VoIP.

3.2.20.1 VoIP-based systems must be compliant with SIP (RFC 3261—reference [15]) and RTP (RFC 3550—reference [11]).

3.2.20.2 Encryption should be implemented if VoIP is used, e.g., IPsec encryption using AES128 or a stronger algorithm.

3.2.21 QoS and/or QoE shall be monitored if possible.

NOTE – If Internet is used (VoIP) for the voice transmission, quality cannot be guaranteed. In either case, a voice recorder is needed as described in 2.4.2.

3.2.22 SIP-based IP telephony (e.g., Cisco) is used by most of the space agencies and should be used for IP telephony.

3.2.23 For secure voice conferencing where video is required (e.g., VSee) screen-sharing software shall be used for secure video chat. The system used must be previously agreed.

3.2.24 Recorded audio shall be stored as files (see 4.2.3).

NOTES

- 1 It is important that every console position have access to a black phone, mobile phone, or satellite phone.
- 2 It is important that every console position have the phone numbers of the positions on the others MCC. It is assumed that all console phone numbers will be available on a project-specific manning roster that is available to all partners. For example, for the

ISS project these numbers are also available at the NASA orbit manned roster that is accessible to all the space agencies participating in the project.

- 3 It is desirable that redundant voice systems on diversified power supply feeds be available.
- 4 It is desirable that alternate routes be available to patch around disrupted connections between MCCs.

3.3 DISCUSSION: EXAMPLE OF VOIP IMPLEMENTATION

An example of a VoIP implementation is the NASA Marshall Space Flight Center's (MSFC's) use of the local Internet Voice Distribution System (IVoDS).

IVoDS is based on a conference server/gateway server concept. The conference server contains the user loop configuration database (voice groups). Once authenticated with a conference server, users can monitor/talk (based on user permissions) within the loops for which they are privileged.

The conference servers are affiliated with gateway servers, which provide a matrix of voice transport circuit channels (e.g., E1/T1), accept multiple TDM circuits, and break down the voice channels for conference access.

IVoDS is scalable based on voice transport circuit quantity/type (gateway), as well as soft client quantity (conference). IVoDS is also scalable so as to be capable of providing fault-tolerant voice services for the real-time flight control team within the MCC.

IVoDS authentication is compatible with two-factor authentication, as well as X.500 management (reference [28]). IVoDS provides encrypted sessions between client and server, although it can be configured to operate unencrypted. Additionally, traditional network techniques such as access control lists and Virtual Private Network (VPN) can be used to better protect voice access to the required level.

Figure 3-2 shows the configuration schema of IVoDS in the next page.

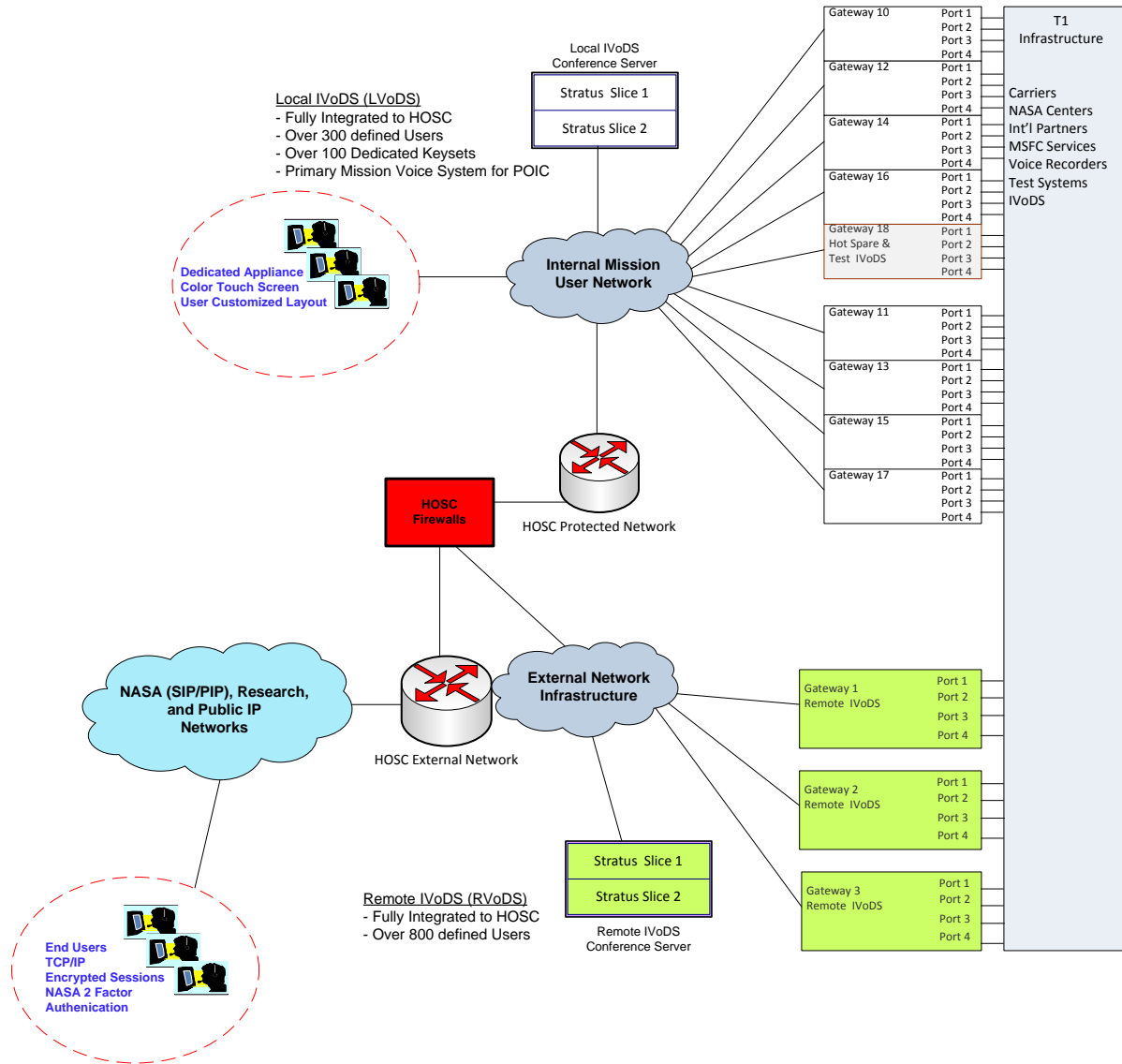


Figure 3-2: IVoDS Functionality Schema inside an MCC and to External Facilities

3.4 TERRESTRIAL COMMUNICATION BETWEEN MCCS

3.4.1 OVERVIEW

There are different voice system transport media that can carry the audio signals and audio files between the different MCCs. These transport media could be MPLS, IP, and/or satellite links:

- MPLS is an efficient and cheap network technology (e.g., leased line or commercial carrier). It is widely used, but has some problems for voice transmission:
 - Interruptions in the voice system for several seconds must be accepted. Interruptions can result from rerouting generated when a big company is

connected to the network or important maintenance takes place somewhere in the world. In spite of the best class (voice class) being used, these problems can cause interruptions up to 30 seconds. This is an important factor, as the applications must be more robust in order to survive these 'normal' issues.

- Clocking can be an issue and should be taken into account (a fall-back solution should be available).
- IP: IPv4 and IPv6 protocols work quite well inside of the MPLS network, having the same mentioned limitations. Public Internet can also be used; it works well but does not provide QoS.
- Satellite links are occasionally used for transmission of compressed voice data (see reference [22]).

RECOMMENDED STANDARD FOR VOICE AND AUDIO COMMUNICATIONS

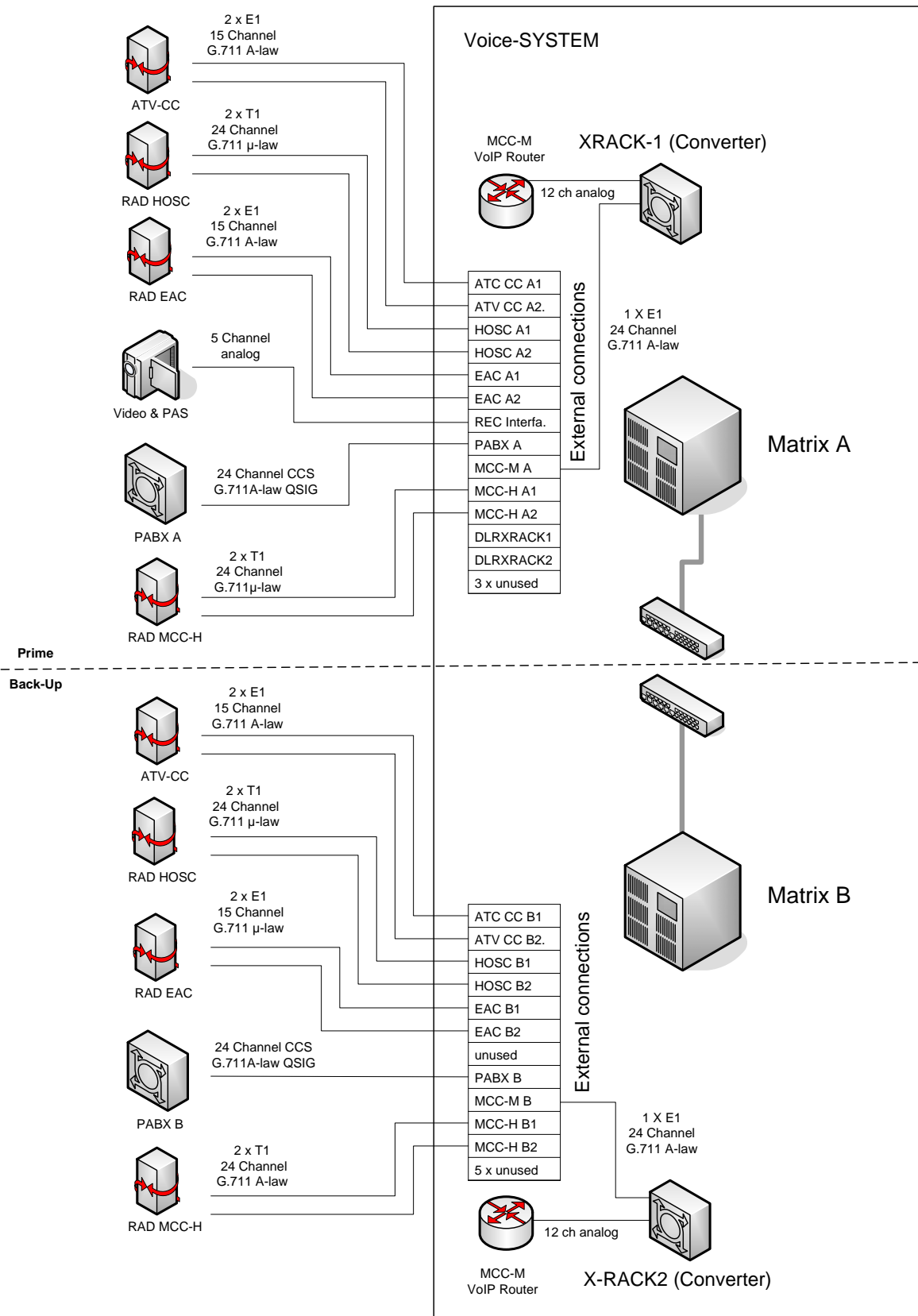


Figure 3-3: Example of Voice Connections between MCCs

3.4.2 TERRESTRIAL VOICE SYSTEMS BETWEEN MCCS: RECOMMENDED STANDARDS

3.4.2.1 A packet-switched network (MPLS/IP) should be used, in spite of the communication problems (packet loss).

NOTE – The quality is good enough and the costs considerably lower than circuit-switched alternatives or dedicated lines (e.g., ISDN).

3.4.2.2 The voice system shall use a reliable network (at least 99 percent availability).

3.4.2.3 The voice system in an MCC should be highly available (redundant and 99 percent available).

NOTES

1 It is desirable that redundant voice systems on diversified power supply feeds be available and that every console position should have access to a black phone, mobile phone, or satellite phone.

2 It is desirable that every console position have the phone numbers of the positions on the others MCC. Normally they are also available at the NASA orbit manned roster.

3.4.2.4 It shall be possible to define roles (positions) with the related permissions (T/L/M).

3.4.2.5 It shall be possible to define access permissions to the users to the associated roles (T/L/M) for all voice loops. X.500 (e.g., LDAP) authentication should be used.

NOTE – Maintaining appropriate access controls can be quite labor intensive since the user information and role assignments can be very dynamic.

3.4.2.6 It shall be possible to define and store voice formats.

3.4.2.7 Voice formats should be stored with unique human readable names.

3.4.2.8 Voice formats shall be specified in an interagency ICD.

3.4.2.9 Changes in the voice formats shall be specified in an Engineering Change Request (ECR).

3.4.2.10 It shall be possible to dynamically exchange the voice formats with an interruption of less than 3 seconds.

3.4.2.11 It shall be possible to edit the voice formats (offline) and apply them as required in 3.4.2.6 and 3.4.2.8.

NOTE – Changes for different activities (e.g., Mission, test or simulations) need to be well defined and agreed upon in advance between the different MCCs. Normally that is clearly reflected in a tool such as the Onboard Short Term Plan Viewer (OSTPV).

3.4.2.12 PTT should be implemented in all kinds of keysets (manual or automatic).

NOTE – PTT can be implemented physically via a button or pedal, or via VOX, where the voice level is monitored and the PTT is activated automatically.

3.4.2.13 VOX may be implemented on keysets.

3.4.2.14 If VOX is implemented, the VOX threshold must be adjustable between –26 dB and –94 dB.

3.4.2.15 The VOX system should also have a hang system (VOX time out) from 1 to 3 seconds to remain engaged during brief speech pauses. It must be possible to save and restore the VOX threshold settings for each individual user in a console position.

3.4.2.16 G.711 should be used for communications between MCCs and S/G if there are not data-rate limitations.

3.4.2.17 Voice class should be used if an MPLS network is implemented; the highest priority class should be used with any other network technology.

3.4.2.18 A minimum of 256 kb/s must be reserved per voice channel.

3.4.2.19 The codec shall not be changed during the transition through network links.

NOTES

- 1 Lossy voice codecs need to be employed carefully, and it is best that voice encoding occur only once (ideally) between end points. Multiple encode/decode cycles significantly affect voice quality and add latency.
- 2 Examples of a protocol and codec stack are IP/UDP/RTP/G.711 from end to end or IP/UDP/RTP/G.729 from end to end.

3.4.2.20 Synchronization:

- a) The clocking between the MCC voice systems must be synchronized.
- b) A fallback clock-synchronization mechanism should be available in case of loss of the primary synchronization mechanism.
- c) Clock synchronization via Global Positioning System (GPS) or Internal T1 clock shall be implemented in the external interface (e.g., E1/T1) of all the connected MCCs.

3.4.2.21 There shall be no more than 10 ms latency for communications inside of a control room and no more than 200 ms for connections to other rooms or facilities.

3.4.2.22 It shall be possible to have channelization. (association of voice loops to physical channels, e.g., S/G to channel 5).

NOTE – Voice formats and channelization need to be defined and agreed to between the MCCs before the systems can be used to communicate with each other. The voice formats are defined in the interagency ICDs.

3.4.2.23 It shall be possible to apply signaling to different loops. The signaling should be compatible to the specifications in reference [3]. CAS and CCS shall be selectable.

NOTE – In analog systems signaling is always present. In digital systems an open line may be used without signaling, but for important loops like S/G, signaling is required. This includes CAS signaling (often called ‘robbed bit’ in the Americas) on the T1s and CCS signaling for the PABX.

3.4.2.24 Voice recording shall be used for predefined loops between MCC.

NOTE – Voice recording is mandatory in human space missions.

3.4.2.25 All systems designed after 2015 must use VoIP.

3.4.2.26 Encryption should be implemented if VoIP is used, e.g., IPsec encryption using AES128 or a stronger algorithms.

3.4.2.27 High-performance voice conferencing systems using TDM with G.711 μ -law or A-law voice encoding shall be used in an MCC.

3.4.2.28 Voice quality shall not be sacrificed in deference to reducing data rate. End-to-end configuration must be considered (e.g., use G.711 and increase the data rate, do not change the code to reduce the data rate).

3.4.2.29 MOS shall be 4.0 or greater.

NOTE – Voice is considered mission critical and has the highest priority in the definition of the traffic in the network. Voice channels need to be considered first for bandwidth/data-rate allocation.

3.4.2.30 If E1/T1 or MPLS are used, even though the data are encrypted over the external network, encryption shall be used for the internal network.

3.4.2.31 It shall be possible to have voice channel or voice loop privatization for medical-related communications or for astronauts’ family loops also called PFC (Private family conference).

3.4.2.32 When privatized, a given voice channel or loop must be inaccessible to any users of the voice system besides those authorized by the ground controller or equivalent position (opt-in).

NOTE – Privatized voice loops will override the permissions hierarchy specified in 3.2.

3.4.2.33 QoS and/or QoE shall be monitored if possible.

NOTE – If Internet is used (VoIP) for the voice transmission, quality cannot be guaranteed. In any case, a voice recorder is needed as described in 2.4.2.

3.4.2.34 All systems designed after the year 2015 must use VoIP.

NOTE – If Internet is used (VoIP) for the voice transmission, quality cannot be guaranteed. In any case, a voice recorder is needed as described in 2.4.2.

3.4.2.35 SIP-based IP telephony is used by most of the space agencies and should be used for IP telephony.

3.4.2.36 For secure voice conferencing where video is required (e.g., VSee) screen sharing software shall be used for secure video chat. The system used must be previously agreed.

3.4.2.37 Soft keysets should not be used for real-time operations with an MCC.

3.4.2.38 Recorded audio shall be transmitted as files using the established file distribution methodologies.

NOTE – Redundancy of the network and voice system (matrix or switch) should be implemented for voice communications between MCCs.

3.5 TERRESTRIAL COMMUNICATION BETWEEN AN MCC AND EXTERNAL ENTITIES

3.5.1 OVERVIEW

In many cases temporary voice loops are needed for some tests, public relations events, or special work with external facilities. In most of the systems it is possible to patch one or more phones into a voice loop (or several loops) to establish external communication.

An example of an extension of an MCC to external entity is the Remote IVoDS system. Remote IVoDS provides a complete voice package for end users (e.g., universities) via any available IP-supported network (e.g., the Internet) and an MCC circuit matrix as shown in figure 3-2.

There are other similar applications, like external soft keysets connected to the MCC via the Internet. Because of the many communications problems of soft keysets, such as stability

issues, loss of connection, and re-login necessitated by unexpected disconnections, they are not recommended for real-time operations.

PSTN dial-in is used widely for public relations events, especially when the event takes place outside of the closed voice system. Normally the user makes a phone call to a number on a PABX and the call is patched to a voice loop. In those cases there are several classic problems:

- a) The user does not have PTT capabilities: the device may be a microphone or black phone (hardwired telephony communications).
- b) The user does not have experience/training/etiquette to talk in a voice loop.
- c) In communication using video, lip synchronization must be included.

Another possibility is to use a mobile phone to call a gateway and, with a passcode, access some voice loops from the mobile phone.

NOTE – As a general provision, the ‘Plain Old Telephone System’ (POTS) may be used for PR events.

3.5.2 TERRESTRIAL VOICE SYSTEMS BETWEEN AN MCC AND EXTERNAL FACILITIES: RECOMMENDED STANDARDS

3.5.2.1 PTT must be supported on the interface.

3.5.2.2 It shall be possible to have constant keying if a keyset is patched to a POTS system.

3.5.2.3 If the external facility does not support PTT, the functionality must be provided by the MCC.

3.5.2.4 Lip sync shall be supported for public relations events.

NOTES

1 It is desirable that timestamps be used in the video and voice systems if possible to enable an automatic lip sync (requires special equipment).

2 It is desirable that video and audio be sent embedded (HD-SDI, reference [31]) from space and synchronized onboard if possible.

3.5.2.5 It shall be possible to separate the embedded audio from the video.

3.5.2.6 SIP-based IP telephony (e.g., Cisco) is used by most of the space agencies and should be used for IP telephony.

3.5.2.7 For secure voice conferencing where video is required (e.g., VSee) screen sharing software shall be used for secure video chat. The system used must be previously agreed.

3.5.2.8 It shall be possible to have voice channel or loop privatization for medical-related communications or for astronauts' family loops also called PFC (Private family conference).

3.5.2.9 When privatized, a given voice channel or loop must be inaccessible to any users of the voice system besides those authorized by the ground controller or equivalent position (opt-in).

NOTE – Privatized voice loops will override the permissions hierarchy specified in 3.2.

3.5.3 VOIP TRANSMISSION USING IP OR PSTN NETWORKS

VoIP shall be transmitted using IP or PSTN networks.

NOTE – For this communication technology, some extra considerations need to be taken into account:

- Security concerns, including authentication and authorization, need to be addressed, especially for remote access. X.500 (reference [28])/LDAP authentication, firewall border control, VPN access, and/or two-factor authentication may be required.
- Soft switch is typically used to control connections at the interface between circuit-switched and packet-switched networks.
- Soft switch is less expensive than a standard gateway, but has the disadvantage of requiring software running on a computer.

4 CASE 2: SPACE-TO-GROUND VOICE SYSTEMS

4.1 OVERVIEW

Within the frame of manned space flight missions, voice communication is the most important part among all types of S/G communications. Voice is considered mission critical.

There are two types of S/G communications: short-distance communications and long-distance communications.

- Short-distance communications are confined to the Earth/Moon or other planetary systems. A short-distance voice communication system implies real-time communications and the use of S/G voice loops.
- Long-distance communications include interplanetary or deep-space communications. As distance imposes communication delays beyond approximately 30 seconds, file transfer may be used rather than real-time communication.

The exchange of voice and audio communications between vehicles in space and Earth-bound control centers necessarily involves satellites and ground stations as well as the space and deep-space networks. S/G voice communications use the following assets:

- Voice communication primarily uses Radio Frequency (RF) links, with different frequencies (S-band, Ku-band, and Ultra High Frequency [UHF]).
- During launch, the Soyuz vehicle uses VHF1 to communicate to MCC-Moscow (MCC-M); uplink: 130.167 MHz; downlink: 121.750 MHz for communications to the ISS and S/G. VHF2 is used for docking operations for direct communications to the ISS.
- Zhengzhou uses systems similar to the Soyuz vehicle.
- Space X uses G.729 for communications between ISS and the Dragon capsule. It also uses RF links for S/G communications.
- Other vehicles like Dragon, HTV, Progress, or Cygnus are not manned and have only extensions of the ISS voice system when they are docked to the station. Dragon and CST-100 Starliner will be inhabited and have their own space-to-ground communications links
- For Extra Vehicular Activities (EVA), UHF (414.2 MHz) is used with a direct link to the spacecraft. The communication to the Earth uses the spacecraft's system, which is usually S-band.
- On the ISS, Internet is available over the Ku-band. Various Internet applications can be used by the astronauts. The connection uses Tracking and Data Relay Satellite System (TDRSS).

- The voice loops S/G 1 and 2 use Modified Residual Excited Linear Predictive Coding, a proprietary Motorola Protocol (via S-band). The new S/G loops, like 3 and 4, use G.729 (via Ku-band).

4.2 SHORT-DISTANCE VOICE COMMUNICATION

4.2.1 VOICE AND AUDIO DATA FORMAT STANDARDS

4.2.1.1 For voice and audio data formats, one of the following standards should be used:

- a) ITU-T G.729/G.729A (reference [9]);
- b) ITU-T G.711 (reference [1]);
- c) AAC: ISO/IEC 13818-7 or ISO/IEC 14496-3 (references [20], [21]; see also reference [E2]).

4.2.1.2 G.729/G.729A may be used in any ground-to-space communication links; G.729 alone (without G.729A) is preferred.

4.2.1.3 G.711 and AAC shall be used for communication links when data rates are adequate.

4.2.1.4 AAC may be used if high-quality audio transmission is required in real-time voice communication.

4.2.1.5 AAC uses variable bit rates from 8 kb/s to 529 kb/s depending on the data-rate limitations of the communication link between ground and spacecraft. For good voice and audio quality, at least 128 kb/s shall be used.

4.2.1.6 Video and audio should be sent embedded (HD-SDI, reference [31]) from space and synchronized onboard if possible

4.2.2 VOICE AND AUDIO DATA TRANSMISSION

4.2.2.1 Overview

Short-distance transmission mechanisms for voice and audio data between ground and spacecraft include voice over CCSDS data links and VoIP.

4.2.2.2 Voice over CCSDS Data Links

Advanced Orbiting Systems (AOS) Virtual Channel Access (VCA), Bitstream, and Insert services (reference [29]) should be used to transmit voice and audio data.

NOTE – While the frames of voice and audio data have fixed size and are produced with fixed rate, any isochronous service provided by CCSDS data link protocols can be used to transmit them.

4.2.2.3 Voice over IP

4.2.2.3.1 The IP network connecting the ground and the spacecraft shall be implemented with standards conforming to reference [22].

4.2.2.3.2 The RTP/UDP/IP/CCSDS Encapsulation (reference [32])/AOS (reference [29]) protocol stack should be used in VoIP for ground-to-space voice systems.

4.2.3 VOICE AND AUDIO DATA FORMATS FOR FILES

The coding and compression mechanism for non-real-time audio file transmission shall conform to the AAC specification (reference [20]; see also reference [E2]).

NOTE – The recommendation is to migrate to AAC. MP3 (ISO/IEC 13818-3) (reference [E3]) is not recommended because it is a legacy format; however, MP3 is still used in many audio files and probably will continue to be used for some years.

4.3 LONG-DISTANCE AUDIO COMMUNICATION

4.3.1 OVERVIEW

As distance imposes communication delays beyond approximately 30 seconds, the mission could choose to use file transfer rather than real-time communication.

4.3.2 RECOMMENDED STANDARDS

4.3.2.1 The coding and compression mechanism for audio file transmission shall conform to AAC.

NOTE – If using speech-to-text and text-to-voice conversion, the same application needs to be used in space and on the ground. This application may or may not have an interface or be related to the voice system.

4.3.2.2 Audio file transmission should be scheduled with high priority.

NOTE – Voice and audio are considered mission critical for human space flight.

4.3.2.3 Files should be transmitted by any general file transmission protocol, e.g., via CFDP Class 1 or Class 2 (reference [6]).

NOTE – CFDP can be implemented directly over a CCSDS Data Link Layer protocol such as AOS, TC, TM, or Proximity-1, or over an internetworking protocol such as IP (reference [13]) or DTN (references [7] and [8]).

4.3.2.4 For long-distance missions like Mars and beyond, DTN technology should be used.

4.4 DISCUSSION—TELECONFERENCING

Public and private tele/videoconferences using commercial products like VSee or Skype are used by the astronauts to communicate from the international space station to the ground and vice versa, using IP via a proxy.

4.5 SPACE-TO-GROUND VOICE SYSTEMS: RECOMMENDED STANDARDS

4.5.1 Real-time audio should be encoded according to G.729 for low-data-rate delivery, and with G.711 if high enough data rate is available (at least 128 kb/s per channel).

NOTE – In the particular case of the ISS, Ku- and S-band may be used. For this reason, G.711 can be used with the high bandwidth of Ku-band. G.729 is preferred for S/G voice loops and communications with vehicles where S-band is used.

4.5.2 Users and permissions (roles, e.g., T/L/M) associated to the login role shall be used on the ground segment.

4.5.3 For long distance, file-based transfer, AAC format files should be used via CFDP using either Class 1 or Class 2 (reference [6]).

NOTE – CFDP can be implemented directly over a CCSDS Data Link Layer protocol such as AOS, TC, TM, or Proximity-1, or over an internetworking protocol such as IP (reference [13]) or DTN (references [7] and [8]).

4.5.4 PTT in S/G shall be implemented. If automatic detection is used, it must be calibrated with different voice levels (between -26 and -94 dB).

NOTE – Quindar tones are still used as part of the communication, but they are not an ongoing requirement.

4.5.5 Channelization:

- a) If a direct link to the spacecraft is available, CCSDS standard data-link mechanisms (e.g., reference [24], [25], [29], or [26]) may be used.
- b) If there is no direct connection to the spacecraft (e.g., if the the communication is through another MCC and not via a single space link extended via CCSDS CSTS), the requirements 3.2.8–3.2.13 are applicable.

4.5.6 Signaling:

a) Signaling shall be used for S/G loops (CAS or CSS).

NOTE – In analog systems signaling is always present. In digital systems an open line can be used without signaling.

b) Channels and the kind of signaling should be well defined in advance and described in the ICD.

4.5.7 Encryption and channel privatization must be used for medical conferences and experiments using medical data.

NOTE – Security restrictions need to be taken into account if commercial products are used, especially using the Internet.

4.5.8 QoS and/or QoE should be monitored.

NOTE – If the Internet is used (e.g., VoIP) for the voice transmission, quality cannot be guaranteed.

4.5.9 Voice recording equipment shall be available (see 2.4.2).

4.5.10 SIP-based soft phone (e.g., Cisco IP Communicator) should be used if the station laptops are used for IP telephony. On the ground any IP device compatible with SIP-based software (e.g., Cisco IP Communicator) may be used.

NOTE – When designing a crewed mission, alternative conferencing or IM systems such as VSee or Skype need to be taken into account.

4.5.11 Secure voice conferencing that requires video (e.g., VSee) and screen sharing software shall be used for secure video chat.

NOTE – A high-bandwidth link (e.g., Ku-band) would be required for this application.

4.5.12 Multiple voice channels (redundancy) must be available because of the importance of voice communication in space missions.

5 CASE 3: EMERGENCY VOICE COMMUNICATION

5.1 OVERVIEW

Emergency communications is normally performed between MCCs using a black phone, a mobile phone connected to a commercial cellular service, or a satellite phone. The ground controllers must have access to a list of telephone numbers for the console positions they need to contact in the event of an emergency communication situation where the primary voice communication system is unavailable.

In space, when normal means of voice communication are not available from the primary vehicle as a result of an emergency situation, noncritical systems (including primary radio equipment) need to be powered down to conserve power and bandwidth. Low-power communications equipment can then be used to transmit voice and a limited amount of command/telemetry. In these emergency situations, voice communications can be encoded as G.729 data and transmitted over CCSDS AOS VCA (virtual channel access) service with no IP/UDP/RTP protocol wrapping (requiring a data rate of roughly 10.5 kb/s).

Support for unscheduled NASA Very High Frequency (VHF) ground stations will be for spacecraft contingency and/or spacecraft emergency conditions, when S-band S/G communication is unavailable due to S-band communication failures caused by problems such as:

- ISS/Soyuz fire;
- ISS/Soyuz depressurization;
- ISS/Soyuz crew medical emergencies;
- extended SHO (scheduled order) gaps during any complications on the ISS.

During this type of emergency, ISS VHF1, VHF2, and Soyuz VHF systems will be activated simultaneously to maximize communications coverage to ensure contact with the crew during an emergency ingress of Soyuz.

Examples of configurations used during an emergency situation with the Russian and U.S. ground segments are listed below:

- ISS VHF1 is used for S/G communication over Russian ground stations (RS ISS S/G 1 MH21) and also used as an alternate link for S/G communications when ISS is over NASA VHF ground stations. Uplink: 139.208 MHz; downlink: 143.625 MHz.
- Soyuz VHF2 is used for S/G over Russian ground stations (RS ISS S/G 2 MH24) and space-to-space communication. This is the only VHF frequency pair carried on Soyuz. Soyuz VHF2 communication through NASA VHF ground stations is authorized for emergency use only per Flight Rule (D3-1). Uplink: 130.167 MHz; downlink: 121.750 MHz.

- Goddard Voice will configure DFRC and WSC S/G circuits. MCC-M will configure RS ISS S/G 1 MH21 (ISS VHF1) and/or RS ISS S/G 2 MH24 (Soyuz VHF2).
- The ISS VHF transmitter will be constant keyed, which is necessary to verify signal reception and link integrity, as well as to ensure minimum required link margin is used.

Orion and Space X use CCSDS standards for emergency communications.

In cases involving landers, rovers, orbiting constellations, and orbiting relays, Proximity links should be considered (references [25], [26], and [27]). This protocol uses a frequency band in the 400 MHz band so as to reduce complexity of the vehicle: 437.1000 MHz, 440.7425 MHz, 444.3850 MHz, and 448.0275 MHz.

In the specific case of an astronaut using an Extravehicular Mobility Unit (EMU), a communications carrier assembly leverages two (redundant) UHF channel transmitters and three single-channel receivers to provide voice (and limited telemetry) communication.

5.2 EMERGENCY VOICE COMMUNICATION: RECOMMENDED STANDARDS

5.2.1 The voice system shall use a reliable network (at least 99 percent availability).

5.2.2 The voice system in an MCC should be highly available (redundant and 99 percent of available).

NOTES

- 1 It is desirable that a redundant voice system on diversified power supply feeds be available.
- 2 It is desirable that alternate routes be available to patch around disrupted connections between MCCs.
- 5.2.3** Every console position should have access to a black phone, mobile phone, or satellite phone.

NOTE

- 1 It is important that every console position have the phone numbers of the positions on the other MCC.
- 2 It is desirable that emergency procedures be simulated on a regular basis to allow ground personnel to have adequate training.
- 3 The ISS program has a requirement that allows for 30 minutes of crew time per month for VHF emergency communications checkouts, which allows for two 15-minute checkouts monthly.

5.2.4 Emergency channel voice should be encoded as G.729 (reference [9]) and transmitted over CCSDS AOS VCA service using 8 KHz with no IP/UDP/RTP or VoIP protocol wrapping.

5.2.5 PTT shall be used on the ground.

5.2.6 Signaling, or loops keyed (constant keying) shall be used from the spacecraft.

5.2.7 Voice, cautions, and warnings shall have the highest priority in the communications channels.

NOTE – All the other parameters discussed in 2.6 are not applicable here.

6 CASE 4: ISS RENDEZVOUS, PROXIMITY, AND DOCKING OPERATIONS VOICE COMMUNICATIONS

6.1 OVERVIEW

6.1.1 INTRODUCTION

Space rendezvous of an active, or ‘chaser’, spacecraft with an (assumed) passive spacecraft can be divided into several phases and typically starts with the two spacecraft in separate orbits; the spacecraft are typically separated by more than 10,000 km (6,200 mi). For the voice and audio communications covered in this book, the phases are only important because of the voice system used in each case. It must be clear that the vehicles are manned and have voice systems.

6.1.2 RENDEZVOUS

Rendezvous occurs when two spacecraft have the same orbit, e.g., a vehicle and the ISS. It can take hours to days to complete the rendezvous. Vehicles communicate with each other using their S/G voice systems. In the case of line-of-sight, they can contact one another directly vehicle to vehicle, using S/G as backup. S/G is also used to inform the operations personnel on ground.

6.1.3 PROXIMITY

Proximity occurs when two spacecraft are very close to each other, e.g., around 300 m (1–2 hours of operations till docking). Vehicles use direct links for voice communications and S/G to keep the operations teams on the ground up to date (for Soyuz these links start at 200 km). In some cases only direct communication is used in order to avoid any interference.

6.1.4 DOCKING

Docking is considered to be between around 300 m and docking completion (from several minutes up to 1 hour of operations). Vehicles use direct links for voice communications and S/G to keep the operations teams on the ground up to date.

NOTE – VHF is used for near-field, vehicle-to-vehicle communication.

6.2 RENDEZVOUS, PROXIMITY, AND DOCKING OPERATIONS: RECOMMENDED STANDARDS

6.2.1 For a rendezvous, standard S/G communications should be used.

6.2.2 For proximity and docking, the normal spacecraft S/G voice loop should be used, plus spacecraft-to-spacecraft communication.

NOTE – This is a direct connection using VHF or S-band communication channels, going to the ground almost simultaneously. That is a similar system to what is used for EVA. The communication is between the EVA crewman and the vehicle, using the vehicle's voice system.

6.2.3 The ISS/spacecraft voice system should be radiated down the S/G communication link.

NOTES

1 For communication to the ground, ground stations (ground telemetry stations) or TDRSS are used.

2 Soyuz, when approaching proximity operations is done manually, severs the connection to the ground and uses only the VHF/S-band connection from spacecraft to spacecraft.

6.2.4 PTT, signaling, voice encryption, and QoS have the same requirements as in section 3.

NOTE – All the other parameters discussed in 2.6 are not applicable here.

7 CASE 5: SEARCH AND RESCUE VOICE COMMUNICATIONS

7.1 DISCUSSION

7.1.1 INTRODUCTION

At the moment (2016) and for the next few years, the only rescue vehicle on the ISS is the Russian Soyuz capsule. For this reason, this section is strongly oriented to the Russian capsule.

SAR voice communication concerns messaging between spacecraft crew and rescue teams.

Voice communications are crucial in a rescue. During rescue operations, it is essential that standardized communication procedures be used.

The primary means of communication in a rescue is the survival radio or satellite telephone supplied with the survival equipment. Each crew member needs to be familiar with the radio operations.

In addition to the Automatic Direction Finder (ADF) capability, the U.S. government also uses satellites that monitor guard frequencies for emergency locator transmitter broadcasts. For example, a AN/PRC-149 Rescue Radio includes a GPS and a Cospas-Sarsat beacon. It operates on 121.5 MHz, 243.0 MHz, and 406.025 MHz.

7.1.2 COSPAS-SARSAT

The International Cospas-Sarsat Program is a satellite-based SAR distress, alert-detection, and information-distribution system, established in 1979 by Canada, France, the U.S., and the former Soviet Union. It is best known as the system that detects and locates emergency beacons activated by aircraft, ships, and backcountry hikers in distress. Over the years, many countries have joined the project, either as providers of ground segments or as user states. The system consists of a ground segment and a space segment.

More recently, the Cospas-Sarsat system has been designed to detect only alerts transmitted by 406 MHz beacons that incorporate GPS receivers. Such beacons transmit highly accurate positions of distress almost instantly to SAR agencies via the GEOSAR satellites.

7.1.3 SOYUZ SEARCH AND RESCUE CASE SCENARIO

The Soyuz commander or the Flight Directors (FDs) of MCC-M and MCC-Houston (MCC-H) are authorized to formally declare a Soyuz spacecraft emergency which requires an emergency landing. If this were to occur, the following notification requirements will immediately be met:

RECOMMENDED STANDARD FOR VOICE AND AUDIO COMMUNICATIONS

- a) Depending on whether the landing area is within Russia or territories of the former Soviet Union, or in North America, the ISS or MCC-H FD informs the corresponding SAR personnel.
- b) In America, the ISS FD notifies the Air Force Rescue Coordination Center (AFRCC) in Langley, Virginia, and the MCC-H Trajectory Operations Officer (TOPO).
- c) AFRCC and TOPO notify USSPACECOM, the Department of Defense (DoD) manned spaceflight group, and other U.S. agencies/organizations, as required.
- d) Once notified, the AFRCC coordinates the efforts of the local SAR organization that will locate, assist the crew, and transport them to definitive medical care as required. The AFRCC provides the SAR team with checklists and pictures of hazards associated with the Soyuz and how to remove the crew. AFRCC informs the DoD Manned Space Flight Support (DDMS).
- e) The DDMS coordinates and provides command and control for all required DoD assets. The MCC-H Landing Support Officer (LSO) provides coordination with these external agencies. If LSO is not on console, TOPO provides coordination.

NOTE – The Soyuz contingency landing areas need to be evaluated with respect to terrain, experience of the landing support personnel, SAR capabilities, access to definitive medical care, and in-place international support agreements. However, since Soyuz requires a ballistic reentry with a g-load near 8–10 g, the assumption is the decision to perform an emergency landing is made only in the direst of circumstances.

- f) To mitigate the risk from the above factors, the MCC-H Spaceflight Meteorology Group forecasts GO/NO GO conditions. Once a decision is made to land the Soyuz at an emergency site, the ISS FD, upon concurrence with MCC-M, can call in the MCC-H Spaceflight Meteorology Group to update the wind forecasts if time permits. This assumes MCC-H/M will receive the updated forecasts before a final landing area decision is required.
- g) In Russia, MCC-M coordinate directly with the rescue teams. The crew uses a Satellite telephone to communicate with the rescue teams.

NOTES

- 1 Splashdown was used by the Apollo missions and will be used by Space-X Dragon and Orion in the near future. From the voice point of view, there are no differences between a landing on the water or on the land.
- 2 Chinese Zhengzhou and Russian Soyuz capsules normally land on the ground but may also land on water. NASA's Commercial Crew Program's Boeing Starliner CST-100 plans ground landings also.
- 3 Soyuz and Zhengzhou have a Cospas-Sarsat radio and an Iridium satellite phone in the capsule.

- 4 The Soyuz cosmonauts' rescue kit has a portable radio transceiver (VHF and HF).

7.2 SEARCH AND RESCUE VOICE COMMUNICATION: RECOMMENDED STANDARDS

7.2.1 For SAR the standard S/G voice system should be used as long as possible.

NOTE – After landing, it is important that unnecessary equipment be shut down and only communications be active.

7.2.2 The crew should use an emergency radio or satellite telephone.

7.2.3 The SAR radio in the vehicle should be a battery-powered radio, e.g., PRC-112G, B-85581 or PRC-149, having UHF, VHF, GPS, satellite radio, and a Cospas-Sarsat beacon.

ANNEX A

PROTOCOL IMPLEMENTATION CONFORMANCE
STATEMENT (PICS) PROFORMA

(NORMATIVE)

A1 INTRODUCTION

A1.1 OVERVIEW

A1.1.1 Columns

A1.1.1.1 General

In order to reduce the size of tables in the PICS proforma, notations have been introduced that have allowed the use of a multi-column layout, where the columns are headed 'Status' and 'Support'. The definition of each of these follows.

A1.1.1.2 Status Column

The 'Status' column indicates the level of support required for conformance to the standard. The values are as follows:

- M** Mandatory: support is required.
- C** Conditional: support depends on conditions stated below table.
- O** Optional: support is permitted for conformance to the standard. If implemented, it must conform to the specifications and restrictions contained in the standard. These restrictions may affect the optionality of other items.
- O.n** Support of at least one of the options labeled with the same number *n* is mandatory.
- n/a** Not applicable.

A1.1.1.3 Support Column

The 'Support' column is completed by the supplier or implementer to indicate the level of implementation of each feature. The proforma has been designed such that the only entries required in the 'Support' column are:

- Y** Yes, the feature has been implemented.
- N** No, the feature has not been implemented.
- The item is not applicable.

A1.1.2 Item Reference Numbers

Within the PICS proforma, each line that requires implementation detail to be entered is numbered at the left-hand edge of the line. This numbering is included as a means of uniquely identifying all possible implementation details within the PICS proforma. The need for such unique referencing has been identified by the testing bodies.

The means of referencing individual responses is to specify the following sequence:

- a reference to the smallest subsection enclosing the relevant item;
- a solidus character, ‘/’;
- the reference number of the row in which the response appears;
- if, and only if, more than one response occurs in the row identified by the reference number, then each possible entry is implicitly labeled a, b, c, etc., from left to right, and this letter is appended to the sequence.

A2 COMPLETION OF THE PICS

The implementer shall complete all entries in the column marked ‘Support’. In certain subsections of the PICS proforma, further guidance for completion may be necessary. Such guidance shall supplement the guidance given in this subsection and shall have a scope restricted to the subsection in which it appears. In addition, other specifically identified information shall be provided by the implementer where requested. No changes shall be made to the proforma except the completion as required. Recognizing that the level of detail required may, in some instances, exceed the space available for responses, a number of responses specifically allow for the addition of appendices to the PICS.

Voice formats should be specified between the Agencies in order to carry out the testing. For cases 3, 4 and 5 the dedicated voice loops and the frequencies used must be specified if more than one Agency is involved.

A3 REFERENCED BASE STANDARDS

Voice and Audio Communications (V&A in this document and in the yellow book) is the only base standard referenced in the PICS. In the tables below, numbers in the Reference column refer to applicable subsections within this document.

A4 GENERAL INFORMATION**A4.1 IDENTIFICATION OF THE PICS**

Date of statement (yyyy-mm-dd)	
PICS version	
System Conformance Statement cross-reference	
Other information	

A4.2 IDENTIFICATION OF THE SYSTEM SUPPLIER / TEST LABORATORY CLIENT

Organization name	
Contact name	
Address	
Telephone	
E-mail	
Other information	

A4.3 IDENTIFICATION OF THE IMPLEMENTATION UNDER TEST

Implementation name	
Implementation version	
Voice system	
Codec used	
Type of keyset (hard, soft-keyset, etc.)	
Type of transport layer	
Special configuration (port, IP addresses, etc.)	
Voice loops definition (Voice loops used)	
Voice format	
Other information (e.g., PaBX numbers, patched loop)	

NOTE – To test cases 3, 4, and 5, a separated PICS will be needed. Since these cases have been used operationally for decades with very clear established communication rules between RSA and NASA, the PICS for these cases is not included in this book.

A4.4 EQUIPMENT CONFIGURATION FOR VOICE SYSTEM IN DETAIL

Source Device IP address / Ports / PABX number	
Source Device Netmask (VoIP)	
Source Device Gateway (VoIP)	
Source Device (T1 or E1)	
Destination Device (T1 or E1)	
Destination IP Addresses / Ports /PABX number	
Network Protocol	
Source /destination UDP ports (if any)	
Codec (G.711, G.728, etc.)	
Voice loop Name (s)	
Voice Format Source / Voice Format Destination	
Other information (e.g., micro or telephone with mute function)	

A4.5 GLOBAL STATEMENT OF CONFORMANCE

Are all mandatory features implemented? (Yes or No)	
---	--

NOTE – If a positive response is not given to this box, then the implementation does not conform to the standard.

A5 REQUIREMENTS LIST

A5.1 INTERFACES

Item	Transport Protocol Feature	Reference	Status	Support
1	T1 (TDM)	3.2.19/3.4.2.27	O.1	
2	E1 (TDM)	3.2.19/3.4.2.27	O.1	
3	VoIP	3.2.20/3.4.2.25/3.4.2.34	O.1	
4	Phone line (POTS)	3.5.2.2	O.1	
5	VHF (satellite links for S/G)		O.1	

NOTE – The O.1 options specify the interface used to the communication between the Agencies or to externals. When one option is chosen it becomes mandatory for the communication. It is possible to have also more than one communication interface.

A5.2 SUPPORTED PARAMETERS

Item	Protocol Feature	Reference	Status	Support
1	MPLS /IP network	3.2.1	O.2	
2	Internet /IP network	3.2.1	O.2	
3	Compatible voice formats	3.2.4/3.2.6	M	
4	Common voice loops (pre-defined)	3.2.3	M	
5	PTT	3.2.10/3.4.2.12/ 3.5.2.1/4.5.4/ 5.2.5	M	
6	Common Codec	3.2.13	C1	
7	Synchronization (E1/T1)	3.4.2.20	M	
8	Latency	3.2.14	O	
9	Channelization (voice format)	3.2.16	C2	
10	Signaling	3.2.15	C3	
11	Voice Recording	3.2.14	M	
12	Private channels (medical)	3.4.2.31– 3.4.2.32	M	
13	Redundancy	3.4.2.3/4.5.12/ 5.2.2	C4	
14	Audio file format	4.2.3/4.3.2.1	M	
15	User role assignment and security	3.2.2–3.2.3/ 3.4.2.4–3.4.2.5/ 4.5.2	M	
16	Keypad	3.4.2.37	M	

C1: O if converters are used; otherwise M.

C2: M if there are more voice loops than channels; otherwise O.

C3: M if S/G; otherwise O.

C4: M inside an MCC; otherwise O.

A5.3 INTEROPERABILITY

Item	Protocol Feature	Reference	Status	Support
1	Real time voice	3.2/3.4/3.5/4.2	M	
2	Playback voice and audio	3.2.18/3.4.2.24/4.5.9	M	
3	End user capabilities	3.2.10/3.2.11/ 3.4.2.12/3.4.2.13/ 3.5.2.2	O	

A5.4 PROTOCOLS

Item	Protocol Feature	Reference	Status	Support
1	RTP	3.2.20.1/4.2.2.3.2	O.3	
2	RTCP	3.2.21/3.4.2.33/4.5.8	O.3	
3	SRTP	3.2.20.2/3.4.2.26	O.3	
4	E1/T1	3.4/3.5	O.3	
5	TCP/IP	3.4/3.5	O.3	
6	UDP	3.2.20/3.4.2.25/ 3.4.2.34/4.2.2.3.2	O.3	
Item	VoIP			
7	SIP	3.2.20/3.4.2.25/3.4.2.34	O.3	
8	H.323	3.2.20/3.4.2.25/3.4.2.34	O.3	
9	MGCP	3.2.20/3.4.2.25/3.4.2.34	O.3	
10	MEGACO (H.248)	3.2.20/3.4.2.25/3.4.2.34	O.3	
Item	Codec			
11	G.711	3.2.13/3.2.19/3.4.2.16/ 3.4.2.27/4.2.1.1/4.5.1	O.4	
12	G.729	4.2.1.2/4.5.1/5.2.4	O.4	
13	G.728	4.2	O.4	
Item	Audio files transmission			
14	AAC	4.2.1	O.5	
15	MP3	4.3.2	O.5	
16	DTN	4.3.2.3/4.3.2.4/4.5.3	O	
17	CFDP	4.5.3	O	
Item	Radio			
18	VHF	7.2.3	O	

A5.5 SUPPORTED OPERATIONS

Item	Protocol Feature	Reference	Status	Support
1	MCC to MCC	3.4	M	
2	MCC to the Public Affairs	3.5	M	
3	MCC to S/G	4.2	O	
4	Emergency communications	5.2	O	
5	Docking / Rendezvous communications	6.2	O	
6	SAR	7.2	O	
7	Audio and Data file transfer	4.3	M	

NOTE – In this case, the optional items can be achieved through terrestrial connections between MCCs, e.g., S/G through NASA or RSA.

A5.6 PRIVACY**A5.6.1 General**

Item	Protocol Feature	Reference	Status	Support
1	Data Privacy (private channels)	3.4.2.31-3.4.2.32/ 3.5.2.8-3.5.2.9 4.5.7	M	
2	Data Integrity	3.2.20.2/3.4.2.26	M	
3	Authentication of communicating entities (LDAP, X.500)	3.2.3/3.4.2.5	M	
4	Control of access of resources (LDAP, X.500)	3.2.3/3.4.2.5	M	

A5.6.2 Encryption Procedures for Private Voice

Encryption procedures are applicable for private or medical voice. Encryption is typically peer-to-peer based on the application being used. Tests are not required for the PICS proforma. Tests of individual applications are to be performed prior to use in flight.

ANNEX B

SECURITY, SANA, AND PATENT CONSIDERATIONS

(INFORMATIVE)

B1 SECURITY CONSIDERATIONS

B1.1 INTRODUCTION

Security concerns specific to audio communications exist where there are requirements to ensure that transmitted audio not be disclosed, altered, spoofed, or redistributed without authorization.

Manned space systems frequently have requirements to provide capabilities for private audio communications between flight crew and ground operations personnel, physicians, families, and/or news media outlets. Ground systems that receive and/or redistribute such audio have both similar privacy requirements and additional threats and countermeasures.

The focus of this section is on the specific protocols and methods recommended earlier in this document. It may be necessary to implement security services at other layers within the protocol stack to account for a) audio and voice distribution, b) different classes of audio data or end users, or c) protection of data during unprotected portions of the complete end-to-end transmission (e.g., across ground networks for public relations events). The specification of security services at other layers is outside the scope of this document.

B1.2 SECURITY CONCERNS WITH RESPECT TO THIS CCSDS DOCUMENT

B1.2.1 General

The most common method of applying security to digital audio is through the use of multimedia container formats. Containers provide a file-based mechanism for exchanging, processing, and storing interleaved fragments of audio, metadata, and/or other content (e.g., annotations added to a recording). Most container files may be read or written by ordinary computer systems in the same manner as other files.

For the transmission of voice in real time, as is needed for operations, the security is implemented using codecs (as described previously in this book) and encryption of the data over the network, as is done in MPLS or the E1/T1 interfaces.

For voice communication, it is necessary that the person who is speaking or listening in a voice loop have the required access permissions (role) and have a user ID and password controlled by the system, permitting logging into the voice system as well.

If VoIP is used, the transmission of the voice data must fulfill the same security requirements as the transmission of data in an IP network. For VoIP transmission, Internet Protocol Security/VPN tunnels are recommended.

Typical VoIP protocols (e.g., SIP, RTP, Cisco VoIP, Skype, or Asterisk) have their own security features, but are less suitable for space business and transmission over the Internet. More information about VoIP security can be found in Voice over IP Security Alliance (VoIPSA) recommendations (see <https://www.voipsa.org>).

B1.2.2 Data Privacy

B1.2.2.1 General

The first major area of security concern is privacy, the requirement that the audio not be heard by any other than the intended recipient(s).

Limited privacy can be achieved by employing protected data transport at lower layers and restricting distribution. Virtual channel assignment over CCSDS RF links is not, by itself, an effective method of ensuring privacy.

Better privacy can be achieved using cryptographic techniques, which may be applied to all or part of a container format. Since any transcoding or decoding system must be able to read and pass metadata in order to process the stream, audio metadata must be readable at these points. It is therefore highly desirable to have the capability to perform selective encryption of container file fields (e.g., audio but not metadata).

VoIP data commonly uses RTP (reference [11]) as a transport layer together with UDP (reference [10]), regardless of which codecs are used at the application layer. SRTP (reference [12]) is an RTP protocol extension which can be used to encrypt RTP data streams.

For PDH, as in the E1/T1 voice systems (see references [3], [4], [5]), the authentication described in annex subsection B1.2 must be taken into account.

B1.2.2.2 Metadata Flags

Metadata flags, commonly used in the music industry (MPEG3 ID3V2.X), can be used for the description of voice data. Flags could indicate type of conversation (e.g., family or medical conference).

Another type of metadata is time stamps. The best way to do lip sync is to use a common time source for video and audio.

In general, the metadata flags must be protected like the rest of the voice data transmitted. The same requirements are applicable here.

B1.2.2.3 IP/Port Configuration

A default port number should be used rather than an arbitrary port assignment, e.g., Skype. If commercial applications are used (e.g., VSee or Skype), the installation of the commercial software must fulfill the security requirements of the respective space agency (e.g., NASA HQ Skype Configuration and Appropriate Use Guidelines). These sorts of applications are commonly used for public relations and educational events.

B1.2.3 Data Integrity

The second major area of security concern is data integrity. Data integrity requires that the audio not be altered, whether to introduce false audio or to interfere with decoding. Integrity verification mechanisms can be used for data validation to prevent security problems due to noncompliant or malignant data.

Like privacy, data integrity is commonly provided at other layers of the protocol stack or through a container format.

For VoIP data streams, the SRTP (reference [12]) is an RTP protocol extension which can be used to provide data integrity and replay protection.

For PDH using the E1/T1 voice communication systems, the transmission should be encrypted at least at the network transport layer (like in MPLS).

B1.2.4 Authentication of Communicating Entities

Source authentication is the requirement that the audio be attributable to a known origin. Closely related to data integrity, it can also be used for data validation.

Source authentication, if provided, is commonly provided at other layers of the protocol stack or through a container format.

For VoIP data streams, SRTP (reference [12]) is an RTP protocol extension which can be used to provide message authentication.

Using PDH systems, the source can be identified with the role. Each person and the assigned role is saved in an LDAP database or similar data bank, where the source can be identified.

The keyset can be identified as well, and because of the role, the manning roster or the shift plan allows identification of the person logged in.

B1.2.5 Control of Access to Resources

The fourth major area of security concern is access control, the requirement that the voice data be restricted from unauthorized further use by the recipient(s).

In an MCC, the user is identified by role, and the access and loop permissions are defined using X.500 (reference [28]). For users outside of an MCC (for supporting experiments or for special events) two-factor authentication and a VPN should be used to have acceptable access control.

To have access to a voice system, each user must have a user ID and a password and/or a token code. User login will give the user a predefined role with restricted permissions to their specific role.

Each role has listen, talk, or monitoring capabilities, and the role cannot be changed by the users themselves.

Access control can be provided at other layers of the protocol stack. Various methods also exist for attempting access control through the inclusion of proprietary metadata in a container format, although it should be noted that many similar techniques in the past have been defeated by determined attackers.

B1.2.6 Availability of Resources

Voice communication is the most essential resource in human space. The voice equipment must be maintained for emergency cases as well as for EVA and operations.

The access to a control room in an MCC must be restricted, and access to remote users must be controlled (see annex subsection B1.2.5. Access to voice equipment and the configuration management terminals must be under strict control.

Redundant equipment should be available in case of a hacker attack or failure, especially in space and in the MCC. Different communication channels and frequencies should be used in order to support communication in emergency cases (section 5).

User education is an important factor. Using VoIP implies many risks, as is described in this book. Regular user-awareness training should be done in a similar way to mandatory security training for Internet access, e-mails, phishing, etc.

B1.2.7 Auditing of Resource Usage

User access to voice recording and audio data should be logged using X.500 or other methods.

Auditing a voice system is quite easy. Each loop is recorded, and the system records the users logged in using the different loops. Doing a playback is possible to identify the source and recipients as described before. It is a common practice to identify problems by reviewing the loops to know who said what and when. That helps especially in time-critical situations to identify problems or errors, and to have lessons learned to improve the communications during operations.

Accessing audio files inside an MCC is easy to control because files are transmitted using FTP or other protocols requiring a user ID and password. Control room access is also

restricted; however, outside of an MCC, methods over which the MCC has no control could be used to copy files, e.g., a pen drive.

B1.3 POTENTIAL THREATS AND ATTACK SCENARIOS

Especially during public affairs events where external equipment is used, there is a risk of inadvertent or malicious content being injected into one or more voice loops, or of a loop's content being made public.

Another risk is that someone could gain entry to the voice system using someone else's role or credentials. This could cause confusion, damage, or loss of data.

B1.4 CONSEQUENCES OF NOT APPLYING SECURITY TO THE TECHNOLOGY

If confidentiality is not implemented, audio communications might be heard by unauthorized entities, resulting in disclosure of sensitive or private information.

Without source authentication or integrity verification, valid audio communications could be corrupted, or invalid audio could be substituted in its place.

Without access controls, authorized entities might be able to redistribute sensitive or proprietary information to unauthorized third parties.

B2 SANA CONSIDERATIONS

The recommendations of this document do not require any action from SANA.

NOTE – The systems described in this document are either standalone or connect to existing communications facilities. Those communication facilities, however, would require a SANA registry entry. Motion Imagery is another service among many utilizing a given spacecraft's communication system and therefore does not require an additional SANA entry.

B3 PATENT CONSIDERATIONS

It is expected that implementation of this Recommended Standard by space-agency users will occur through the use of commercial off-the-shelf equipment that implements the referenced standards, and that patent-rights issues for such equipment will have been settled between the equipment manufacturer and the patent-right holders. It is not expected that space-agency users will develop new equipment based on the standards referenced herein. Therefore patent rights for the referenced standards are outside the scope of this Recommended Standard.

ANNEX C

ACRONYMS AND ABBREVIATIONS

(INFORMATIVE)

<u>Term</u>	<u>Meaning</u>
AAC	Advance Audio Coding/Codec
ADF	automatic direction finder
AFRC	NASA Armstrong Flight Research Center
AFRCC	Air Force Rescue Coordination Center
AOS	Advanced Orbiting Systems
API	application programming interface
ASR	automatic speech recognition
ATM	asynchronous transfer mode
BW	bandwidth
CCSDS	Consultative Committee for Space Data Systems
CFDP	CCSDS File Delivery Protocol
codec	code-decode
dBFS	decibels relative to full scale
DDMS	U.S. Department of Defense Manned Space Flight Support Office
DFRS	NASA's Dryden Flight Research Center (officially redesignated as the NASA Armstrong Flight Research Center [AFRC])
DoD	U.S. Department of Defense
DTN	Delay-Tolerant Networking
EMU	extravehicular mobility unit
ENCAP	CCSDS Encapsulation Service

RECOMMENDED STANDARD FOR VOICE AND AUDIO COMMUNICATIONS

EVA	extra vehicular activities
FD	flight director
FL/RL Comms	wide band TDM communication equipment (forward link / return link)
FTP	File Transfer Protocol
GPS	Global Positioning System
HOSC	Huntsville Operations Support Center
ICD	interface control document
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISS	International Space Station
ITU-T	International Telecommunication Union-Telecommunication Standardization Sector
IVoDS	Internet Voice Distribution System
LDAP	Lightweight Directory Access Protocol
LSO	Landing Support Officer
LVoDS	local IVoDS
MCC	mission control center
MCC-H	Mission Control Center-Houston
MCC-M	Mission Control Center-Moscow
MGCP	Media Gateway Control Protocol
MOC	mission operations center
MOVE	Mission Operations Voice Enhancement
MPLS	Multiprotocol Label Switching
MSFC	NASA Marshall Space Flight Center
NAT	network address translation

RECOMMENDED STANDARD FOR VOICE AND AUDIO COMMUNICATIONS

PABX	private automatic branch exchange
PCM	pulse code modulation
PDH	plesiochronous digital hierarchy
PIP	private Internet Protocol
POIC	Payload Operations Integration Center
PSN	packet switched network
PSTN	public switched telephone network
PTT	push-to-talk
PW	Pseudowire
QoE or QX	quality of experience
QoS	quality of service
RF	radio frequency
RFC	Request for Comment
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
RVoDS	remote IVoDS
S/G	space-to-ground
SAR	search and rescue
SHO	scheduled order
SIP	Session Initiation Protocol
SRTCP	Secure Real-time Transport Control Protocol
SRTP	Secure Real-time Transport Protocol
T/L/M	talk/listen/monitor
TCP	Transmission Control Protocol
TCP/IP	Transmission Control Protocol/Internet Protocol

RECOMMENDED STANDARD FOR VOICE AND AUDIO COMMUNICATIONS

TDM	time division multiplexing
TDMoIP	TDM over IP
TDRSS	Tracking and Data Relay Satellite System
TOPO	Trajectory Operations Officer
TTS	text-to-speech
UDP	User Datagram Protocol
UHF	ultra high frequency
VCA	Virtual Channel Access
VHF	very high frequency
VoIP	voice over IP
VoIPSA	Voice over IP Security Alliance
VOX	Voice Operated eXchange (aka voice operated switch, voice level detection, or voice activity level)
VPN	virtual private network
WSC	White Sands Complex

ANNEX D**DTN BUNDLE PROTOCOL FOR VOICE AND AUDIO
TRANSMISSION****(INFORMATIVE)****D1 MODE 1: REAL TIME DELIVERY, BEST EFFORT**

Mode 1 provides best-effort delivery of data and should be used for unicast or multicast data delivery when the data is to be utilized immediately upon arrival (perhaps for near-real-time decision making). The data will be forwarded using the DTN Bundle Protocol without custody transfer and will be available to the receiving codec/application on a best-effort basis. Therefore the data will be ‘played’ in the order it is received. Missing, damaged, or out-of-order packets will be ignored with subsequent codec responses being typical. Additional items such as excessive latency or jitter may be deleterious to the codec response.

D2 MODE 2: DELAYED DELIVERY, WITH PACKET RESTORATION

Mode 2 provides delivery of data with the prospect of no packets dropped and should be used for single or multi-transport of data when the data can be utilized after a sufficient latency to allow reassembly of the data. The data will be available to a codec/application once the underlying protocol has sufficient time to reassemble the data stream.

Therefore the data will be ‘played’ in its entirety subject to data being wholly lost on a frame basis. Missing and damaged bundles will be forwarded prior to reassembly with out-of-order bundles to be made available to the end codec/application. Additional items such as excessive latency or jitter will not affect codec/application response.

D3 STORE AND FORWARD ISSUES

The Delay Tolerant Networking Bundle Protocol (DTN-BP) enables store and forward operation when multiple hops or links are required to send data from source to receiver. There is an additional space link reliability protocol available known as Licklider Transmission Protocol (LTP), which will note loss of DTN bundles (which would contain 1– n frames depending on system design) and request retransmission of missing bundles.

Use of the LTP reliability mechanism will result in delivery of out-of-order audio frames to the application layer (because of the lag in retransmitting missing bundles). This may or may not be acceptable to the end-user application, and consideration of this out-of-order condition must be made.

In current lab experiments, it has been seen that an effective use of DTN to transmit audio is to transmit bundles unreliably, accept the consequences of missing data for the real-time

audio stream, and use another mechanism for recovery of missing data. This mechanism could be an application buffer to store retransmitted bundles separately for later reconstruction of the audio stream in non-real time, or to record the audio on board and later transmit as an audio file for playback after the file is completely received without errors. Further experimentation is required to provide definite parameters for the transmission of voice and audio files using DTN.¹

¹ As this is refined, it will be added to the voice and audio specifications.

ANNEX E

INFORMATIVE REFERENCES

(INFORMATIVE)

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- [E3] *Information Technology—Generic Coding of Moving Pictures and Associated Audio Information—Part 3: Audio*. 2nd ed. International Standard, ISO/IEC 13818-3:1998. Geneva: ISO, 1998.
- [E4] *Digital Audio Interface—Part 3: Consumer Applications*. Edition 3.1 (2009-12-10). IEC 60958-3:2006+AMD1:2009 CSV. Geneva: IEC, 2009.
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- [E8] *Perceptual Evaluation of Speech Quality (PESQ): An Objective Method for End-to-End Speech Quality Assessment of Narrow-Band Telephone Networks and Speech Codecs*. ITU-T Recommendation P.862. Geneva: ITU, 2001.