Tait DMR and TaitNet MPT-IP Voice Recorder Protocol

Integration Guide

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CONTENTS

PREFACE

Purpose

The purpose of this document is to provide information on how to integrate a voice recorder to a Tait DMR or TaitNet MPT-IP network using VRP (Voice Recorder Protocol), a Tait proprietary protocol.

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Scope

This document applies to Tait DMR trunking (Tier 3) networks, Tait DMR conventional (Tier 2) networks and TaitNet MPT-IP networks. The differences, if any, relating to each network type are specified in the document.

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1	April 2019	Creation based on VRP standard from S. Whelan/M. Dean

Terms and Abbreviations

Term	Definition
AMBE	Advanced multiband excitation vocoder
CSRC	Contributing source
DMR	Digital Mobile Radio - an open radio standard
MPT 1327	Industry standard for trunked radio communications networks
MSB	Most significant bit
PT	Payload type
RFC	Request For Comments
RTP	Real Time Protocol
SSRC	Synchronization source
VoIP	Voice over IP - a technology that allows telephone calls to be made over computer networks such as the Internet. VoIP converts analog voice signals into digital data packets and supports real-time, two-way transmission of conversations using Internet Protocol (IP).
VRP	Tait proprietary Voice Recorder Protocol

1. OVERVIEW

Tait DMR and MPT-IP controllers have a dedicated voice recorder interface available. It allows for the recording of all audio streams going through the controller. Up to two voice recorders can be connected to the controller for redundancy purposes.

Voice recorder applications should distinguish between group and individual calls.

Voice recorder applications should indicate the priority level of a call, particularly for emergency calls.

Refer to the respective system manuals for more information about voice recorder and controller configuration:

- Tait TN9300 DMR Trunked System Manual, MNB-00003-xx
- Tait TN9300 DMR Conventional System Manual, MNB-00005-xx
- TaitNet MPT-IP System Manual, MNA-00026-xx

2. PROTOCOL

2.1 Connection

The Voice Recorder Protocol is a connectionless protocol. As soon as the IP addresses of the voice recorders are entered and saved into the controller configuration, the controller starts sending voice packets to the voice recorder(s).

The system manuals contain the UDP port used for VRP.

2.2 General

VRP version is 2.0.

VRP packets conform to RTP RFC 3550 and RTP Header Extensions RFC 5285, except where explicitly stated in this document.

The VRP packets have the following RTP characteristics:

- Payload type (PT) is 0 for MPT-IP G.711 (µlaw) and 100 for DMR AMBE+2
- Sequence numbers may be initialized to 0 at the beginning of each call but they may be random as specified in the RFC, depending on the source of the audio. Each logical channel has its own sequence number.
- The synchronization source (SSRC) will change every over
- Contributing source (CSRC) is not used and is set to 0
- All integers are in network byte order, most significant bit (MSB) first

2.3 Usage

The radio network may handle many simultaneous calls. Each of these calls (group or individual) will result in a separate VRP stream, each with a different UUID (see 2.4). When a call is created, the radio network sends a VRP packet to the voice recorder to indicate the start of a call. This packet has no audio payload. The call state flag is set to 'Call Start'.

When audio is transmitted by the radio network, a copy of it is always sent to the voice recorder in a VRP packet. The Call state is set to 'No Change'. The Source Unit field contains the address of the unit that transmitted this audio. If the transmitter was unable to be determined, the value will be 0. Once the address of the transmitting unit has been determined, this value will be changed to the corresponding address.

No VRP packets are sent between voice transmissions.

The source of audio for the stream can change at any time as other sources override the stream. Either the change of source or the start of a new over will result in a change in SSRC, and possible reset in the Timestamp field in the RTP header.

Voice Recorder applications should use the time elapsed between packets being received to calculate the amount of silence to insert into the recorded stream. Silence could also be inserted during playback.

Voice Recorder applications should not rely on changes in the RTP timestamp value to calculate the amount of silence to insert into the stream.

At the end of the call the radio network will send a VRP packet that consists of a Call state flag on this set to 'Call End'. This packet has no audio payload. This packet may be sent many seconds after the last packet containing audio, if the call was allowed to timeout on the radio network.

2.4 VRP Header Packet

The format is as follows:

<table>
<tr><th colspan="12">RTP with Extension Header</th></tr>
<tr><th>bit offset</th><th>0-1</th><th>2</th><th>3</th><th>4-7</th><th>8</th><th>9-11</th><th>12-15</th><th>16-19</th><th>20-23</th><th>24-27</th><th>28-31</th></tr>
<tr><td>0</td><td>V</td><td>P</td><td>X</td><td>CC</td><td>M</td><td colspan="2">PT</td><td colspan="4">Sequence number</td></tr>
<tr><td>32</td><td colspan="11">Timestamp</td></tr>
<tr><td>64</td><td colspan="11">SSRC identifier</td></tr>
<tr><td>96</td><td colspan="7">Extension header ID</td><td colspan="4">Extension header length</td></tr>
<tr><td>128</td><td colspan="11">Called address</td></tr>
<tr><td>160</td><td colspan="11">Caller address</td></tr>
<tr><td>192</td><td colspan="11">Source unit address</td></tr>
<tr><td>224</td><td colspan="11">Source channel</td></tr>
<tr><td>256</td><td colspan="3">Call type</td><td>Call state</td><td colspan="3">Call flags</td><td colspan="2">RSSI</td><td colspan="2">BER/SINAD</td></tr>
<tr><td>288</td><td colspan="11">UUID 0-31</td></tr>
<tr><td>320</td><td colspan="11">UUID 32-63</td></tr>
<tr><td>352</td><td colspan="11">UUID 64-95</td></tr>
<tr><td>384</td><td colspan="11">UUID 96-127</td></tr>
<tr><td>416</td><td colspan="7">Reserved</td><td colspan="2">Encryption method</td><td colspan="2">Key ID</td></tr>
<tr><td>448</td><td colspan="11">Initialization vector</td></tr>
</table>

Figure 1: VRP Header Format

Field name	Size	Value	Description
V (version)	2 bits	2	Version number of RTP protocol (version 2)
P (padding)	1 bit	0	If 1, indicates there are extra padding bytes at the end of the RTP packet
X (extension)	1 bit	1	1 indicates presence of a *Profile extension header* between standard header and payload data
CC (CSRC count	4 bits	0	The number of CSRC identifiers that follow the fixed header (not used with MPT-IP AIS RTP = 0)
M (marker)	1 bit	0	If 1, means that the current data has some special relevance for the application
PT (payload type)	7 bits	0	Indicates the format of the payload and determines its interpretation by the application. For G.711 μ-law, PT = 0. For DMR PT=100
Sequence number	2 bytes	varies	Incremented by one for each RTP data packet sent and is to be used by the receiver to detect packet loss and to restore packet sequence
Timestamp	4 bytes	varies	Used to enable the receiver to play back the received samples at appropriate intervals. The granularity of the timing is application specific. When several media streams are present, the timestamps are independent in each stream, and should not be relied upon for media synchronization.
SSRC	4 bytes	varies	Synchronization source identifier identifies a stream of packets. A randomly assigned value is generated for each 'over' and remains the same for every packet in the 'over'. The random 32-bit number reduces the chance of two streams having the same SSRC value.
Extension header ID	2 bytes	40961 (decimal) A001 (hex)	Identifies the header extension defined in this document
Extension header length	2 bytes	11	Total number of 4-byte "words" in the header extensions (but excluding *this* word) =11
Called address	32 bits	varies	DMR raw number corresponding to the called unit or group (as explained below)
Caller address	32 bits	varies	DMR raw number corresponding to the calling unit or group (as explained below)
Source unit address	32 bits	varies	DMR raw number corresponding to the transmitting party (as explained below)
Source channel	4 bytes	varies	The ID of the device that generated this packet originally
Call type	4 bits	varies	0: Individual Voice 1: Group Voice

Field name	Size	Value	Description
Call state	4 bits	varies	0: No state change 1: Start of call 2: End of call This value is set by the controller. Other devices generating VRP packets should set this value to 0.
Call flags	8 bits	varies	DMR only => bits 0-2: Call Priority (3 being the highest) MPT-IP only => bit 0: High Priority Call bit 3: Broadcast Call bit 7: Emergency Call
RSSI	8 bits	varies	Devices that cannot provide this information should set this value to 0. RSSI is the **signed** signal strength reported by the base station, to the nearest dB.
BER/SINAD	8 bits	varies	Devices that cannot provide this information should set this value to 0. This is only relevant for MPT-IP: this corresponds to the SINAD level reported by the base station, to the nearest dB.
UUID	128 bits	varies	The UUID uniquely identifies this call across all controllers for the life of the system. Can be used to retrieve call records. This value is set by the controller. Other devices generating VRP packets set this value to 0.
Encryption Method	8 bits	varies	0: no encryption
Key ID	1 byte	varies	Fixed size input of the encryption algorithm. Not used.
Initialization vector	4 bytes	varies	Fixed size input of the encryption algorithm. Not used.

Table 1: VRP Header Fields

The 32-bit addresses contain the 24-bit DMR raw number. This means that the 8 most significant bits (MSB) are to be discarded. The DMR raw number can be converted to MPT1327, MPT1343, ANN or DMR standards addresses. The conversion methods are available in the respective system manuals. For conventional networks, or trunked networks using DMR Native Dialing, the voice recorder should display the raw DMR address. For other trunked networks, a voice recorder will need to be configured with the same fleet configuration information as the radio network in order to be able to translate the DMR raw number to the correct fleet address. Note that different fleets may use different numbering formats, refer to the relevant trunked system manual for further details. In either case, aliases could be displayed instead of radio and group IDs.

2.5 Payload

G.711 payload consists of 20ms of audio. This corresponds to 160 Bytes of audio.

AMBE payload contains three 20ms voice code words of error corrected audio, plus any embedded signaling or SYNC data. The voice code words are accompanied by FEC result bits which are used to assist the voice decoder to produce nice voice in the presence of errors or missing frames. Each voice codeword comprises 49 data bits plus additional 8 bits FEC information as follows (for a total of 24 Bytes):

<table>
<tr><td colspan="8">0 MSB</td><td colspan="2"></td><td colspan="10">1</td><td colspan="10">2</td><td colspan="2">3</td></tr>
<tr><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>0</td><td>1</td></tr>
<tr><td colspan="8">VCW 0 FEC Info</td><td colspan="7" rowspan="2">Spare 0</td><td colspan="17" rowspan="2">VCW 0 b (48-32)</td></tr>
<tr><td colspan="3">TTL</td><td colspan="3">CW</td><td>M</td><td>L</td></tr>
<tr><td colspan="32">VCW 0 b (31-0)</td></tr>
<tr><td colspan="8">VCW 1 FEC Info</td><td colspan="7" rowspan="2">Spare 1</td><td colspan="17" rowspan="2">VCW 1 b (48-32)</td></tr>
<tr><td colspan="3">TTL</td><td colspan="3">CW</td><td>M</td><td>L</td></tr>
<tr><td colspan="32">VCW 1 b(31-0)</td></tr>
<tr><td colspan="8">VCW 2 FEC Info</td><td colspan="7" rowspan="2">Spare 2</td><td colspan="17" rowspan="2">VCW 2 b (48-32)</td></tr>
<tr><td colspan="3">TTL</td><td colspan="3">CW</td><td>M</td><td>L</td></tr>
<tr><td colspan="32">VCW 2 b (31-0)</td></tr>
</table>

Figure 2: AMBE+2 Generic Voice Burst

Mnemonic	Bits	'C' I/F Name	Description
TTL	3	ttl_errs	Total number of errors detected
CW	3	cw_errs[1]	Number of errors in the Golay Codeword
L	1	LOST_FRAME	Indicates that a voice frame was lost, the vocoder predicts missing voice
M	1	CNI_FRAME	Indicates that the voice frame should be muted and replaced by comfort noise

Table 2 AMBE+2 Voice FEC Information