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

**iTeh STANDARD PREVIEW**  
*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.

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# 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 1<sup>st</sup> 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, 6<sup>th</sup> edition, June 2004 (International Standard ISO/IEC 18051)*

ECMA-323, *XML Protocol for Computer Supported Telecommunications Applications (CSTA) Phase III, 3<sup>rd</sup> 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*

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### 4 Terminology

#### 4.1 General terms

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The following table summarizes some of the commonly used terminologies used in this Technical Report.

application	Refers to all components that control and observe uaCSTA instances that may be present in an application/computing domain such as an application residing on any type of computing device (desktop PC, mobile computing device, CTI middleware, SIP B2BUA, SIP UA, etc.).
uaCSTA	<u>u</u> ser <u>a</u> gent <u>C</u> STA – This refers the mechanism to transport CSTA messages over a SIP session, and the type of call control, called 1 <sup>st</sup> party call control, that allows an application to control and observe only the device or devices it is directly involved with. A desktop application controlling its associated phone, for example.
uaCSTA device	Refers to a SIP UA, such as a SIP or PBX phone, that supports the CSTA features over SIP as described in this TR.
phone	A SIP UA that supports calls. It is also a PBX device that supports calls. Examples of the types of phones that can be supported by this application protocol are described in clause 6.
B2BUA	A back-to-back-user-agent is a type of user agent that acts both as a user agent server (receives requests) and acts as a user agent client (generates requests). B2BUA is used to implement SIP third party call control (3PCC) features. uaCSTA could be used to augment the existing B2BUA functionality for enhanced call control.

## 4.2 SIP/CSTA Terminology Mappings

The following table shows some of the common SIP terms and how they are referenced using CSTA and vice versa.

SIP Term	CSTA Term	Notes
User Agent or UA	device	A SIP UA can be modelled as a type of CSTA device. Whenever the term device is used in this TR, it is referring to a SIP UA.
SIP URI	device identifier or deviceID	A SIP URI can be used to address a SIP UA or a PBX phone in the same way that a CSTA device identifier can be used to address a CSTA device. One of the formats of a Device Identifier is URI. Whenever the term DeviceID is used in this TR, it is referring to a URI format of CSTA deviceID. Refer to Annex A for a description of the ways to represent a user and a user's device with a URI.
TEL URI	device identifier or deviceID	A TEL URI can be used to address a device or a dialled number in the same way that a CSTA device identifier can be used to address a CSTA device. Refer to Annex A for a description of the ways to represent a device and a dialled number with a TEL URI.

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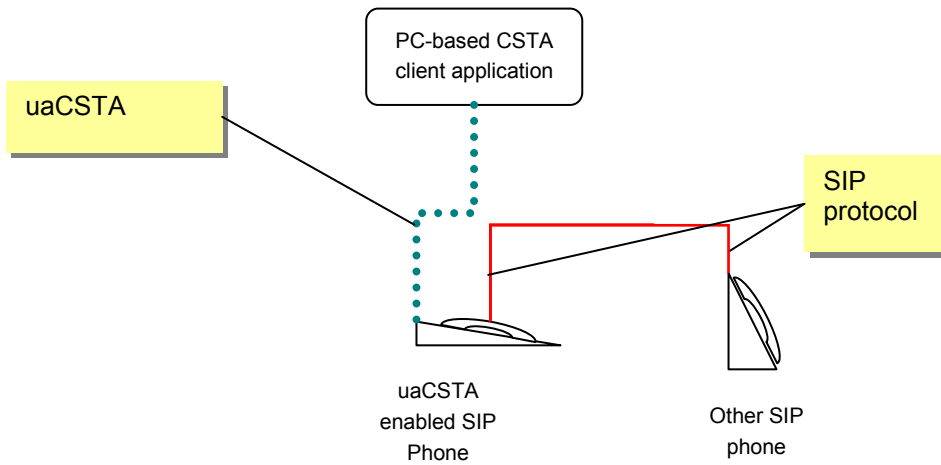
## 5 Example Environments for uaCSTA

The following figures illustrate how uaCSTA is used to pass ECMA-323 messages between an application and a user agent that supports this protocol. uaCSTA provides a standards-based protocol for applications to use to request a UA to invoke features.

The term uaCSTA device refers to a SIP UA, such as a SIP phone, that supports the ECMA-323 messages over SIP as described in this Technical Report.

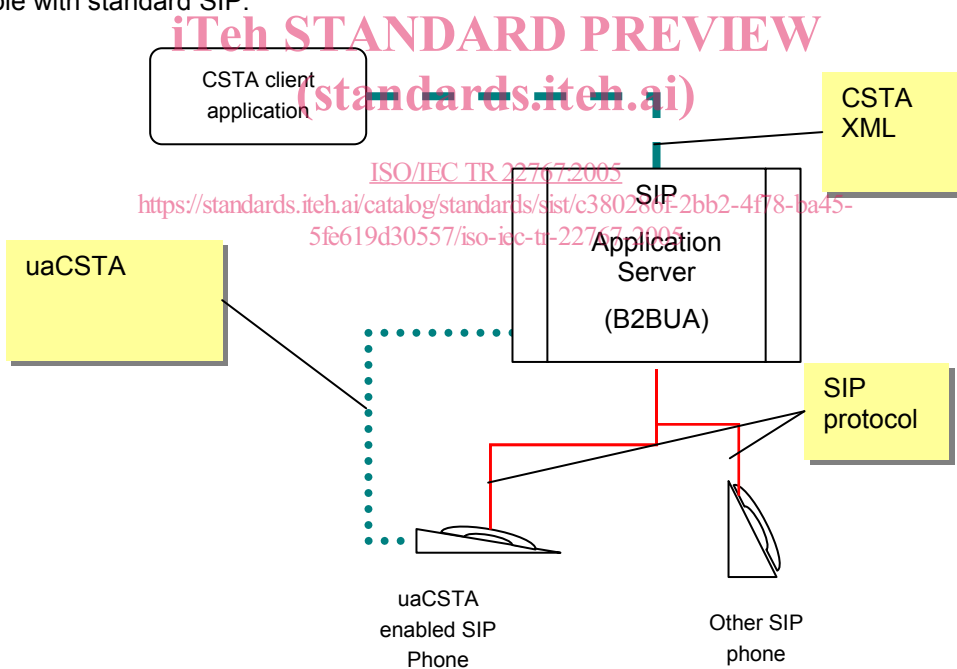
### 5.1 Controlling and Observing a SIP Phone

In this environment, a PC-based application can use uaCSTA to directly control and observe its associated SIP UA phone.



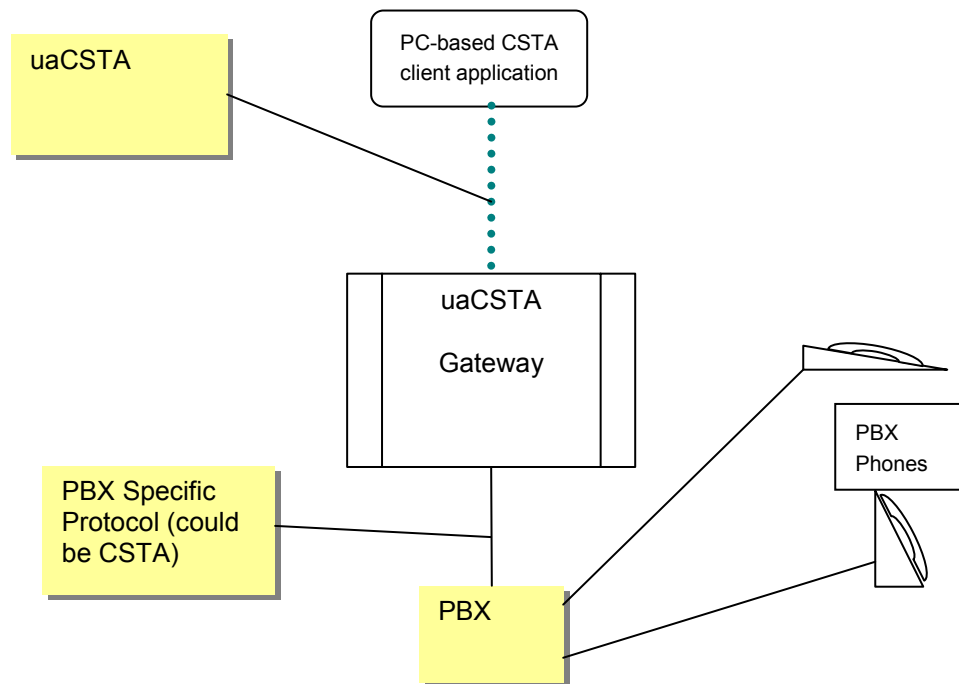
### 5.2 Controlling and Observing a SIP Phone by Augmenting B2BUA Functionality

In this environment, uaCSTA is used to augment existing B2BUA functionality to invoke features on the SIP phone not possible with standard SIP.



### 5.3 Controlling a PBX Phone

In this environment, a PC-based application can use uaCSTA to control its associated PBX phone that is part of a PBX/IP Switch. A SIP/CSTA Gateway is a type SIP UA that terminates SIP and converts ECMA-323 messages to/from an application to/from PBX dependent protocols. The uaCSTA Gateway allows an application to control only its associated device(s) – (1<sup>st</sup> party call control).



## 6 Example User Agent Configurations

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CSTA defines two elements that are relevant for user agents:

- A logical element – the set of attributes/features/services that have an association with the control and/or observation of calls at a UA. For example (line) appearances and attributes such as forwarding. Some phones provide more than one addressable logical elements (e.g. multi line phones). Some phones share a logical element among multiple physical devices (e.g. bridged appearance phones).
- A physical element – the set of attributes/features/services that have any association with the control or observation of the physical components (speaker, microphone, display, etc.) of a UA.

Some UAs provide a single address which can be used to address all of its (logical and physical) elements while other UAs provide multiple addresses so that requests can be directed to a specific logical element on the UA (e.g. a specific line on a multi-line phone).

This chapter discusses the typical types of phones that can be supported by this application protocol, the types of elements supported by phone type, and how applications can address each element.

### 6.1 Single Line Phone UA

A single line phone is a type of user agent that supports only one logical element (addressable line) not shared with any other device and a single physical element (see 6.1.3.3.2 of ECMA-269). The single line can support one or more calls at the same time.

For a single line phone, a single CSTA deviceID (e.g. SIP URI) is used to reference both the phone's logical and physical elements.

For example if the phone has registered with the contact address of sip:1234@domain1, the various elements of the phone could be addressed via CSTA services as follows:

- phone's logical element with a call control feature: To originate a call from the single line, an application would provide sip:1234@domain1 as the callingDevice in the CSTA Make Call service.

- phone's logical element with a logical device feature: To control the forwarding feature for the single line, an application would provide sip:1234@domain1 as the deviceID in the CSTA Set Forwarding service.
- phone's physical element with a physical device feature: To set the speaker mute for the device, an application would provide sip:1234@domain1 as the deviceID in the CSTA Set Speaker Mute service.

## 6.2 Multi Line Phone UA

A multi line phone is a type of user agent that supports more than one addressable line (see 6.1.3.3.3 of ECMA-269). Each line can support one or more calls at the same time.

For a multi line phone, there are multiple logical elements (lines) and a single physical element.

For a multi line phone, each logical element (line) is addressed with a unique CSTA deviceID (e.g. SIP URI). CSTA services that reference a phone's logical element use the deviceID (e.g. SIP contact URI) associated with the desired line. CSTA services that apply to the physical element address it via any one of the SIP contact URIs since they all, in this phone configuration, refer to the same device.

For example if the phone has registered with the contacts of sip:1234@domain1 and sip:5678@domain1, the various elements of the phone could be addressed via CSTA services as follows:

- phone's logical component with a call control feature:
  - To originate a call from the line addressed via sip:1234@domain1, an application would provide sip:1234@domain1 as the callingDevice in the CSTA Make Call service.
  - To originate a call from the line addressed via sip:5678@domain1, an application would provide sip:5678@domain1 as the callingDevice in the CSTA Make Call service.
- phone's logical element with a logical device feature:
  - To control the forwarding feature for the line addressed via sip:1234@domain1, an application would provide sip:1234@domain1 as the deviceID in the CSTA Set Forwarding service.
  - To control the forwarding feature for the line addressed via sip:5678@domain1, an application would provide sip:5678@domain1 as the deviceID in the CSTA Set Forwarding service.
- phone's physical element with a physical device feature: To set the speaker mute for the device, an application would provide either one of the following:
  - sip:1234@domain1 as the deviceID in the CSTA Set Speaker Mute service. Since this contact is only associated with one physical device, it would be unambiguous.
  - sip:5678@domain1 as the deviceID in the CSTA Set Speaker Mute service. Since this contact is only associated with one physical device, it would be unambiguous.

## 6.3 Bridged Appearance Phone and other Advanced UA Configurations

A bridged-appearance phone is a type of user agent that contains a logical element (e.g. a line) shared with another phone (see 6.1.3.3.5 of ECMA-269).

CSTA specifies many different kinds of bridged appearances that reflect the different behaviours and different capabilities for calls at shared appearances. (See bridged calls in section of ECMA-269.)

uaCSTA does not specify (nor preclude) the behaviour or the addressing of bridged appearances and other advanced UA configurations.

## 7 SIP Transport Mechanism for CSTA Messages

This clause describes how:

- SIP is used to establish a CSTA application session
- CSTA service request and response messages are transported over SIP
- a CSTA Monitor is started and CSTA events are transported over SIP.

The mechanism described here uses the SIP INVITE and INFO methods to transport ECMA-323 messages provided in a SIP message body.

The SIP Content-type used for an ECMA-323 message is application/csta+xml. This MIME media type is used specifically for encapsulating ECMA-323 messages.

A user agent using this mechanism could be any type of SIP UA such as a SIP phone or a SIP user agent that is front-ending a PBX, like the CSTA/SIP server as shown in [Clause 5](#).

**NOTE** The SIP mechanism described here to establish a CSTA application session and transport ECMA-323 messages is one of many possible mechanisms. Other options such as using SIP SUBSCRIBE/NOTIFY methods with a CSTA event package(s) could be used. SIP extensions to methods such as REFER may provide another option.

### 7.1 Establishing a CSTA Application Session

Before ECMA-323 service requests are sent to a SIP UA, a CSTA application session must be established between the application and the SIP UA.

ECMA-269 specifies a number of options for establishing a CSTA application session. This mechanism uses the "Implicit Association created using CSTA Request System Status" option as described in 7.2 of ECMA-269.

The application creates an application session by establishing a SIP dialog with the user agent using a SIP INVITE method that includes a Content-Disposition header indicating "signal" and "handling=required" to mandate support for the application/csta+xml MIME type.

An ECMA-323 Request System Status service request is included in the SIP INVITE body with the Content-Type application/csta+xml.

The Request-URI header in the SIP INVITE could specify a particular user agent instance (e.g. globally unique contact URI) or a SIP Address-of-Record URI (a user's published address). If the Request-URI specifies a SIP Address-of-Record for which multiple devices have registered contacts, the application should be aware that a SIP proxy may pick a specific contact to forward this INVITE to or the proxy may parallel fork the INVITE to all of the contacts. This behaviour depends upon the proxy design, policy, scripting, etc. In general if the INVITE is forked, the application will get the contact URIs from all instances and can choose which specific contact it wishes to monitor and/or control using uaCSTA.

If the UA supports this MIME type and can establish this application session, it responds to this SIP Invite with a 200 OK (line 2 below) that includes an ECMA-323 Request System Status service response in the message body. The application sends a SIP ACK to complete the dialog establishment (line 3).

If the UA does not support this MIME type, the SIP UA must provide a 415 (Unsupported Media Type) response as defined in IETF RFC 3261.

The application-control session exists for as long as the application is interested in controlling the SIP user agent which might, for some deployments, persist as long as the user agent is operational. Once the application-control session is no longer needed, it can be terminated using the SIP BYE method.