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RTP Payload Format for Tactical Secure Voice Cryptographic
Interoperability Specification (TSVCIS) Codec

Abstract

This document describes the RTP payload format for the Tactical Secure Voice Cryptographic Interoperability Specification (TSVCIS) speech coder. TSVCIS is a scalable narrowband voice coder supporting varying encoder data rates and fallbacks. It is implemented as an augmentation to the Mixed Excitation Linear Prediction Enhanced (MELPe) speech coder by conveying additional speech coder parameters to enhance voice quality. TSVCIS augmented speech data is processed in conjunction with its temporally matched Mixed Excitation Linear Prediction (MELP) 2400 speech data. The RTP packetization of TSVCIS and MELPe speech coder data is described in detail.

Status of This Memo

This is an Internet Standards Track document.

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1. Introduction

This document describes how compressed Tactical Secure Voice Cryptographic Interoperability Specification (TSVCIS) speech as produced by the TSVCIS codec [TSVCIS] [NRLVDR] may be formatted for use as an RTP payload. The TSVCIS speech coder (or TSVCIS speech-aware communications equipment on any intervening transport link) may adjust to restricted bandwidth conditions by reducing the amount of augmented speech data and relying on the underlying MELPe speech coder for the most constrained bandwidth links.

Details are provided for packetizing the TSVCIS augmented speech data along with MELPe 2400 bps speech parameters in an RTP packet. The sender may send one or more codec data frames per packet, depending on the application scenario or based on transport network conditions, bandwidth restrictions, delay requirements, and packet loss tolerance.

1.1. Conventions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

Best current practices for writing an RTP payload format specification were followed [RFC2736] [RFC8088].

1.2. Abbreviations

The following abbreviations are used in this document.

AVP:	Audio/Video Profile
AVPF:	Audio/Video Profile Feedback
CELP:	Code-Excited Linear Prediction
FEC:	Forward Error Correction
LPC:	Linear-Predictive Coding
LSB:	Least Significant Bit
MELP:	Mixed Excitation Linear Prediction
MELPe:	Mixed Excitation Linear Prediction Enhanced
MSB:	Most Significant Bit
MTC:	Modified Count
NATO:	North American Treaty Organization
NRL:	Naval Research Lab

PLC: Packet Loss Concealment

SAVP: Secure Audio/Video Profile

SAVPF: Secure Audio/Video Profile Feedback

SDP: Session Description Protocol

SSRC: Synchronization Source

SRTP: Secure Real-Time Transport Protocol

TSVCIS: Tactical Secure Voice Cryptographic Interoperability Specification

VAD: Voice Activity Detect

VDR: Variable Date Rate

2. Background

The MELP speech coder was developed by the US military as an upgrade from the LPC-based CELP standard vocoder for low-bitrate communications [MELP]. ("LPC" stands for "Linear-Predictive Coding", and "CELP" stands for "Code-Excited Linear Prediction".) MELP was further enhanced and subsequently adopted by NATO as "MELPe" for use by its members and Partnership for Peace countries for military and other governmental communications as international NATO Standard STANAG 4591 [MELPE].

The Tactical Secure Voice Cryptographic Interoperability Specification (TSVCIS) is a specification written by the Tactical Secure Voice Working Group (TSVWG) to enable all modern tactical secure voice devices to be interoperable across the US Department of Defense [TSVCIS]. One of the most important aspects is that the voice modes defined in TSVCIS are based on specific fixed rates of the Naval Research Lab's (NRL's) Variable Date Rate (VDR) Vocoder, which uses the MELPe standard as its base [NRLVDR]. A complete TSVCIS speech frame consists of MELPe speech parameters and corresponding TSVCIS augmented speech data.

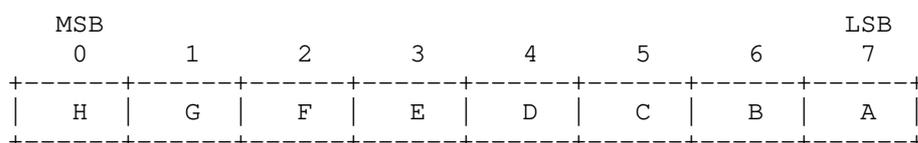
In addition to the augmented speech data, the TSVCIS specification identifies which speech coder and framing bits are to be encrypted and how they are protected by forward error correction (FEC) techniques (using block codes). At the RTP transport layer, only the speech coder-related bits need to be considered and are conveyed in unencrypted form. In most IP-based network deployments, standard link encryption methods (Secure Real-Time Transport Protocol (SRTP), VPNs, FIPS 140 link encryptors, or Type 1 Ethernet encryptors) would be used to secure the RTP speech contents.

TSVCIS augmented speech data is derived from the signal processing and data generated by the MELPe speech coder. For the purposes of this specification, only the general parameter nature of TSVCIS will be characterized. Depending on the bandwidth available (and FEC requirements), a varying number of TSVCIS-specific speech coder parameters need to be transported. These are first byte-packed and then conveyed from encoder to decoder.

Byte packing of TSVCIS speech data into packed parameters is processed as per the following example, where

Three-bit field: Bits A, B, and C (A is MSB; C is LSB)

Five-bit field: Bits D, E, F, G, and H (D is MSB; H is LSB)



600 bps	0	1	N/A	7
Comfort Noise	1	0	1	2
TSVCIS Data	1	1	N/A	var.

Table 1: TSVCIS/MELPe Frame Bitrate Indicators and Frame Length

The total number of bits used to describe one MELPe frame of 2400 bps speech is 54, which fits in 7 octets (with two rate code bits). For MELPe 1200 bps speech, the total number of bits used is 81, which fits in 11 octets (with three rate code bits and four unused bits). For MELPe 600 bps speech, the total number of bits used is 54, which fits in 7 octets (with two rate code bits). The comfort noise frame consists of 13 bits, which fits in 2 octets (with three rate code bits). TSVCIS packed parameters will use the last code combination in a trailing byte as discussed in Section 3.2.

It should be noted that CODB for MELPe 600 bps mode MAY deviate from the value in Table 1 when bit 55 is used as an alternating 1/0 end-to-end framing bit. Frame decoding would remain distinct as CODA being zero on its own would indicate a 7-byte frame for either a 2400 or 600 bps rate, and the use of 600 bps speech coding could be deduced from the RTP timestamp (and anticipated by the Session Description Protocol (SDP) negotiations).

3.1.1. 2400 bps Bitstream Structure

The 2400 bps MELPe RTP payload is constructed as per Figure 2. Note that CODA MUST be filled with 0 and CODB SHOULD be filled with 0 as per Section 3.1. CODB MAY contain an end-to-end framing bit if required by the endpoints.

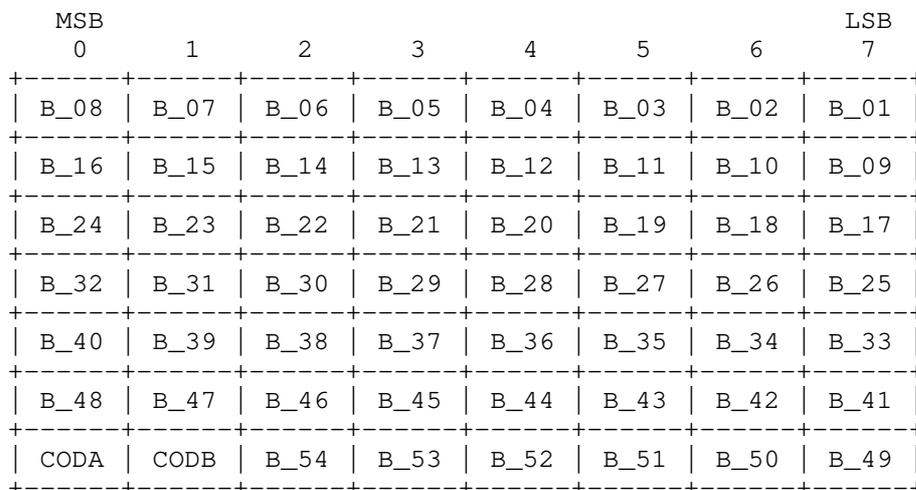
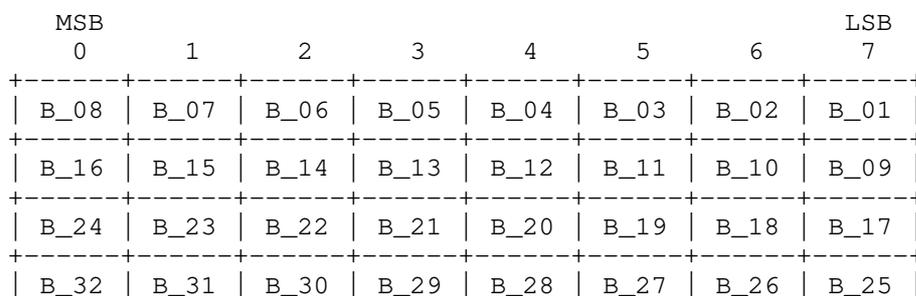


Figure 2: Packed MELPe 2400 bps Payload Octets

3.1.2. 1200 bps Bitstream Structure

The 1200 bps MELPe RTP payload is constructed as per Figure 3. Note that CODA, CODB, and CODC MUST be filled with 1, 0, and 0, respectively, as per Section 3.1. RSV0 MUST be coded as 0.



B_40	B_39	B_38	B_37	B_36	B_35	B_34	B_33
B_48	B_47	B_46	B_45	B_44	B_43	B_42	B_41
B_56	B_55	B_54	B_53	B_52	B_51	B_50	B_49
B_64	B_63	B_62	B_61	B_60	B_59	B_58	B_57
B_72	B_71	B_70	B_69	B_68	B_67	B_66	B_65
B_80	B_79	B_78	B_77	B_76	B_75	B_74	B_73
CODA	CODB	CODC	RSV0	RSV0	RSV0	RSV0	B_81

Figure 3: Packed MELPe 1200 bps Payload Octets

3.1.3. 600 bps Bitstream Structure

The 600 bps MELPe RTP payload is constructed as per Figure 4. Note CODA MUST be filled with 0 and CODB SHOULD be filled with 1 as per Section 3.1. CODB MAY contain an end-to-end framing bit if required by the endpoints.

MSB							LSB
0	1	2	3	4	5	6	7
B_08	B_07	B_06	B_05	B_04	B_03	B_02	B_01
B_16	B_15	B_14	B_13	B_12	B_11	B_10	B_09
B_24	B_23	B_22	B_21	B_20	B_19	B_18	B_17
B_32	B_31	B_30	B_29	B_28	B_27	B_26	B_25
B_40	B_39	B_38	B_37	B_36	B_35	B_34	B_33
B_48	B_47	B_46	B_45	B_44	B_43	B_42	B_41
CODA	CODB	B_54	B_53	B_52	B_51	B_50	B_49

Figure 4: Packed MELPe 600 bps Payload Octets

3.1.4. Comfort Noise Bitstream Definition

The comfort noise MELPe RTP payload is constructed as per Figure 5. Note that CODA, CODB, and CODC MUST be filled with 1, 0, and 1, respectively, as per Section 3.1.

MSB							LSB
0	1	2	3	4	5	6	7
B_08	B_07	B_06	B_05	B_04	B_03	B_02	B_01
CODA	CODB	CODC	B_13	B_12	B_11	B_10	B_09

Figure 5: Packed MELPe Comfort Noise Payload Octets

3.2. TSVCIS Bitstream Definition

The TSVCIS augmented speech data as packed parameters MUST be placed immediately after a corresponding MELPe 2400 bps payload in the same RTP packet. The packed parameters are counted in octets (TC). The preferred placement SHOULD be used for TSVCIS payloads with TC less than or equal to 77 octets; this is shown in Figure 6. In the preferred placement, a single trailing octet SHALL be appended to include a two-bit rate code, CODA and CODB (both bits set to one), and a six-bit modified count (MTC). The special modified count value

of all ones (representing an MTC value of 63) SHALL NOT be used for this format as it is used as the indicator for the alternate packing format shown next. In a standard implementation, the TSVCSIS speech coder uses a minimum of 15 octets for parameters in octet packed form. The modified count (MTC) MUST be reduced by 15 from the full octet count (TC). Computed MTC = TC-15. This accommodates a maximum of 77 parameter octets (the maximum value of MTC is 62; 77 is the sum of 62+15).

	MSB				LSB				
	0	1	2	3	4	5	6	7	
1	T008	T007	T006	T005	T004	T003	T002	T001	
2	T016	T015	T014	T013	T012	T011	T010	T009	
3	T024	T023	T022	T021	T020	T019	T018	T017	
4	T032	T031	T030	T029	T028	T027	T026	T025	
5	T040	T039	T038	T037	T036	T035	T034	T033	
6	T048	T047	T046	T045	T044	T043	T042	T041	
7	T056	T055	T054	T053	T052	T051	T050	T049	
8	T064	T063	T062	T061	T060	T059	T058	T057	
9	T072	T071	T070	T069	T068	T067	T066	T065	
10	T080	T079	T078	T077	T076	T075	T074	T073	
11	T088	T087	T086	T085	T084	T083	T082	T081	
12	T096	T095	T094	T093	T092	T091	T090	T089	
13	T104	T103	T102	T101	T100	T099	T098	T097	
14	T112	T111	T110	T109	T108	T107	T106	T105	
15	T120	T119	T118	T117	T116	T115	T114	T113	
					
TC+1	CODA	CODB	modified octet count						

Figure 6: Preferred Packed TSVCSIS Payload Octets

In order to accommodate all other NRL VDR configurations, an alternate parameter placement MUST use two trailing bytes as shown in Figure 7. The last trailing byte MUST be filled with a two-bit rate code, CODA and CODB (both bits set to one), and its six-bit count field MUST be filled with ones. The second to last trailing byte MUST contain the parameter count (TC) in octets (a value from 1 and 255, inclusive). The value of zero SHALL be considered as reserved.

	MSB				LSB			
	0	1	2	3	4	5	6	7
1	T008	T007	T006	T005	T004	T003	T002	T001
2	T016	T015	T014	T013	T012	T011	T010	T009
				
TC+1	octet count							
TC+2	CODA	CODB	1	1	1	1	1	1

Figure 7: Length Unrestricted Packed TSVICIS Payload Octets

3.3. Multiple TSVICIS Frames in an RTP Packet

A TSVICIS RTP packet payload consists of zero or more consecutive TSVICIS coder frames (each consisting of MELPe 2400 and TSVICIS coder data), with the oldest frame first, followed by zero or one MELPe comfort noise frame. The presence of a comfort noise frame can be determined by its rate code bits in its last octet.

The default packetization interval is one coder frame (22.5, 67.5, or 90 ms) according to the coder bitrate (2400, 1200, or 600 bps). For some applications, a longer packetization interval is used to reduce the packet rate.

A TSVICIS RTP packet without coder and comfort noise frames MAY be used periodically by an endpoint to indicate connectivity by an otherwise idle receiver.

TSVICIS coder frames in a single RTP packet MAY have varying TSVICIS parameter octet counts. Its packed parameter octet count (length) is indicated in the trailing byte(s). All MELPe frames in a single RTP packet MUST be of the same coder bitrate. For all MELPe coder frames, the coder rate bits in the trailing byte identify the contents and length as per Table 1.

It is important to observe that senders have the following additional restrictions:

- * Senders SHOULD NOT include more TSVICIS or MELPe frames in a single RTP packet than will fit in the MTU of the RTP transport protocol.
- * Frames MUST NOT be split between RTP packets.

It is RECOMMENDED that the number of frames contained within an RTP packet be consistent with the application. For example, in telephony and other real-time applications where delay is important, the fewer frames per packet, the lower the delay. However, for bandwidth-constrained links or delay-insensitive streaming messaging applications, more than one frame per packet or many frames per packet would be acceptable.

Information describing the number of frames contained in an RTP packet is not transmitted as part of the RTP payload. The way to determine the number of TSVICIS/MELPe frames is to identify each frame type and length, thereby counting the total number of octets within the RTP packet.

3.4. Congestion Control Considerations

The target bitrate of TSVICIS can be adjusted at any point in time, thus allowing congestion management. Furthermore, the amount of encoded speech or audio data encoded in a single packet can be used for congestion control, since the packet rate is inversely proportional to the packet duration. A lower packet transmission rate reduces the amount of header overhead but at the same time increases latency and loss sensitivity, so it ought to be used with care.

Since UDP does not provide congestion control, applications that use RTP over UDP SHOULD implement their own congestion control above the UDP layer [RFC8085] and MAY also implement a transport circuit breaker [RFC8083]. Work in the RMCAT Working Group [RMCAT] describes the interactions and conceptual interfaces necessary between the application components that relate to congestion control, including the RTP layer, the higher-level media codec control layer, and the lower-level transport interface, as well as components dedicated to congestion control functions.

4. Payload Format Parameters

This RTP payload format is identified using the TSVCIS media subtype, which is registered in accordance with [RFC4855] and per the media type registration template from [RFC6838].

4.1. Media Type Definitions

Type name: audio

Subtype name: TSVCIS

Required parameters: Clock Rate (Hz): 8000

Optional parameters:

ptime:

the recommended length of time (in milliseconds) represented by the media in a packet. It SHALL use the nearest rounded-up ms integer packet duration. For TSVCIS, this corresponds to the following values: 23, 45, 68, 90, 112, 135, 156, and 180. Larger values can be used as long as they are properly rounded. See Section 6 of [RFC4566].

maxptime:

the maximum length of time (in milliseconds) that can be encapsulated in a packet. It SHALL use the nearest rounded-up ms integer packet duration. For TSVCIS, this corresponds to the following values: 23, 45, 68, 90, 112, 135, 156, and 180. Larger values can be used as long as they are properly rounded. See Section 6 of [RFC4566].

bitrate:

specifies the MELPe coder bitrates supported. Possible values are a comma-separated list of rates from the following set: 2400, 1200, 600. The modes are listed in order of preference; the first is preferred. If "bitrate" is not present, the fixed coder bitrate of 2400 MUST be used.

tcmax:

specifies the TSVCIS maximum value for the TC supported or desired, ranging from 1 to 255. If "tcmax" is not present, a default value of 35 is used.

Channels:

1

Encoding considerations: This media subtype is framed and binary; see Section 4.8 of [RFC6838].

Security considerations: Please see Section 8 of RFC 8817.

Interoperability considerations: N/A

Published specification: [TSVCIS]

Applications that use this media type: N/A

Fragment identifier considerations: N/A

Additional information:

Deprecated alias names for this type: N/A

Magic number(s): N/A

File extension(s): N/A

Macintosh file type code(s): N/A

Person & email address to contact for further information:

Victor Demjanenko, Ph.D. <victor.demjanenko@vocal.com>

Intended usage: COMMON

Restrictions on usage: The media subtype depends on RTP framing and hence is only defined for transfer via RTP [RFC3550]. Transport

within other framing protocols is not defined at this time.

Author: Victor Demjanenko, Ph.D.

Change controller: IETF; contact <avt@ietf.org>

Provisional registration? (standards tree only): No

4.2. Mapping to SDP

The mapping of the above-defined payload format media subtype and its parameters SHALL be done according to Section 3 of [RFC4855].

The information carried in the media type specification has a specific mapping to fields in the Session Description Protocol (SDP) [RFC4566], which is commonly used to describe RTP sessions. When SDP is used to specify sessions employing the TSVCSIS codec, the mapping is as follows:

- * The media type ("audio") goes in SDP "m=" as the media name.
- * The media subtype (payload format name) goes in SDP "a=rtpmap" as the encoding name.
- * The parameter "bitrate" goes in the SDP "a=fmtp" attribute by copying it as a "bitrate=<value>" string.
- * The parameter "tcmx" goes in the SDP "a=fmtp" attribute by copying it as a "tcmx=<value>" string.
- * The parameters "ptime" and "maxptime" go in the SDP "a=ptime" and "a=maxptime" attributes, respectively.

When conveying information via SDP, the encoding name SHALL be "TSVCIS" (the same as the media subtype).

An example of the media representation in SDP for describing TSVCSIS might be:

```
m=audio 49120 RTP/AVP 96
a=rtpmap:96 TSVCSIS/8000
```

The optional media type parameter "bitrate", when present, MUST be included in the "a=fmtp" attribute in the SDP, expressed as a media type string in the form of a semicolon-separated list of parameter=value pairs. The string "value" can be one or more of 2400, 1200, and 600, separated by commas (where each bitrate value indicates the corresponding MELPe coder). An example of the media representation in SDP for describing TSVCSIS when all three coder bitrates are supported might be:

```
m=audio 49120 RTP/AVP 96
a=rtpmap:96 TSVCSIS/8000
a=fmtp:96 bitrate=2400,600,1200
```

The optional media type parameter "tcmx", when present, MUST be included in the "a=fmtp" attribute in the SDP, expressed as a media type string in the form of a semicolon-separated list of parameter=value pairs. The string "value" is an integer number in the range of 1 to 255 representing the maximum number of TSVCSIS parameter octets supported. An example of the media representation in SDP for describing TSVCSIS with a maximum of 101 octets supported is as follows:

```
m=audio 49120 RTP/AVP 96
a=rtpmap:96 TSVCSIS/8000
a=fmtp:96 tcmx=101
```

The parameter "ptime" cannot be used for the purpose of specifying the TSVCSIS operating mode due to the fact that, for certain values, it will be impossible to distinguish which mode is about to be used

(e.g., when ptime=68, it would be impossible to distinguish whether the packet is carrying one frame of 67.5 ms or three frames of 22.5 ms).

Note that the payload format (encoding) names are commonly shown in upper case. Media subtypes are commonly shown in lower case. These names are case insensitive in both places. Similarly, parameter names are case insensitive in both the media subtype name and the default mapping to the SDP a=fmtp attribute.

4.3. Declarative SDP Considerations

For declarative media, the "bitrate" parameter specifies the possible bitrates used by the sender. Multiple TSVCIS rtpmap values (such as 97, 98, and 99, as used below) MAY be used to convey TSVCIS-coded voice at different bitrates. The receiver can then select an appropriate TSVCIS codec by using 97, 98, or 99.

```
m=audio 49120 RTP/AVP 97 98 99
a=rtpmap:97 TSVCIS/8000
a=fmtp:97 bitrate=2400
a=rtpmap:98 TSVCIS/8000
a=fmtp:98 bitrate=1200
a=rtpmap:99 TSVCIS/8000
a=fmtp:99 bitrate=600
```

For declarative media, the "tcmx" parameter specifies the maximum number of octets of TSVCIS packed parameters used by the sender or the sender's communications channel.

4.4. Offer/Answer SDP Considerations

In the Offer/Answer model [RFC3264], "bitrate" is a bidirectional parameter. Both sides MUST use a common "bitrate" value or values. The offer contains the bitrates supported by the offerer, listed in its preferred order. The answerer MAY agree to any bitrate by listing the bitrate first in the answerer response. Additionally, the answerer MAY indicate any secondary bitrate or bitrates that it supports. The initial bitrate used by both parties SHALL be the first bitrate specified in the answerer response.

For example, if offerer bitrates are "2400,600" and answerer bitrates are "600,2400", the initial bitrate is 600. If other bitrates are provided by the answerer, any common bitrate between the offer and answer MAY be used at any time in the future. Activation of these other common bitrates is beyond the scope of this document.

The use of a lower bitrate is often important for a case such as when one endpoint utilizes a bandwidth-constrained link (e.g., 1200 bps radio link or slower), where only the lower coder bitrate will work.

In the Offer/Answer model [RFC3264], "tcmx" is a bidirectional parameter. Both sides SHOULD use a common "tcmx" value. The offer contains the tcmx supported by the offerer. The answerer MAY agree to any tcmx equal to or less than this value by stating the desired tcmx in the answerer response. The answerer alternatively MAY identify its own tcmx and rely on TSVCIS ignoring any augmented data it cannot use.

5. Discontinuous Transmissions

A primary application of TSVCIS is for radio communications of voice conversations, and discontinuous transmissions are normal. When TSVCIS is used in an IP network, TSVCIS RTP packet transmissions may cease and resume frequently. RTP synchronization source (SSRC) sequence number gaps indicate lost packets to be filled by Packet Loss Concealment (PLC), while abrupt loss of RTP packets indicates intended discontinuous transmissions. Resumption of voice transmission SHOULD be indicated by the RTP marker bit (M) set to 1.

If a TSVCIS coder so desires, it may send a MELPe comfort noise frame

as per Appendix B of [SCIP210] prior to ceasing transmission. A receiver may optionally use comfort noise during its silence periods. No SDP negotiations are required.

6. Packet Loss Concealment

TSVCIS packet loss concealment (PLC) uses the special properties and coding for the pitch/voicing parameter of the MELPe 2400 bps coder. The PLC erasure indication utilizes any of the errored encodings of a non-voiced frame as identified in Table 1 of [MELPE]. For the sake of simplicity, it is preferred that a code value of 3 for the pitch/voicing parameter be used. Hence, set bits P0 and P1 to one and bits P2, P3, P4, P5, and P6 to zero.

When using PLC in 1200 bps or 600 bps mode, the MELPe 2400 bps decoder is called three or four times, respectively, to cover the loss of a low bitrate MELPe frame.

7. IANA Considerations

IANA has registered TSVCS as specified in Section 4.1. The media type has been added to the IANA registry for "RTP Payload Format Media Types" (<https://www.iana.org/assignments/rtp-parameters>).

8. Security Considerations

RTP packets using the payload format defined in this specification are subject to the security considerations discussed in the RTP specification [RFC3550] and in any applicable RTP profile such as RTP/AVP [RFC3551], RTP/AVPF [RFC4585], RTP/SAVP [RFC3711], or RTP/SAVPF [RFC5124]. However, as discussed in [RFC7202], it is not an RTP payload format's responsibility to discuss or mandate what solutions are used to meet such basic security goals as confidentiality, integrity, and source authenticity for RTP in general. This responsibility lies with anyone using RTP in an application. They can find guidance on available security mechanisms and important considerations in [RFC7201]. Applications SHOULD use one or more appropriate strong security mechanisms. The rest of this section discusses the security-impacting properties of the payload format itself.

This RTP payload format and the TSVCS decoder, to the best of our knowledge, do not exhibit any significant non-uniformity in the receiver-side computational complexity for packet processing and thus are unlikely to pose a denial-of-service threat due to the receipt of pathological data. Additionally, the RTP payload format does not contain any active content.

Please see the security considerations discussed in [RFC6562] regarding Voice Activity Detect (VAD) and its effect on bitrates.

9. References

9.1. Normative References

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