Mars Microphone Instrument Users Manual
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1. Instrument Overview

The Mars Microphone (MM) is a small (50 g) instrument to sample sounds on Mars. It forms a part of the Russian LIDAR instrument, and was developed at the University of California at Berkeley Space Sciences Laboratory under funding from the Planetary Society. It will fly on the NASA Mars 98 lander mission.

Power and telemetry resources are extremely limited, so only a few short sounds will be sent. These sounds will be selected by a integrated loudness criteria : the sound sample with the largest sum of amplitudes (over the sound sample), collected during the time required to transmit the previous sound sample, will be saved and transmitted next. Flash memory in the MM allows these sounds to be collected, saved, and transmitted over a period of days, during which power will be cycled on and off.

In addition to these short sound samples, a more continuous record of sound levels will be collected in the form of integrated power in 6 filter bands. These filters are integrated for a programmable interval from 1 second to 10 minutes (longer integrations give more continuous time coverage with less telemetry, but with lower time resolution). Five of the six filters divide the sound spectrum, each filter covering about an octave. The sixth filter gives the total power in the instrument pass band.

A complete description of the MM to LIDAR electrical and mechanical interface can be found in the "Mars Microphone to LIDAR Interface Specification Document" (David Curtis, U.C.Berkeley)

2. Instrument Modes

The instrument operation is controlled by an 8 bit mode word. This mode word is divided into a number of fields which control different operations in the instrument. Bit zero if the least significant bit, and bit 7 is the most significant bit.

Bits	Name	Function
0,1	Gain	These bits select the amplifier gain. 00 is the lowest gain (1x) 01 is 4x 10 is 16x 11 is 64x
2	Sample Rate	This sets the bandwidth both for the filters and the sampling. 0 is LF (Low Frequency) 1 is HF (High Frequency)
3,4	Filter Rate	This sets the filter sample rate.

		00 is 10 minute accumulation 01 is 1 minute accumulation 10 is 10 second accumulation 11 is 1 second accumulation
5	Telem. Mode	This sets the maximum rate that filter packets can be sent. 0 is 50% filter packets 1 is 10% filter packets
6	Comp. Bits	This sets the sample compression mode 0 is 2-bit compression 1 is 4-bit compression
7	Silence	This sets the silence compression mode 0 is disabled 1 is enabled

Telemetry Mode sets the maximum filter packet read-out rate. If packet collection is fast enough that a filter packet is not yet ready when it is time to send a filter packet by this scheme, then a sample packet will be sent instead. Filter packet collection is double-buffered, so that when one packet completes, collection starts on the second. If both packets complete before a filter packet has been read out, filter data collection is suspended until a packet is telemetered.

Sample Rate sets the sample rate and filter bandwidths (3db):

Mode:	LF	HF
SampleRate:	5 KHz	20 KHz
Sample Time:	10.6sec	2.7sec
Filter 0:	90-2000 Hz	DC-10 KHz
Filter 1:	50-200 Hz	250-500 Hz
Filter 2:	110-350 Hz	500-1000 Hz
Filter 3:	250-700 Hz	1-2.5 KHz
Filter 4:	450-1400 Hz	2.5-5 KHz
Filter 5:	700-2500 Hz	5-10 KHz

3. Instrument Power-On

At power-on, the instrument checks the Flash memory contents. If a Shut-Down command was sent within 10 seconds before the power-off, then the instrument status will be in Flash, including any collected Filter and Sample packets. This is called a Warm Start. If Shut-Down was not sent before the last instrument power-down, the Flash will contain no information. This is called a Cold Start. The instrument is set to its default mode on a Cold Start (Mode 0), and the sound sample and filter data memories are empty. A Time command should be sent in the first 10 seconds after every power-on so that collected data will be correctly time-stamped. The instrument waits up to 10 seconds for this command before starting to collect data.

4. Instrument Telemetry

Instrument telemetry is generated in 2047 byte packets. These packets are collected periodically from the instrument. The first 5 bytes and last byte contain the same information for all packet types:

Byte 0:	Start Sync, AA (hexadecimal) fixed value
Byte 1:	Packet sequence number (reset to zero by a Cold Start, increments each packet.
Byte 2:	Instrument mode (at packet transmit time)
Byte 3:	Error code (see below)
Byte 4:	Reset Counter (number of resets since last Cold Start)
Byte 5:	Packet Type (see below)
Byte 2046:	End Sync, BB (hexadecimal) fixed value

The Error code byte contains two fields:

MSB							LSB
7	6	5	4	3	2	1	0
< Error Type			>	< Error Count			>

Error Count is the number of errors that have occurred since the last Cold Start. Error Type is a code for the latest error:

Error Type	Meaning
0	No error
1	Bad Command Received
2	Flash memory write failure
3	Flash memory access time-out
4	Flash memory erase failure
5	No time code received in first 10 seconds after power-on

Packet Type	Contents
1	Filter Packet
2	Sample Packet
3	Diagnostic Packet

The contents of the remaining packet bytes depends on the packet type, as described below.

4.1 Filter Packets

These packets contain the samples from the filter accumulators. Each packet contains one or more blocks of contiguous samples. A new block is started at the beginning of a new packet, when the instrument mode has changed, or on a warm-start. Each block starts with a header which has the time, mode, and sample count, followed by the filter accumulations.

Filter Block Header Format

Byte of Block	Contents
0,1	Sample Count (LSB first)
2	Instrument mode during collection
3,4,5	Time code at start of first filter sample accumulation (LSB first)

Following the block header are 6 bytes per filter accumulation sample, one byte per filter accumulation, ordered by filter number (see section 2).

Filter accumulations are the sum of the absolute values of the sample values (each sample value is in the range -32767 to +32767). The filter sample rate is fixed at 20 KHz, independent of the instrument mode sample rate. The accumulation includes all samples in the accumulation interval as set in the instrument mode Filter Rate (1 sec, 10 sec, 1 min, 10 min). Upon completion the filter accumulations are divided by a number which depends on the instrument mode Filter Rate:

Filter Rate	Filter Divisor
00 (10 min)	1048576
01 (1 min)	131072
10 (10 sec)	16384
11 (1 sec)	2048

After dividing the filter accumulations, they are log-compressed to one byte per sample using the table in Appendix A. The hexadecimal byte value is the sum of the row and column labels for a value greater or equal to the number in the array, but less than the next entry. For example, if the compressed value is 55 hex, then the divided filter accumulator was between 336 and 351 inclusive.

4.2 Sample Packets

A sound sample is compressed into one or more packets of data, depending on the instrument mode and the sound data. Each sample packet contains a header with the following information:

Packet Byte	Contents
6	Instrument mode during sound sample collection
7	Sound sample packet sequence number
8,9,10	Time at end of sound sample (LSB first)

Sound data is compressed in groups of up to 50 19 sample blocks. Each packet starts with a new group. Each group starts with a group header byte.

Group header byte value	Meaning
0	End of sound sample (remaining packet bytes

	diagnostic data - see below)
255	End of packet (remaining packet bytes unused)
1-50	Sound sample group block count
129-178	Silence group count +128

A silence group indicates a group all of whose values are below the silence threshold. No data is sent for this group beyond the group header byte. The group represents 19 times the group block count (Group header byte value minus 128) samples of silence. Silence groups should only occur if the Silence bit is set in the instrument mode.

Sound sample group header byte is followed by a gain offset byte, and then group block count (Group header byte value) 19 sample blocks. The 19 sample blocks are compressed to either 10 or 6 bytes each, depending on the Comp Bits field in the instrument mode (a zero value corresponds to 2 bit compression, which gives 6 bytes per 19 samples, while a one indicates 4 bit compression, with 10 bytes per 19 samples).

4.2.1 4-Bit Sample Decompression

Each 19 sample block of 10 bytes is coded as follows:

	MSB 7 6 5 4	3 2 1 0 LSB
Byte 0	< Sample 0 >	< Gain >
Byte 1	< Sample 2 >	< Sample 1 >
Byte 2	< Sample 4 >	< Sample 3 >
Byte 3	< Sample 6 >	< Sample 5 >
Byte 4	< Sample 8 >	< Sample 7 >
Byte 5	< Sample 10 >	< Sample 9 >
Byte 6	< Sample 12 >	< Sample 11 >
Byte 7	< Sample 14 >	< Sample 13 >
Byte 8	< Sample 16 >	< Sample 15 >
Byte 9	< Sample 18 >	< Sample 17 >

4.2.2 2-Bit Sample Decompression

Each 19 sample block of 6 bytes is coded as follows:

	MSB 7 6	5 4	3 2	1 0 LSB
Byte 0	< Sample 0 >		< Gain >	
Byte 1	<Sample 2 >	<Sample 1>	< Table >	
Byte 2	<Sample 6 >	<Sample 5>	<Sample 4 >	<Sample 3 >
Byte 3	<Sample 10>	<Sample 9>	<Sample 8 >	<Sample 7 >

Byte 4	<Sample 14><Sample 13> <Sample 12><Sample 11>
Byte 5	<Sample 18><Sample 17> <Sample 16><Sample 15>

The algorithms for both the 2 and 4 bit compression schemes are proprietary techniques from Sensory Inc., a privately held start-up company specializing in speech recognition technology. For more information visit their website at:

http://www.sensoryinc.com