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TECHNICAL SPECIFICATION

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PEMEA Audio Video Extension

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Foreword

This Technical Specification (TS) has been produced by ETSI Special Committee Emergency Communications (EMTEL).

Modal verbs terminology

In the present document "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the ETSI Drafting Rules (Verbal forms for the expression of provisions).

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Executive summary

The Pan-European Mobile Emergency Application (PEMEA) architecture provides a framework to enable applications supporting emergency calling functionality to contact emergency services while roaming. PEMEA caters for a range of extension capabilities, including Audio_Video which provides an audio and/or video real-time communication capability between the App user and the PSAP. The present document provides a specification for an Audio_Video capability for PEMEA.

Introduction

Audio-video communications are a common communications capability in all modern communication Apps. These Apps support dedicated and multi-party video communications on a range of platforms including mobile phones, tablets, laptops and browsers. While some applications use SIP to establish communications sessions, the vast majority use WebRTC media streams, generally with proprietary signalling.

Using simple WebRTC Audio_Video communications enables equal access to emergency services for both abled and disabled citizens from almost any device, safely and easily while keeping with the modern trend towards all Web-based communications.

The specification in the present document does not preclude PEMEA from being used to support and initiate other protocols or implementations.

The present document assumes a working knowledge of PEMEA and familiarity with the PEMEA specification ETSI TS 103 478 [1]. Terms common to the PEMEA specification are not redefined or explained in detail in the present document.

1 Scope

The present document describes the PEMEA Audio_Video (PAV) capability, and the need for this functionality. The required entities and actors are identified along with the protocol, specifying message exchanges between entities. The message formats are specified and procedural descriptions of expected behaviours under different conditions are detailed.

2 References

2.1 Normative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

Referenced documents which are not found to be publicly available in the expected location might be found at https://docbox.etsi.org/Reference/.

NOTE: While any hyperlinks included in this clause were valid at the time of publication, ETSI cannot guarantee their long term validity.

The following referenced documents are necessary for the application of the present document.

[1] ETSI TS 103 478 (V1.1.1): "Emergency Communications (EMTEL); Pan-European Mobile Emergency Application".

[2] IETF RFC 2617 (June 1999): "HTTP Authentication: Basic and Digest Access Authentication".

[3] IETF RFC 6750 (October 2012): "The OAuth 2.0 Authorization Framework: Bearer Token Usage".

[4] IETF RFC 6455 (December 2011): "The WebSocket Protocol".

[5] IETF RFC 8445 (July 2018): "Interactive Connectivity Establishment (ICE): A Protocol for Network Address Translator (NAT) Traversal".

[6] IETF RFC 8839 (January 2021): "Session Description Protocol (SDP) Offer/Answer Procedures for Interactive Connectivity Establishment (ICE)".

[7] IETF RFC 7742 (March 2016): "WebRTC Video Processing and Codec Requirements".

[8] IETF RFC 7874 (May 2016): "WebRTC Audio Codec and Processing Requirements".

[9] IETF RFC 6716 (September 2012): "Definition of the Opus Audio Codec".

[10] IETF RFC 8251 (October 2017): "Updates to the Opus Audio Codec".

[11] IETF RFC 7587 (June 2015): "RTP Payload Format for the Opus Speech and Audio Codec".

[12] IETF RFC 8838 (January 2021): "Trickle ICE: Incremental Provisioning of Candidates for the Interactive Connectivity Establishment (ICE) Protocol".

[13] IETF RFC 4566 (July 2006): "SDP: Session Description Protocol".

[14] IETF RFC 8840 (January 2021): "A Session Initiation Protocol (SIP) Usage for Incremental Provisioning of Candidates for the Interactive Connectivity Establishment (Trickle ICE)".

[15] IETF RFC 8863 (January 2021): "Interactive Connectivity Establishment Patiently Awaiting Connectivity (ICE PAC)".

[16] IETF RFC 5766 (April 2020): "Traversal Using Relays around NAT (TURN): Relay Extensions to Session Traversal Utilities for NAT (STUN)".

2.2 Informative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

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The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] ETSI TS 103 756 (V1.1.1): "Emergency Communications (EMTEL); PEMEA Instant Message Extension".

[i.2] ETSI TS 103 871 (V1.1.1): "Emergency Communications (EMTEL); PEMEA Real-Time Text Extension".

[i.3] IETF RFC 7519 (May 2015): "JSON Web Token (JWT)".

[i.4] IETF RFC 8656 (February 2020): "Traversal Using Relays around NAT (TURN): Relay Extensions to Session Traversal Utilities for NAT (STUN)".

[i.5] IETF RFC 3551 (July 2003): "RTP Profile for Audio and Video Conferences with Minimal Control".

[i.6] IETF RFC 3389 (September 2002): "Real-time Transport Protocol (RTP) Payload for Comfort Noise (CN)".

3 Definition of terms, symbols and abbreviations

3.1 Terms

For the purposes of the present document, the following terms apply:

security: techniques and methods used to ensure:

- **authentication** of entities accessing resources or data.
- **authorization** of authenticated entities prior to accessing or obtaining resources and/or data.
- **privacy** of user data ensuring access only to authenticated and authorized entities.
- **secrecy** of information transferred between two authenticated and authorized entities.
- **trusted:** is used as defined in ETSI TS 103 478 [1].

3.2 Symbols

Void.

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

AEAD	Authenticated Encryption with Associated Data
AES	Advanced Encryption Standard
AESGCM	Advanced Encryption Standard key used with GCM
AP	Application Provider

App	Application
CN	Comfort Noise
CPE	Customer Premises Equipment
DHE	Diffie-Hellman key Exchange
ECDHE	Elliptic-Curve Diffie-Hellman key Exchange
ECDSA	Elliptic Curve Digital Signature Algorithm
EDS	Emergency Data Send (message)
ETSI	European Telecommunications Standards Institute
FID	Flow Identification
GCM	Galois/Counter Mode
HTTP	Hyper-Text Transfer Protocol
HTTPS	Secure HTTP
ICE	Interactive Connectivity Establishment
IETF	Internet Engineering Task Force
IM	Instant Messenger
IP	Internet Protocol
JSON	JavaScript Object Notation
JWT	JSON Web Token
MAC	Message Authentication Code
NAT	Network Address Translation
Pa	PEMEA Application to AP interface
PAV	PEMEA Audio_Video
PCMA	Pulse Code Modulation A-Law
PCMU	Pulse Code Modulation µ-Law
PEMEA	Pan-European Mobile Emergency Application
PIM	PSAP Interface Module
PSAP	Public Safety Answering Point
PSP	PSAP Service Provider
RFC	Request For Comments
RSA	Rivest Shamir Adleman public key encryption algorithm
RTC	Real-Time Communications
RTP	Real-Time Protocol
RTT	Real-Time Text
SDP	Session Description Protocol
SFU	Selective Forwarding Units
SIP	Session Initiation Protocol
STUN	Session Traversal Utilities for NAT
TLS	Transport Layer Security
tPSP	terminating PSP
TS	Technical Specification
TURN	Traversal Using Relay around NAT
UCS	Universal multiple-octet Coded Character Set
UDP	User Datagram Protocol
URI	Uniform Resource Identifier
UTC	Coordinated Universal Timer
UTF-8	UCS Transformation Format (8-bit words)
WMS	WebRTC MediaStream

4 PEMEA capability extensions

4.1 Overview of extension in PEMEA

PEMEA extension capabilities are defined in ETSI TS 103 478 [1] and are implemented through the use of "reach-back" URIs. The Application Provider (AP) node advertises capabilities as part of the initial forward message through the network, the Emergency Data Send (EDS) message, and the terminating PSAP Service Provider (PSP) or PSAP responds with the subset of capabilities that it supports, thus binding the emergency session between the AP and the terminating emergency node.

Specifically, the capabilities are sent as information elements in the apMoreInformation element of the EDS message. The information element and apMoreInformation structures are defined in clauses 10.3.11 and 10.3.12 of ETSI TS 103 478 [1]. An information element in a PEMEA EDS message identifies a capability and each capability is made up of three distinct parts:

- typeOfInfo: what function does the information element serve;
- protocol: the specific semantics for using the function;
- value: the URI through which the service is invoked.

Table 10 in ETSI TS 103 478 [1] identifies an initial set of "typeOfInfo" values used to specify a range of capability extensions for PEMEA. However, beyond the Location_Update and SIP_Request values described in Table 11 of ETSI TS 103 478 [1], protocols are left for further study and definition in subsequent specifications such as the present document. ETSI TS 103 756 [i.1] describes the concrete specification for PEMEA Instant Message protocol and ETSI TS 103 871 [i.2] describes the concrete specification for PEMEA Real-Time Text.

4.2 Service support indication and response

4.2.1 Service definition

ETSI TS 103 478 [1] defines the "Audio_Video" typeOfInfo in Table 10, but does not elaborate further on protocols in Table 11. The present document provides a concrete definition of the "Audio_Video" typeOfInfo in PEMEA through the specification of a protocol value. The definition in Table 1 shall be considered as an extension to Table 11 in ETSI TS 103 478 [1].

Table 1: Extended AP Information Type Protocol Registry

Info type Value	Protocol Token	Description
Audio_Video	PEMEA	Audio_Video functionality is supported using the PEMEA WebRTC message exchange protocol

4.2.2 Service support indication

An AP needing to indicate that the Application it is serving can support real-time text using the PEMEA protocol would include the following information element in the apMoreInformation element of the EDS associated with the emergency session:

```
<information typeOfInfo="Audio_Video" protocol="PEMEA">
    https://ap.example.pemea.help/37agq1cyusbo
</information>
```

4.2.3 Service support response

A terminating node that can support the "Audio_Video" "PEMEA" capability includes this capability in the apMoreInformation element returned to the AP in the onCapSupportPost. This is described in clause 11.1.4 of ETSI TS 103 478 [1] with the value for "Audio_Video" "PEMEA" provided in the example below.

```
<apMoreInformation xmlns="urn:pemea:apps:xml:ns:pemea:base">
    <information typeOfInfo="Audio_Video" protocol="PEMEA"/>
</apMoreInformation>
```

4.2.4 Auto response service

The original intent of many emergency applications was to provide ancillary data to the PSAP that was associated with an emergency voice call that the PSAP had, or soon would, receive. As a consequence, a PIM or tPSP usually notifies the PSAP-CPE when an EDS has arrived, but does not respond to the AP until a PSAP Call-Taker has answered the call. Operating in this manner allows for smart routing solutions ensuring that only the PSAP with the call binds the PEMEA session to the AP, ensuring that the data is always available to the PSAP Call-Taker rather than it being missing because it went to the wrong PSAP.

ETSI TS 103 478 [1] identifies some types of capabilities, most notably the SIP_Request capabilities, as being responded to automatically, that is, the PIM or tPSP sends an immediate onCapSupportPost message with all supported capabilities if the EDS contains a SIP_Request capability. This functionality is described in clause 8 of ETSI TS 103 478 [1] and came about because there was no way for the App to make a voice call until it has a destination SIP URI, so there was no possible way for the data to not be available at the destination PSAP.

Another reason for auto-response is that no conventional carrier/mobile voice call will be placed as part of the emergency communication. That is, only PEMEA advanced services will be used for communicating between the Caller and the PSAP Call-Taker.

The (PAV) capability falls into this latter category of services, that is, it is used in place of a conventional carrier/mobile voice call. Consequently, a PSAP (PIM or tPSP) supporting this capability and with the capacity to handle the communication shall respond to the AP with an onCapSupportPost message immediately on receipt of an EDS containing a PAV capability. The onCapSupportPost message shall contain the PAV capability along with any other capabilities that the PSAP supports.

If the PSAP does not support the PAV capability but does support another form of proffered multi-media communication, such as IM or RTT, then the PSAP should respond with only those capabilities.

5 Architecture

5.1 Overview

The PEMEA Audio_Video (PAV) capability is WebRTC-based, so it is necessary to understand a little bit about WebRTC in order to understand what is signalled, to whom and between which entities. This also helps to understand explicitly what is normatively specified in the present document, what is semantic, and what is normatively referred to from the present document but normatively specified in other documents.

The PAV capability was realized with disability usage in mind and the usage model for this necessitates multi-party communications where the Caller and PSAP Call-Taker represent two parties, but a third-party sign-language interpreter is also, more than likely, expected to participate in the call.

5.2 Architecture and high-level WebRTC flows

WebRTC was designed to use peer-to-peer communications between web browsers, and this means that media stream connections are between peers. In an Audio_Video flow, each participant can have 2 streams to send, one audio stream from a microphone and one video stream from a camera. It is not mandatory to have always both streams, and a participant could only negotiate its audio stream and not initially negotiate its video stream.

PEMEA is structured around the AP being the gateway between the App and the PSAP. This model requires communications to occur, first between the App and the AP over the proprietary Pa interface and then between the AP and the associated PSAP service using the protocol mechanisms defined in the specific extension capability document. For most services, this approach is fine, however, for latency and delay intolerant services this approach may need to be relaxed somewhat to ensure that the capability delivers the required functionality. Such a case occurs with media streams that are providing a conversation, the streams should be as direct as possible, so this is the approach allowed for the PAV capability. Signalling for the session goes via the AP and to the PAV Service in the PSAP, while the media streams may go directly between the Caller and the PAV or via the AP if security measure require it. This shown in Figure 1.

Media streams, where possible, should be as direct as possible between the communicating parties, however, owing to a range of networking impediments, communications often do not happen in this fashion and so it is necessary to use a TURN Server in the media path to ensure connectivity. The role of the TURN Server is to provide a common contact point between different communication parties through which the media streams can flow. In practical terms, in PSAP deployments firewalls exist between the Call-Taker equipment and any incoming media streams, so a TURN Server is always required. Consequently, a TURN Server is included as a mandatory component of the PAV architecture. The TURN server specification is defined in IETF RFC 5766 [16].

In the core WebRTC specifications a media server may be used to aggregate streams reducing the number of required peer connections, to allow transcoding between the participants, or to control the flow of the streams during the conversation. However, demultiplexing multiple streams on single connection increases complexity which is undesirable and unwarranted in emergency calling-applications where the number of participants in the communication is small. Consequently, PAV requires stream sets to be negotiated independently between participants using an architectural approached referred to as Selective Forwarding Units (SFU). Development of PEMEA Apps has been centred around equivalent access to emergency services, and so allowing control over how to display the videos of different participants is crucial to improve the accessibility and usability of these Apps.

Figure 1 depicts the PAV service architecture. It includes not only a TURN Server to allow NAT traversal, but also a Media Server to allow the control of the media streams as described in clause 15. It can also be used to record conversations for audit purposes.

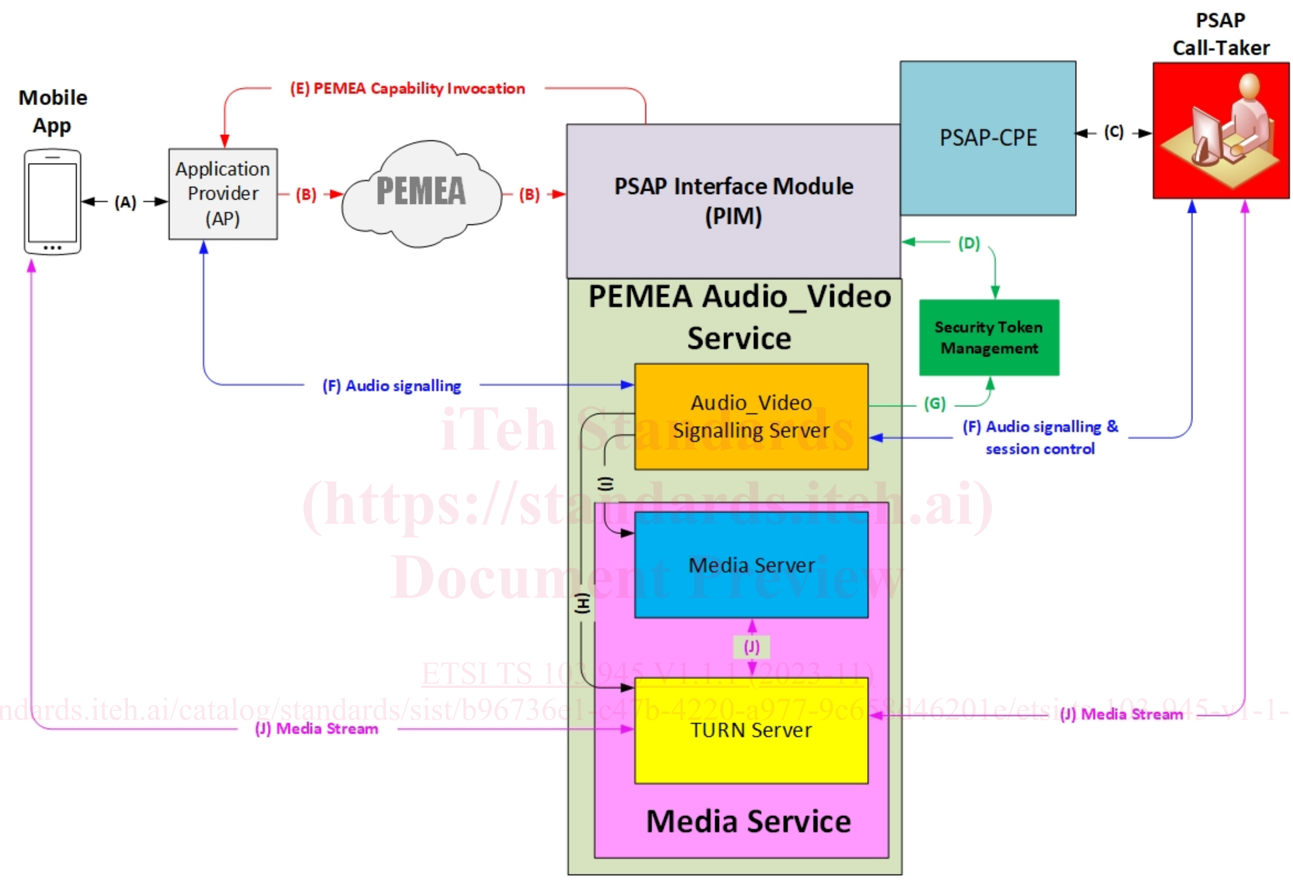

Figure 1: Audio_Video architecture for PEMEA

A. Pa interface, the application makes a call to the AP indicating the PAV capability. Invocation information is returned to the App over this interface too.

B. The AP packages the information from the App into an EDS message and sends it into the PEMEA network via the Ps interface. The EDS arrives at the PIM over the Pp interface. The PIM sends an onCapSupportPost message to the AP binding the connection between the AP and the PIM.

C. The PIM notifies the PSAP-CPE which in turn notifies the PSAP Call-Taker. The call is answered and controlled by the PSAP Call-Taker over this interface also. This includes requesting the creation of a PAV session. The PSAP Call-Taker is also able to request connection credentials for additional participants over this interface.

D. On direction from the PSAP Call-Taker (or automatically as described in clause 4.2.4) the PIM creates a new PAV signalling instance and requests Bearer tokens from the Security Token Management system. It shall generate at least a token for the PSAP Call-Taker and a token for the Caller.