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Information technology — Telecommunications and information exchange between systems — Using CSTA for SIP phone user agents (uaCSTA)

*Technologies de l'information — Télécommunications et échange
d'information entre systèmes — Utilisation de CSTA pour agents
d'utilisateurs de téléphone de SIP (uaCSTA)*

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Foreword

ISO (the International Organization for Standardization) and IEC (the International Electrotechnical Commission) form the specialized system for worldwide standardization. National bodies that are members of ISO or IEC participate in the development of International Standards through technical committees established by the respective organization to deal with particular fields of technical activity. ISO and IEC technical committees collaborate in fields of mutual interest. Other international organizations, governmental and non-governmental, in liaison with ISO and IEC, also take part in the work. In the field of information technology, ISO and IEC have established a joint technical committee, ISO/IEC JTC 1.

International Standards are drafted in accordance with the rules given in the ISO/IEC Directives, Part 2.

The main task of the joint technical committee is to prepare International Standards. Draft International Standards adopted by the joint technical committee are circulated to national bodies for voting. Publication as an International Standard requires approval by at least 75 % of the national bodies casting a vote.

In exceptional circumstances, the joint technical committee may propose the publication of a Technical Report of one of the following types:

- type 1, when the required support cannot be obtained for the publication of an International Standard, despite repeated efforts;
- type 2, when the subject is still under technical development or where for any other reason there is the future but not immediate possibility of an agreement on an International Standard;
- type 3, when the joint technical committee has collected data of a different kind from that which is normally published as an International Standard (“state of the art”, for example).

Technical Reports of types 1 and 2 are subject to review within three years of publication, to decide whether they can be transformed into International Standards. Technical Reports of type 3 do not necessarily have to be reviewed until the data they provide are considered to be no longer valid or useful.

Attention is drawn to the possibility that some of the elements of this document may be the subject of patent rights. ISO and IEC shall not be held responsible for identifying any or all such patent rights.

ISO/IEC TR 22767, which is a Technical Report of type [3], was prepared by Ecma International (as ECMA TR/87) and was adopted, under a special “fast-track procedure”, by Joint Technical Committee ISO/IEC JTC 1, *Information technology*, Subcommittee SC 6, *Telecommunications and information exchange between systems*, in parallel with its approval by national bodies of ISO and IEC.

Introduction

This Technical Report illustrates how CSTA can be used over a SIP session to control and observe SIP user agents (uaCSTA).

This Technical Report is part of a suite of Ecma CSTA Phase III Standards and Technical Reports.

All of the Standards and Technical Reports in this suite are based upon the practical experience of Ecma member companies and each one represents a pragmatic and widely based consensus.

Information technology — Telecommunications and information exchange between systems — Using CSTA for SIP phone user agents (uaCSTA)

1 Scope

The Session Initiation Protocol (SIP) is a control (signalling) protocol for creating, modifying, and terminating sessions with one or more participants. These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences.

CSTA standardizes a very powerful and flexible set of application services to observe and control voice and non-voice media calls as well as control and observe non-call related features.

This Technical Report describes how CSTA can be used to provide a subset of CSTA call control functionality, called 1st party call control, for SIP user agents. The term uaCSTA (for user agent CSTA) refers to transporting ECMA-323 (CSTA XML) messages over a SIP session.

uaCSTA leverages SIP mechanisms to provide a highly featured, robust, and extensible set of features to support applications in the Enterprise environment.

uaCSTA can be implemented by several different types of SIP user agents:

- directly by a SIP user agent on a SIP phone
- uaCSTA can also be implemented by a SIP B2BUA to augment 3PCC functionality
- by a proxy server that is front-ending a PBX.

2 Purpose

- Describe the relevant portions of the CSTA Standards – The two primary CSTA Standards, Services for Computer Supported Telecommunications Applications (ECMA-269) and the XML Protocol for CSTA (ECMA-323), are relatively large standards (combined over 1100 pages). Due to their size it is sometimes difficult for SIP developers without prior knowledge of CSTA standards to quickly find the relevant parts of the CSTA standards needed to implement basic features. This TR shows the relevant CSTA concepts and how they can be used to implement a CSTA-based application protocol without having to read all of the CSTA Standards.
- Terminology – Although many of the concepts are similar, different terms are used in SIP and CSTA to describe the same concepts. Since CSTA is designed to be protocol independent, it is helpful to show how the abstract terminology of CSTA is mapped to SIP specific terminology.
- Extensibility - A SIP phone that is being developed to support a specific application may initially need to only support a small subset of the features standardized in CSTA. As the types and complexity of applications using these devices increase, it is expected that these devices will need to support additional, more advanced, features standardized by CSTA, similar to features supported by other types of phones in Enterprise environments. This TR shows how basic features can be extended to support a rich standards-based feature set for applications.

- Interoperability - In order to encourage interoperability of applications and phones supporting this application protocol, additional CSTA Profiles, which include minimal sets of CSTA functionality, are described. These profiles can be extended by implementations to provide a more complete set of call and devices features commonly used by Enterprise applications.

3 Normative references

The following referenced documents are indispensable for the application of this document. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments) applies.

This TR provides informative examples of how to use CSTA concepts as an application protocol for SIP user agents. The following Ecma Standards should be used as the definitive references for CSTA.

ECMA-269, *Services for Computer Supported Telecommunications Applications (CSTA) Phase III, 6th edition, June 2004 (International Standard ISO/IEC 18051)*

ECMA-323, *XML Protocol for Computer Supported Telecommunications Applications (CSTA) Phase III, 3rd edition, June 2004 (International Standard ISO/IEC 18056)*

The following IETF references provide information on the SIP features referenced in this TR:

RFC 3261, *SIP: Session Initiation Protocol, Rosenberg, J. et al, IETF, June 2002*