# TECHNICAL REPORT

# DSL Forum
# TR-104

# DSLHome™
# Provisioning Parameters for VoIP CPE

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**Abstract:**
This document defines provisioning parameters for VoIP CPE as an extension to TR-069.

# Contents

# 1 Introduction

This specification defines the data model for provisioning of a voice-over-IP (VoIP) CPE device by an Auto-Configuration Server (ACS) using the mechanism defined in TR-069 [2].

The goals of this specification are as follows:

- Accommodate VoIP devices that are either embedded as part of an Internet Gateway Device, as defined in [2], or standalone VoIP devices.
- Accommodate VoIP devices that support multiple distinct VoIP services, each potentially with multiple distinct lines.
- Support the use of both SIP [4] and MGCP [5] signaling protocols.
- Support various types of VoIP CPE including VoIP endpoints, SIP outbound proxies, and SIP back-to-back user agents.

## 1.1 Terminology

The following terminology is used throughout this document.

| | |
|---|---|
| **ACS** | Auto-Configuration Server. This is a component in the broadband network responsible for auto-configuration of the CPE for advanced services. |
| **CPE** | Customer Premises Equipment. |
| **Directory Number** | A distinct number by which a Line is addressed. |
| **Endpoint** | A VoIP device that acts as the initiation/termination point for VoIP calls. Examples of Endpoints include VoIP phones and analog terminal adapters (ATAs). |
| **Line** | A separately addressable voice line with a distinct Directory Number. |
| **Parameter** | A name-value pair representing a manageable CPE parameter made accessible to an ACS for reading and/or writing. |
| **Profile** | A group of Lines with common characteristics. |
| **Session** | A single active two-way voice media session. A single Line may support more than one active Session, for example for CPE provided three-way calling. |

## 1.2 Document Conventions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [1].

# 2 Architecture

This document defines a VoiceService as the container associated with the provisioning objects for VoIP CPE. CPE making use of a VoiceService object MUST adhere to all of the data-hierarchy requirements defined in [3]. In the context of [3], the VoiceService object defined in this specification is a Service Object. As defined in [3], individual CPE devices may contain one or more instances of the VoiceService object. The presence of more than one VoiceService object might be appropriate, for example, where a CPE device serves as a management proxy for other non-TR-069 capable VoIP CPE. For example, an Internet Gateway Device might serve as a management proxy for one or more non-TR-069 capable VoIP phones.

A single VoiceService object contains one or more VoiceProfile objects. Each VoiceProfile corresponds to one or more phone lines that share the same basic configuration. Each VoiceProfile object contains one or more Line objects, each of which represents a single distinct phone line.

The VoiceProfile object is included in the model to allow a multi-line voice device to group lines with common characteristics under a single profile. By allowing more than one VoiceProfile, the model allows a single multi-line voice device to have groups of lines that are configured differently from others. One possible use of this structure could be to associate distinct groups of lines with completely separate service providers, each with distinct VoIP servers and configuration requirements. Another possible use could be to distinguish between different levels of service from a single service provider. For example, a single device could provide some “consumer” lines plus some “business” lines, each associated with a distinct VoiceProfile distinguished by their quality characteristics. While the VoiceProfile concept is inherent in the defined model, the need to make use of it is completely optional.

Figure 1 shows the complete object structure for a VoiceService. The following conventions are used in this diagram:

- Optional or conditionally required objects are shown with a dashed border.
- Objects that for which there may be multiple instances are shown a overlapping layered blocks.
- Multiply instanced objects for which instances can be explicitly added or deleted are indicated with an asterisk (“*”).

*Note – some sub-objects are not shown in this figure.*

*Note – while the only protocol-specific objects defined in this specification are for SIP, MGCP, and H.323, the object structure allows for the possibility of parallel objects for other call-control protocols, such as H.248. Specific object definitions for other such protocols are beyond the scope of this specification.*

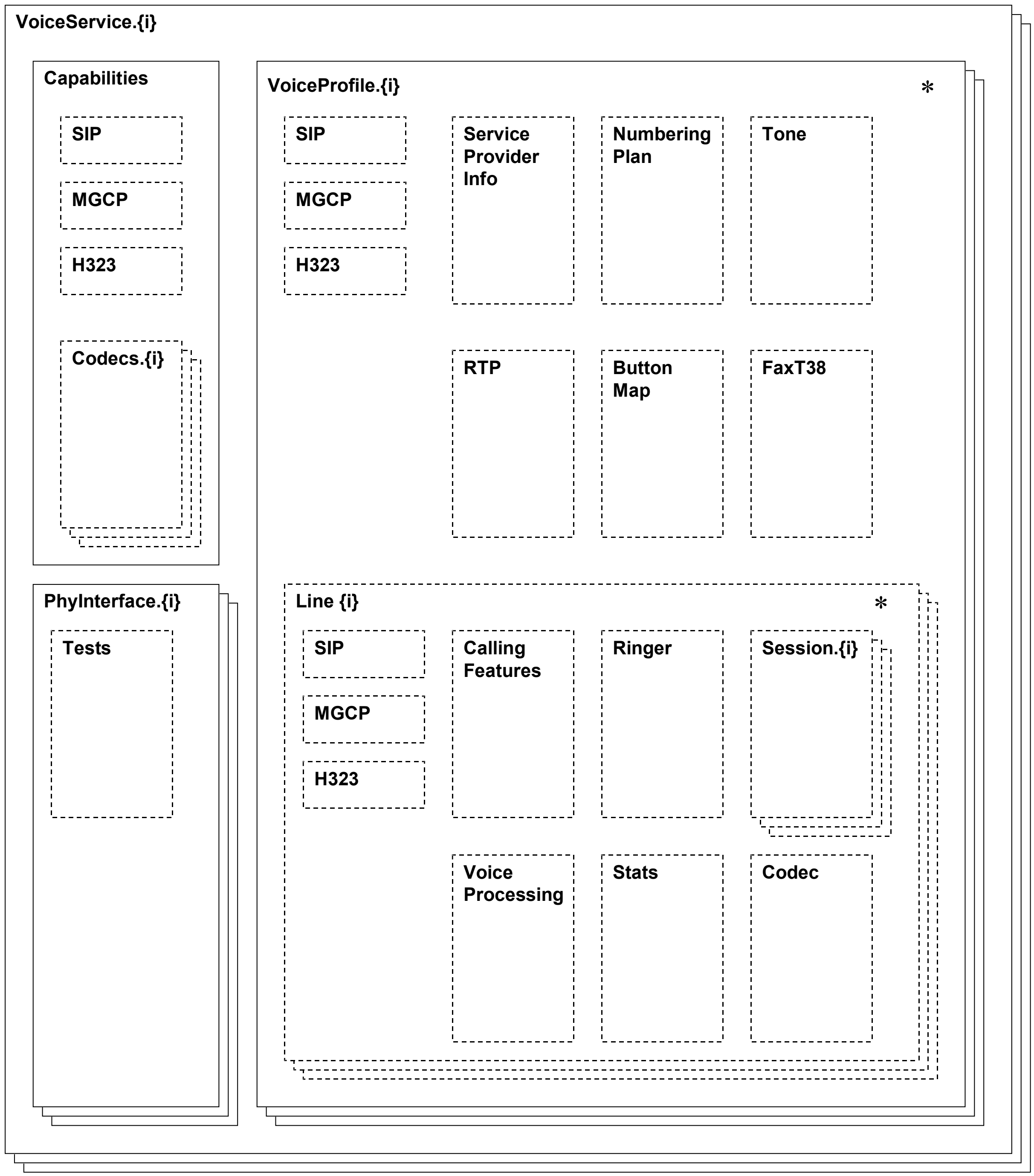


**Figure 1 – VoiceService object structure**

# 3 VoiceService Version 1.0 Data Model

Table 1 lists the objects associated with VoIP CPE and their associated parameters. This table defines version 1.0 of the VoiceService data model.

The notation used to indicate the data type of each parameter, and the notation associating with multi-instance objects, follows the notation defined in [3].

**Table 1 – Definition of VoiceService version 1.0**

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| .VoiceService.{i}. | object | - | The top-level object for VoIP CPE. | - |
| VoiceProfileNumberOfEntries | unsignedInt | - | Number of instances of VoiceProfile. | - |
| .VoiceService.{i}.Capabilities. | object | - | The overall capabilities of the VoIP CPE. | - |
| MaxProfileCount | unsignedInt | - | Maximum total number of distinct voice profiles supported. | - |
| MaxLineCount | unsignedInt | - | Maximum total number of lines supported across all profiles.<br>This parameter is applicable only for a VoIP endpoint. | - |
| MaxSessionsPerLine | unsignedInt | - | Maximum number of voice sessions supported for any given line across all profiles. A value greater than one indicates support for CPE provided conference calling.<br>This parameter is applicable only for a VoIP endpoint. | - |
| MaxSessionCount | unsignedInt | - | Maximum total number of voice sessions supported across all lines and profiles. (This might differ from MaxLineCount if each line can support more than one session for CPE provided conference calling. This value MAY be less than the product of MaxLineCount and MaxSessions-PerLine.) | - |
| SignalingProtocols | string(256) | - | Comma-separated list of signaling protocols supported. Each item is an enumeration of:<br>"SIP"<br>"MGCP"<br>"MGCP-NCS"<br>"H.248"<br>"H.323"<br>Each entry MAY be appended with a version indicator in the form "/X.Y". For example:<br>"SIP/2.0"<br>The list MAY include vendor-specific protocols, which MUST be in the format defined in [3]. For example:<br>"X_EXAMPLE-COM_MyProt" | - |
| Regions | string(256) | - | Comma-separated list of geographic regions supported by the CPE. Each item is the list MUST be an alpha-2 (two-character alphabetic) country code as specified by ISO 3166. | - |

1 The full name of a Parameter is the concatenation of the root object name as defined in [3], the object name shown in the yellow header, and the individual Parameter name.

2 "W" indicates the parameter MAY be writable (if "W" is not present, the parameter is defined as read-only). For an object, "W" indicates object instances can be Added or Deleted.

3 The default value of the parameter on creation of an object instance via TR-069. If the default value is an empty string, this is represented by the symbol <Empty>.

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | An empty list indicates that the CPE does not support region-based customization via the Region parameter in the VoiceService.{i}.-VoiceProfile.{i} object. | |
| RTCP | boolean | - | Support for RTCP. A true value indicates support for the object VoiceService.{i}.VoiceProfile.{i}.-RTP.RTCP.<br>This parameter is applicable only for a VoIP endpoint. | - |
| SRTP | boolean | - | Support for SRTP. A true value indicates support for the object VoiceService.{i}.VoiceProfile.{i}.-.RTP.SRTP.<br>A true value also indicates that the SRTPKeying-Methods and SRTPEncryptionKeySizes parameters in this object are present.<br>This parameter is applicable only for a VoIP endpoint. | - |
| SRTPKeyingMethods | string(256) | - | Comma-separated list of keying protocols supported by this endpoint for SRTP. Each item is an enumeration of:<br>"Null"<br>"Static"<br>"SDP"<br>"IKE"<br>This list MAY include vendor-specific keying methods, which MUST use the format defined in [3].<br>This parameter is applicable only if the SRTP parameter in this object is equal to true. | - |
| SRTPEncryptionKeySizes | string(256) | - | Comma-separated list of unsigned integers, each represented a supported SRTP encryption key size.<br>This parameter is applicable only if the SRTP parameter in this object is equal to true. | - |
| RTPRedundancy | boolean | - | Support for RTP payload redundancy as defined in RFC 2198. A true value indicates support for VoiceService.{i}.VoiceProfile.{i}.RTP.Redundancy.<br>This parameter is applicable only for a VoIP endpoint. | - |
| DSCPCoupled | boolean | - | A true value indicates that the CPE is constrained such that transmitted call control packets use the same DSCP marking as transmitted RTP packets.<br>If the value is true, the CPE MUST NOT support the DSCPMark parameter for call control.<br>This parameter is applicable only for a VoIP endpoint. | - |
| EthernetTaggingCoupled | boolean | - | A true value indicates that the CPE is constrained such that transmitted call control packets use the same Ethernet tagging (VLAN ID Ethernet Priority) as transmitted RTP packets.<br>If the value is true, the CPE MUST NOT support the VLANIDMark or EthernetPriorityMark parameters within a call control object (e.g., SIP, MGCP, or H323).<br>This parameter is applicable only for a VoIP endpoint. | - |
| PSTNSoftSwitchOver | boolean | - | A true value indicates the CPE is capable of supporting the PSO_Activate Facility Action, | |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | which allows a call to be switched to a PSTN FXO (Foreign eXchange Office) line.<br>This parameter is applicable only for a VoIP endpoint. | |
| FaxT38 | boolean | - | Support for T.38 fax. A true value indicates support for the object VoiceService.{i}.-VoiceProfile.{i}.FaxT38.<br>This parameter is applicable only for a VoIP endpoint. | - |
| FaxPassThrough | boolean | - | Support for fax pass-through. A true value indicates support for the parameter VoiceService.-{i}.VoiceProfile.{i}.FaxPassThrough.<br>This parameter is applicable only for a VoIP endpoint. | - |
| ModemPassThrough | boolean | - | Support for modem pass-through. A true value indicates support for the parameter VoiceService.-{i}.VoiceProfile.{i}.ModemPassThrough.<br>This parameter is applicable only for a VoIP endpoint. | - |
| ToneGeneration | boolean | - | Support for tone generation. A true value indicates support for the object VoiceService.{i}.-VoiceProfile.{i}.Tone.<br>A true value also indicates that the Tone-DescriptionsEditable, PatternBasedTone-Generation, and FileBasedToneGeneration parameters in this object are present.<br>This parameter is applicable only for a VoIP endpoint. | - |
| ToneDescriptionsEditable | boolean | - | Indicates whether or not the VoiceService.{i}.-VoiceProfile.{i}.Tone.Description and VoiceService.{i}.VoiceProfile.{i}.Tone.Pattern tables are editable (if entries can be added, removed, or modified).<br>This parameter is applicable only if the Tone-Generation parameter in this object is equal to true. | - |
| PatternBasedToneGeneration | boolean | - | Support for tone generation by pattern specification. A true value indicates support for the object VoiceService.{i}.VoiceProfile.{i}.Tone.<br>If the ToneGeneration parameter in this object is true, at least one of PatternBasedToneGeneration and FileBasedToneGeneration MUST also be true.<br>This parameter is applicable only if the Tone-Generation parameter in this object is equal to true. | - |
| FileBasedToneGeneration | boolean | - | Support for tone generation by file playback. A true value indicates support for the object VoiceService.{i}.VoiceProfile.{i}.Tone.<br>A true value also indicates that the ToneFile-Formats parameter in this object is present.<br>If the ToneGeneration parameter in this object is true, at least one of PatternBasedToneGeneration and FileBasedToneGeneration MUST also be true.<br>This parameter is applicable only if the Tone-Generation parameter in this object is equal to true. | - |
| ToneFileFormats | string(256) | - | Comma-separated list of tone file formats supported. The specified file formats are raw | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | codec data files, using one of the codecs listed below. Each item listed is an enumeration of:<br>"G.711MuLaw"<br>"G.711ALaw"<br>"MP3"<br>"WAV"<br>"AMR"<br>The list MAY include vendor-specific -specific extensions, which MUST use the format defined in [3].<br>Example:<br>"G.711MuLaw, MP3, X_EXAMPLE-COM_MyFileFormat"<br>If the CPE does not support tone files, this parameter MUST be the empty string.<br>This parameter is applicable only if the FileBased-ToneGeneration parameters in this object is equal to true. | |
| RingGeneration | boolean | - | Support for ring generation. A true value indicates support for control of ring generation via the VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Ringer object.<br>A true value also indicates that the Ring-DescriptionsEditable, PatternBasedRing-Generation, and FileBasedRingGeneration parameters in this object are present.<br>This parameter is applicable only for a VoIP endpoint. | - |
| RingDescriptionsEditable | boolean | - | Indicates whether or not the VoiceService.{i}.-VoiceProfile.{i}.Line.{i}.Ringer.Description and VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Ringer.-Pattern tables are editable (if entries can be added, removed, or modified).<br>This parameter is applicable only if the Ring-Generation parameter in this object is equal to true. | - |
| PatternBasedRingGeneration | boolean | - | Support for ring generation by pattern specification. A true value indicates support for the VoiceService.{i}.VoiceProfile.{i}.Line.{i}.-Ringer.Pattern table.<br>A true value also indicates that the RingPattern-Editable parameter in this object is present.<br>This parameter is applicable only if the Ring-Generation parameter in this object is equal to true. | - |
| RingPatternEditable | boolean | - | Indicates whether or not the VoiceService.{i}.-VoiceProfile.{i}.Line.{i}.Ringer.Pattern table is editable (if entries can be added, removed, or modified).<br>This parameter is applicable only if the Pattern-BasedRingGeneration parameter in this object is equal to true. | - |
| FileBasedRingGeneration | boolean | - | Support for ring generation by file playback. A true value indicates support for specification of ringer files in the VoiceService.{i}.VoiceProfile.{i}.-Line.{i}.Ringer.Description table.<br>A true value also indicates that the RingFile-Formats parameter in this object is present. | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | This parameter is applicable only if the Ring-Generation parameter in this object is equal to true. | |
| RingFileFormats | string(256) | - | Comma-separated list of ring file formats supported. Each item listed is an enumeration of:<br>"MIDI"<br>"SMAF" (MMF)<br>"RTTTL" (RTTTL or RTX)<br>"MP3"<br>"WAV"<br>"AMR"<br>The list MAY include vendor-specific -specific extensions, which MUST use the format defined in [3].<br>Example:<br>"MIDI, AMR, X_EXAMPLE-COM_MyFileFormat"<br>If the CPE does not support ring files, this parameter MUST be the empty string.<br>This parameter is applicable only if the FileBased-RingGeneration parameters in this object is equal to true. | - |
| DigitMap | boolean | - | Support for a configurable digit map string. A true value indicates full support for the VoiceService.-{i}.VoiceProfile.{i}.DigitMap parameter. | - |
| NumberingPlan | boolean | - | Support for a configurable numbering plan. A true value indicates support for a configurable numbering plan via the VoiceService.{i}.-VoiceProfile.{i}.NumberingPlan object.<br>This parameter is applicable only for a VoIP endpoint. | - |
| ButtonMap | boolean | - | Support for a configurable button map. A true value indicates support for a configurable button map via the VoiceService.{i}.VoiceProfile.{i}.-ButtonMap object.<br>This parameter is applicable only for a VoIP endpoint. | - |
| VoicePortTests | boolean | - | Support for remotely accessible voice-port tests. A true value indicates support for the VoiceService.{i}.PhyInterface.{i}.Tests object.<br>This parameter is applicable only for a VoIP endpoint. | - |
| .VoiceService.{i}.Capabilities.SIP. | object | - | SIP-specific capabilities. Applicable only if SIP is among the list of supported protocols. | - |
| Role | string | - | The role of this VoIP CPE. Enumeration of:<br>"UserAgent"<br>"BackToBackUserAgents"<br>"OutboundProxy"<br>A single VoiceService instance MUST have only one role. If a device includes the capabilities for more than one role, each role MUST be represented as separate VoiceService instances. | - |
| Extensions | string(256) | - | Comma-separated list of SIP extension methods supported. SIP extension methods MUST be in the form of the method name in upper case.<br>The list MAY include vendor-specific extensions, | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | which MUST use the format defined in [3].<br>Examples:<br>"REFER"<br>"INFO"<br>"X_EXAMPLE-COM_MyExt" | |
| Transports | string(256) | - | Comma-separated list of SIP transport protocols supported. Each entry is an enumeration of:<br>"UDP"<br>"TCP"<br>"TLS"<br>"SCTP"<br>The list MAY include vendor-specific transports, which MUST use the format defined in [3]. | - |
| URISchemes | string(256) | - | Comma-separated list of URI schemes supported beyond the URI schemes required by the SIP specification. Each URI scheme is given by the URI prefix, without the colon separator. Example:<br>"tel, fax" | - |
| EventSubscription | boolean | - | Support for SIP event subscription. A true value indicates support for the VoiceService.{i}.-VoiceProfile.{i}.SIP.EventSubscribe and VoiceService.{i}.VoiceProfile.{i}.Line.{i}.SIP. EventSubscribe.{i} objects. | - |
| ResponseMap | boolean | - | Support for SIP response map. A true value indicates support for the VoiceService.{i}.-VoiceProfile.{i}.SIP.ResponseMap object.<br>This parameter is applicable only for a VoIP endpoint. | - |
| TLSAuthenticationProtocols | string(256) | - | Comma-separated list of authentication protocols supported for TLS transport. Each item listed is an enumeration of:<br>"Null"<br>"MD5"<br>"SHA-1"<br>The list MAY include vendor-specific protocols, which MUST use the format defined in [3].<br>Support for this parameter is applicable only if the Transports parameter in this object includes the value "TLS". | - |
| TLSAuthenticationKeySizes | string(256) | - | Comma-separated list of unsigned integers, each representing a supported TLS authentication key size.<br>Support for this parameter is applicable only if the Transports parameter in this object includes the value "TLS" and the TLSAuthenticationProtocols parameter in this object is present and non-empty and includes at least one value other than "Null". | - |
| TLSEncryptionProtocols | string(256) | - | Comma-separated list of authentication protocols supported for TLS transport. Each item listed is an enumeration of:<br>"Null"<br>"RC4"<br>"RC2"<br>"DES"<br>"3DES" | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | The list MAY include vendor-specific protocols, which MUST use the format defined in [3].<br>Support for this parameter is applicable only if the Transports parameter in this object includes the value "TLS". | |
| TLSEncryptionKeySizes | string(256) | - | Comma-separated list of unsigned integers, each representing a supported TLS encryption key size.<br>Support for this parameter is applicable only if the Transports parameter in this object includes the value "TLS" and the TLSEncryptionProtocols parameter in this object is present and non-empty and includes at least one value other than "Null". | - |
| TLSKeyExchangeProtocols | string(256) | - | Comma-separated list of authentication protocols supported for TLS transport. Each item listed is an enumeration of:<br>"RSA"<br>"DSS"<br>The list MAY include vendor-specific protocols, which MUST use the format defined in [3].<br>Support for this parameter is applicable only if the Transports parameter in this object includes the value "TLS" and the TLSEncryptionProtocols parameter in this object is present and non-empty and includes at least one value other than "Null". | - |
| .VoiceService.{i}.Capabilities.MGCP. | object | - | MGCP-specific capabilities. Applicable only if MGCP is among the list of supported protocols. | - |
| Extensions | string(256) | - | Comma-separated list of optional MGCP packages supported. MGCP packages are listed using the uppercase package abbreviation.<br>The list MAY include vendor-specific extensions, which MUST use the format defined in [3].<br>Examples:<br>"BP"<br>"X_EXAMPLE-COM_MyExt" | - |
| .VoiceService.{i}.Capabilities.H323. | object | - | H.323-specific capabilities. Applicable only if H.323 is among the list of supported protocols. | - |
| FastStart | boolean | - | Support for H323 fast start. A true value indicates support for fast start. | - |
| H235AuthenticationMethods | string(256) | - | Comma-separated list of authentication methods supported. Each item listed is an enumeration of:<br>"dhExch" (Diffie-Hellman)<br>"pwdSymEnc" (password with symmetric encryption)<br>"pwdHash" (password with hashing)<br>"certSign" (certificate with signature)<br>"ipsec" (IPSEC based connection)<br>"tls" (TLS)<br>The list MAY include vendor-specific protocols, which MUST use the format defined in [3]. | - |
| .VoiceService.{i}.Capabilities.Codecs.{i}. | object | - | Table to describe the set of supported codecs. Each entry refers to a distinct combination of codec and bit rate. The table MUST include a distinct entry for each supported combination of these parameters.<br>Applicable only for a VoIP endpoint. | - |
| EntryID | unsignedInt | - | Unique identifier for each entry in this table. | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | [1:] | | | |
| Codec | string(64) | - | Identifier of the type of codec. Enumeration of:<br>"G.711MuLaw"<br>"G.711ALaw"<br>"G.726"<br>"G.729"<br>"G.729a"<br>"G.729e"<br>"G.728"<br>"G.723.1"<br>"G.722"<br>"G.722.1"<br>"G.722.2"<br>"GSM-FR"<br>"GSM-HR"<br>"GSM-EFR"<br>"iLBC"<br>The parameter MAY instead be a vendor-specific codec, which MUST be in the format defined in [3]. For example:<br>"X_EXAMPLE-COM_MyCodec" | - |
| BitRate | unsignedInt | - | Bit rate, in bits per second. The value MUST be among the values appropriate for the specified codec. | - |
| PacketizationPeriod | string(64) | - | Comma-separate list of supported packetization periods, in milliseconds, or continuous ranges of packetization periods. Ranges are indicated as a hyphen-separated pair of unsigned integers. Examples:<br>"20" indicates a single discrete value.<br>"10, 20, 30" indicates a set of discrete values.<br>"5-40" indicates a continuous inclusive range.<br>"5-10, 20, 30" indicates a continuous range in addition to a set of discrete values.<br>A range MUST only be indicated if all values within the range are supported. | - |
| SilenceSuppression | boolean | - | Indicates support for silence suppression for this codec. | - |
| .VoiceService.{i}.VoiceProfile.{i}. | object | W | Object associated with a collection of voice lines with common characteristics. Support for adding and removing profiles is conditional on whether more than one profile is supported as indicated by VoiceService.{i}.Capabilities.MaxProfileCount. By default, a single VoiceProfile object SHOULD be present in a VoiceService, initially in the disabled state. | - |
| Enable | string | W | Enables or disables all lines in this profile, or places it into a quiescent state. Enumeration of:<br>"Disabled"<br>"Quiescent"<br>"Enabled"<br>On creation, a profile MUST be in the Disabled state. | False |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | In the Quiescent state, in-progress sessions remain intact, but no new sessions are allowed. Support for the Quiescent state in a CPE is optional. If this parameter is set to "Quiescent" in a CPE that does not support the Quiescent state, it MUST treat it the same as the Disabled state. | |
| Reset | boolean | W | When written as true, forces the all lines in this profile to be reset, causing it to re-initialize and perform all start-up actions such as registration. Always False when read. | - |
| NumberOfLines | unsignedInt | - | Number of instances of Line within this VoiceProfile.<br>Applicable only for a VoIP endpoint. | 0 |
| Name | string(64) | W | Human-readable string to identify the profile instance. | <Empty> |
| SignalingProtocol | string(64) | W | The protocol to be used for this profile. A single protocol selected from among the available protocols indicated in VoiceService.{i}.-Capabilities.SignalingProtocols. | <Empty> |
| MaxSessions | unsignedInt | W | Limit on the number of simultaneous voice sessions across all lines in this profile. Must be less than or equal to VoiceService.{i}.Capabilities.-MaxSessionCount. (This MAY be greater than the number of lines if each line can support more than one session, for example for CPE provided conference calling.) | 0 |
| DTMFMethod | string(64) | W | Method by which DTMF digits MUST be passed. Enumeration of:<br>"InBand"<br>"RFC2833"<br>"SIPInfo"<br>If the parameter DTMFMethodG711 is non-empty, then this parameter applies only when the current codec is not G.711.<br>The value "SIPInfo" is applicable only if the SignalingProtocol is SIP.<br>This parameter is applicable only for a VoIP endpoint. | "InBand" |
| DTMFMethodG711 | string(64) | W | Method by which DTMF digits MUST be passed if the current codec is G.711. Enumeration of:<br>"InBand"<br>"RFC2833"<br>"SIPInfo"<br>An empty value for this parameter indicates that the value of the DTMFMethod parameter is to apply whether or not the current codec is G.711.<br>The value "SIPInfo" is applicable only if the SignalingProtocol is SIP.<br>This parameter is applicable only for a VoIP endpoint. | <Empty> |
| Region | string(2) | W | The geographic region associated with this profile. This MAY be used by the CPE to customize localization settings.<br>The value MUST be either one value selected from among the available regions indicated in VoiceService.{i}.Capabilities.Regions, or MAY be empty. An empty value indicates that the region is unspecified and the CPE SHOULD use default | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | localization settings.<br>This parameter is applicable only if VoiceService.-{i}.Capabilities.Regions is non-empty. | |
| DigitMap | string(256) | W | Digit map controlling the transmission of dialed digit information. The string defines the criteria to be met as digits are collected before an outgoing request (e.g., a SIP INVITE) can be initiated.<br>The syntax of this parameter is exactly the syntax used by MGCP as defined in [5], section 2.1.5.<br>This parameter is applicable only if the device supports a dialing mechanism for which a dialing plan is needed (for example, a device with an explicit Dial button may not need to be aware of the dialing plan) and if the device does not already support a dialing plan mechanism for this profile (e.g., in-band via MGCP).<br>If this object is supported, the capability VoiceService.{i}.Capabilities.DigitMap MUST be equal to true.<br>Applicable only for a VoIP endpoint. | <Empty> |
| DigitMapEnable | boolean | W | Enables the use of the DigitMap parameter in this object.<br>When enabled, the content of the VoiceService.-{i}.VoiceProfile.{i}.NumberingPlan object, if present, MUST be ignored.<br>This parameter is required if and only if both the DigitMap parameter in this object and the VoiceService.{i}.VoiceProfile.{i}.NumberingPlan object are present.<br>Applicable only for a VoIP endpoint. | True |
| STUNEnable | boolean | W | Enable or disable use of STUN to allow operation through NAT. Note: enabling STUN is to be interpreted as enabling the use of STUN for discovery, not use as a keep-alive mechanism. | False |
| STUNServer | string(256) | W | Domain name or IP address of the STUN server. | <Empty> |
| NonVoiceBandwidthReservedUpstream | unsignedInt | W | For bandwidth-based admission control, indicates the amount of upstream bandwidth, in bits per second, that must be left available for non-voice traffic when determining whether a session can proceed. This parameter is appropriate only in implementations in which the actual bandwidth can be known, such as a VoIP device embedded in a DSL B-NT. | 0 |
| NonVoiceBandwidthReservedDownstream | unsignedInt | W | For bandwidth-based admission control, indicates the amount of downstream bandwidth, in bits per second, that must be left available for non-voice traffic when determining whether a session can proceed. This parameter is appropriate only in implementations in which the actual bandwidth can be known, such as a VoIP device embedded in a DSL B-NT. | 0 |
| PSTNFailOver | boolean | W | Specifies whether or not the CPE SHOULD fail over to PSTN service, if available, on loss of connectivity to the VoIP service. This parameter is appropriate only in implementations in which PSTN fail-over is possible. | False |
| FaxPassThrough | string | W | Specifies the behavior of the CPE for pass-through of fax data. Enumeration of:<br>“Disable”<br>“Auto” | “Auto” |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | “Force”<br>The value “Disable” prevents the CPE from switching to a fax pass-through mode.<br>The value “Auto” allows the CPE to automatically detect fax data to determine whether or not to switch to a fax pass-through mode.<br>The value “Force” forces the CPE to switch to a fax pass-through mode regardless of whether fax signaling is detected.<br>If this parameter is supported, the capability VoiceService.{i}.Capabilities.FaxPassThrough MUST be equal to true.<br>This parameter is appropriate only for a VoIP endpoint. | |
| ModemPassThrough | string | W | Specifies the behavior of the CPE for pass-through of modem data. Enumeration of:<br>“Disable”<br>“Auto”<br>“Force”<br>The value “Disable” prevents the CPE from switching to a modem pass-through mode.<br>The value “Auto” allows the CPE to automatically detect modem data to determine whether or not to switch to a modem pass-through mode.<br>The value “Force” forces the CPE to switch to a modem pass-through mode regardless of whether modem signaling is detected.<br>If this parameter is supported, the capability VoiceService.{i}.Capabilities.ModemPassThrough MUST be equal to true.<br>This parameter is appropriate only for a VoIP endpoint. | “Auto” |
| .VoiceService.{i}.VoiceProfile.{i}.Service-ProviderInfo. | object | - | Information regarding the organization providing service for this voice profile instance. | - |
| Name | string(256) | W | Human-readable string identifying the service provider. | <Empty> |
| URL | string(256) | W | URL of the service provider for this profile instance. | <Empty> |
| ContactPhoneNumber | string(32) | W | Phone number to contact the service provider for this profile instance. | <Empty> |
| EmailAddress | string(256) | W | Email address to contact the service provider for this profile instance. | <Empty> |
| .VoiceService.{i}.VoiceProfile.{i}.SIP. | object | - | Voice profile parameters that are specific to SIP user agents. | -[4] |
| ProxyServer | string(256) | W | Host name or IP address of the SIP proxy server.<br>All SIP signaling traffic MUST be sent to the host indicated by this parameter and the port indicated by the ProxyServerPort parameter unless OutboundProxy parameter is non-empty or a different route was discovered during normal operations SIP routing operation.<br>Regardless of which host the traffic gets sent to (the ProxyServer or the OutboundProxy), the value of this parameter MUST be used to derive the URI placed into the SIP Route header field of all requests originated by this end-point unless a | <Empty> |

[4] Creation of this object occurs on specification of SIP as the SignalingProtocol in the parent object.

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | different proxy host was discovered dynamically during normal SIP routing operations. | |
| ProxyServerPort | unsignedInt [0:65535] | W | Destination port to be used in connecting to the SIP server. | 5060 |
| ProxyServerTransport | string | W | Transport protocol to be used in connecting to the SIP server. Must be chosen from among the transports supported, as indicated by VoiceService.{i}.Capabilities.SIP.Transports. Enumeration of:<br>"UDP"<br>"TCP"<br>"TLS"<br>"SCTP" | "UDP" |
| RegistrarServer | string(256) | W | Host name or IP address of the SIP registrar server.<br>If this parameter is empty, the CPE MUST obtain all of the registrar server configuration information, including host name or IP address, port, and transport protocol, from the corresponding ProxyServer parameters (ProxyServer, Proxy-ServerPort, and ProxyServerTransport), ignoring all of the registrar server parameters (Registrar-Server, RegistrarServerPort and RegistrarServer-Transport). | <Empty> |
| RegistrarServerPort | unsignedInt [0:65535] | W | Destination port to be used in connecting to the SIP registrar server.<br>If the RegistrarServer parameter is empty, the CPE MUST obtain all of the registrar server configuration information, including host name or IP address, port, and transport protocol, from the corresponding ProxyServer parameters (Proxy-Server, ProxyServerPort, and ProxyServer-Transport), ignoring all of the registrar server parameters (RegistrarServer, RegistrarServerPort and RegistrarServerTransport). | 5060 |
| RegistrarServerTransport | string | W | Transport protocol to be used in connecting to the registrar server. Must be chosen from among the transports supported, as indicated by VoiceService.{i}.Capabilities.SIP.Transports. Enumeration of:<br>"UDP"<br>"TCP"<br>"TLS"<br>"SCTP"<br>If the RegistrarServer parameter is empty, the CPE MUST obtain all of the registrar server configuration information, including host name or IP address, port, and transport protocol, from the corresponding ProxyServer parameters (Proxy-Server, ProxyServerPort, and ProxyServer-Transport), ignoring all of the registrar server parameters (RegistrarServer, RegistrarServerPort and RegistrarServerTransport). | "UDP" |
| UserAgentDomain | string(256) | W | CPE domain string. If empty, the CPE SHOULD use its IP address as the domain. | <Empty> |
| UserAgentPort | unsignedInt [0:65535] | W | Port used for incoming call control signaling. | 0 |
| UserAgentTransport | string | W | Transport protocol to be used for incoming call control signaling. Must be chosen from among the transports supported, as indicated by | "UDP" |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | VoiceService.{i}.Capabilities.SIP.Transports.<br>Enumeration of:<br>"UDP"<br>"TCP"<br>"TLS"<br>"SCTP" | |
| OutboundProxy | string(256) | W | Host name or IP address of the outbound proxy. If a non-empty value is specified, the SIP endpoint MUST send all SIP traffic (requests and responses) to the host indicated by this parameter and the port indicated by the OutboundProxyPort parameter. This MUST be done regardless of the routes discovered using normal SIP operations, including use of Route headers initialized from Service-Route and Record-Route headers previously received. The OutboundProxy value is NOT used to generate the URI placed into the Route header of any requests. | <Empty> |
| OutboundProxyPort | unsignedInt [0:65535] | W | Destination port to be used in connecting to the outbound proxy. This parameter MUST be ignored unless the value of the OutboundProxy parameter in this object is non-empty. | 5060 |
| Organization | string(256) | W | Text string to be used in the Organization header. | <Empty> |
| RegistrationPeriod | unsignedInt [1:] | W | Period over which the user agent must periodically register, in seconds. | - |
| TimerT1 | unsignedInt [1:] | W | Value of SIP timer T1, in milliseconds, as defined in RFC 3261. | - |
| TimerT2 | unsignedInt [1:] | W | Value of SIP timer T2, in milliseconds, as defined in RFC 3261. | - |
| TimerT4 | unsignedInt [1:] | W | Value of SIP timer T4, in milliseconds, as defined in RFC 3261. | - |
| TimerA | unsignedInt [1:] | W | Value of SIP timer A, in milliseconds, as defined in RFC 3261. | - |
| TimerB | unsignedInt [1:] | W | Value of SIP timer B, in milliseconds, as defined in RFC 3261. | - |
| TimerC | unsignedInt [1:] | W | Value of SIP timer C, in milliseconds, as defined in RFC 3261. | - |
| TimerD | unsignedInt [1:] | W | Value of SIP timer D, in milliseconds, as defined in RFC 3261. | - |
| TimerE | unsignedInt [1:] | W | Value of SIP timer E, in milliseconds, as defined in RFC 3261. | - |
| TimerF | unsignedInt [1:] | W | Value of SIP timer F, in milliseconds, as defined in RFC 3261. | - |
| TimerG | unsignedInt [1:] | W | Value of SIP timer G, in milliseconds, as defined in RFC 3261. | - |
| TimerH | unsignedInt [1:] | W | Value of SIP timer H, in milliseconds, as defined in RFC 3261. | - |
| TimerI | unsignedInt [1:] | W | Value of SIP timer I, in milliseconds, as defined in RFC 3261. | - |
| TimerJ | unsignedInt [1:] | W | Value of SIP timer J, in milliseconds, as defined in RFC 3261. | - |
| TimerK | unsignedInt [1:] | W | Value of SIP timer K, in milliseconds, as defined in RFC 3261. | - |
| InviteExpires | unsignedInt [1:] | W | Invite request Expires header value, in seconds. | - |
| ReInviteExpires | unsignedInt | W | Re-invite request Expires header value, in | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | [1:] | | seconds. | |
| RegisterExpires | unsignedInt [1:] | W | Register request Expires header value, in seconds. | - |
| RegistersMinExpires | unsignedInt [1:] | W | Register request Min-Expires header value, in seconds. | - |
| RegisterRetryInterval | unsignedInt [1:] | W | Register retry interval, in seconds. | - |
| InboundAuth | string | W | Type of inbound authentication, if any, required. Enumeration of:<br>"None"<br>"Digest" | "None" |
| InboundAuthUsername | string(256) | W | If inbound authentication is required, the username credentials. | <Empty> |
| InboundAuthPassword | string(256) | W | If inbound authentication is required, the password credentials.<br>When read, this parameter returns an empty string, regardless of the actual value. | <Empty> |
| UseCodecPriorityInSDPResponse | boolean | W | When the value is true, in the SDP included in an OK response to an Invite, the first listed codec MUST be the highest priority codec among those offered in the Invite, based on the priorities specified in the table VoiceService.{i}.-VoiceProfile.{i}.Line.{i}.Codec.List.{i}. The list of codecs in the SDP MAY also include other lower priority codecs.<br>When the value is false, there is no specific requirement for choosing the codecs listed in the SDP included in an OK response. | False |
| DSCPMark | unsignedInt [0:63] | W | Diffserv code point to be used for outgoing SIP signaling packets. | 0 |
| VLANIDMark | int[-1:] | W | VLAN ID (as defined in 802.1Q) to be used for outgoing SIP signaling packets for this profile. A value of -1 indicates the default value is to be used.<br>If either the VLANIDMark or EthernetPriorityMark are greater than zero, then the outgoing frames MUST be tagged. Otherwise, the outgoing frames MAY be tagged or untagged. | -1 |
| EthernetPriorityMark | int[-1:] | W | Ethernet priority code (as defined in 802.1D) to be used for outgoing SIP signaling packets for this profile. A value of -1 indicates the default value is to be used.<br>If either the VLANIDMark or EthernetPriorityMark are greater than zero, then the outgoing frames MUST be tagged. Otherwise, the outgoing frames MAY be tagged or untagged. | -1 |
| SIPEventSubscribeNumberOfElements | unsignedInt | - | Indicates the number of EventSubscribe objects. | 0 |
| SIPResponseMapNumberOfElements | unsignedInt [0:999] | - | Indicates the number of SIPResponseMap objects. | 0 |
| .VoiceService.{i}.VoiceProfile.{i}.SIP. EventSubscribe.{i} | object | W | Table to specify the SIP events to which the CPE MUST subscribe.<br>If this table is supported, the capability VoiceService.{i}.Capabilities.SIP.Event-Subscription MUST be equal to true and the parameter SIPEventSubscribeNumberOfElements in the parent object MUST be present. | - |
| Event | string(32) | W | SIP event name to appear in the EVENT header of the SIP SUBSCRIBE request. | <Empty> |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| Notifier | string(256) | W | Host name or IP address of the event notify server. | <Empty> |
| NotifierPort | unsignedInt [0:65535] | W | Destination port to be used in connecting to the event notifier. | 0 |
| NotifierTransport | string | W | Transport protocol to be used in connecting to the event notifier. Must be chosen from among the transports supported, as indicated by VoiceService.{i}.Capabilities.SIP.Transports. Enumeration of:<br>"UDP"<br>"TCP"<br>"TLS"<br>"SCTP" | "UDP" |
| ExpireTime | unsignedInt | W | Subscription refresh timer, in seconds. | - |
| .VoiceService.{i}.VoiceProfile.{i}.SIP.-ResponseMap.{i} | object | W | Each entry in this table specifies the tone and message to be provided to the user for a particular SIP Response received (normally 4xx and 5xx).<br>If this table is supported, the capability VoiceService.{i}.Capabilities.SIP.ResponseMap MUST be equal to true and the parameter SIP-ResponseMapNumberOfElements in the parent object MUST be present.<br>Applicable only for a VoIP endpoint. | - |
| SIPResponseNumber | unsignedInt [100:999] | W | The SIP Response code number. | 100 |
| TextMessage | string(64) | W | The message to be provided on the screen or display of the VoIP device when the SIP response is received.<br>If this parameter is non-empty, display of this text preempts display of the ToneText associated with the specified tone. If this parameter is empty, the ToneText for the specified tone, if any, is displayed instead.<br>This parameter is applicable only for VoIP devices capable text display. | <Empty> |
| Tone | unsignedInt | W | The tone to be played to the user when the SIP response is received.<br>The value corresponds to EntryID of an entry in the table VoiceService.{i}.VoiceProfile.{i}.Tone.-Description. A value of zero, or any value that does not match a valid EntryID, results in no tone played.<br>If VoiceService.{i}.Capabilities.ToneGeneration is equal to false, no tone is played regardless of the value of this parameter. | 0 |
| .VoiceService.{i}.VoiceProfile.{i}.MGCP. | object | - | Voice profile parameters that are specific to MGCP call signaling. | -[5] |
| CallAgent1 | string(256) | W | Host name or IP address of the main MGCP call agent. | <Empty> |
| CallAgentPort1 | unsignedInt [0:65535] | W | Destination port to be used in connecting with the main MGCP call agent. | 0 |
| CallAgent2 | string(256) | W | Host name or IP address of the backup MGCP call agent. | <Empty> |
| CallAgentPort2 | unsignedInt [0:65535] | W | Destination port to be used in connecting with the backup MGCP call agent. | 0 |

[5] Creation of this object occurs on specification of MGCP as the SignalingProtocol in the parent object.

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| RetranIntervalTimer | unsignedInt [0:65535] | W | Message retransfer interval, in seconds. | 1 |
| MaxRetranCount | unsignedInt [0:65535] | W | Max number of message retransfers. | 10 |
| RegisterMode | string | W | Register mode. Enumeration of: “Wildcard” “Individual” | “Wildcard” |
| LocalPort | unsignedInt [0:65535] | W | Port listening for incoming call control signaling. | 0 |
| Domain | string(256) | W | CPE domain string. If empty, the CPE SHOULD use its IP address. | <Empty> |
| User | string(64) | W | User string used in accessing the call agent. | <Empty> |
| DSCPMark | unsignedInt [0:63] | W | Diffserv code point to be used for outgoing MGCP signaling packets. | - |
| VLANIDMark | int[-1:] | W | VLAN ID (as defined in 802.1Q) to be used for outgoing MGCP signaling packets for this profile. A value of -1 indicates the default value is to be used. If either the VLANIDMark or EthernetPriorityMark are greater than zero, then the outgoing frames MUST be tagged. Otherwise, the outgoing frames MAY be tagged or untagged. | -1 |
| EthernetPriorityMark | int[-1:] | W | Ethernet priority code (as defined in 802.1D) to be used for outgoing MGCP signaling packets for this profile. A value of -1 indicates the default value is to be used. If either the VLANIDMark or EthernetPriorityMark are greater than zero, then the outgoing frames MUST be tagged. Otherwise, the outgoing frames MAY be tagged or untagged. | -1 |
| AllowPiggybackEvents | boolean | W | Indicates whether or not piggyback events are allowed to the MGCP call agent. | - |
| SendRSIPImmediately | boolean | W | Indicates whether or not to send RSIP immediately on restart. | - |
| .VoiceService.{i}.VoiceProfile.{i}.H323. | object | - | Voice profile parameters that are specific to H.323 call signaling. | -[6] |
| Gatekeeper | string | W | Host name or IP address of H.323 Gatekeeper. | <Empty> |
| GatekeeperPort | unsignedInt | W | Destination port to be used in connecting to the H.323 Gatekeeper. | 1719 |
| GatekeeperID | string | W | Gatekeeper ID. | <Empty> |
| TimeToLive | unsignedInt | W | In seconds, defines the TimeToLive specification in the registration with the Gatekeeper. | 86400 |
| H235Authentication | boolean | W | Enables or disables usage of H.235 security (H.235v2 Annex D, baseline security profile). | False |
| AuthPassword | string | W | Password to be used when H.235 is enabled. When read, this parameter returns an empty string, regardless of the actual value. | <Empty> |
| SendersID | string | W | In ITU-T based H.235 authentication, the sendersID is the ID of the gateway as received from the Gatekeeper. As long as the endpointID is not received from the Gatekeeper, the sendersID will be applied as configured here. The generalID is the GatekeeperID. | <Empty> |
| DSCPMark | unsignedInt | W | Diffserv code point to be used for outgoing H.323 | 0 |

[6] Creation of this object occurs on specification of H.323 as the SignalingProtocol in the parent object.

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | [0:63] | | signaling packets. | |
| VLANIDMark | int[-1:] | W | VLAN ID (as defined in 802.1Q) to be used for outgoing H.323 signaling packets for this profile. A value of -1 indicates the default value is to be used.<br>If either the VLANIDMark or EthernetPriorityMark are greater than zero, then the outgoing frames MUST be tagged. Otherwise, the outgoing frames MAY be tagged or untagged. | -1 |
| EthernetPriorityMark | int[-1:] | W | Ethernet priority code (as defined in 802.1D) to be used for outgoing H.323 signaling packets for this profile. A value of -1 indicates the default value is to be used.<br>If either the VLANIDMark or EthernetPriorityMark are greater than zero, then the outgoing frames MUST be tagged. Otherwise, the outgoing frames MAY be tagged or untagged. | -1 |
| .VoiceService.{i}.VoiceProfile.{i}.RTP. | object | - | Voice profile parameters related to the voice stream sent via RTP.<br>Applicable only for a VoIP endpoint. | |
| LocalPortMin | unsignedInt [0:65535] | W | Base of port range to be used for incoming RTP streams for this profile. | 0 |
| LocalPortMax | unsignedInt [0:65535] | W | Top of port range to be used for incoming RTP streams for this profile. | 0 |
| DSCPMark | unsignedInt [0:63] | W | Diffserv code point to be used for outgoing RTP packets for this profile. It is RECOMMENDED that by default the DSCP for RTP traffic be set to the value to indicate EF traffic. | - |
| VLANIDMark | int[-1:] | W | VLAN ID (as defined in 802.1Q) to be used for outgoing RTP packets for this profile. A value of -1 indicates the default value is to be used.<br>If either the VLANIDMark or EthernetPriorityMark are greater than zero, then the outgoing frames MUST be tagged. Otherwise, the outgoing frames MAY be tagged or untagged. | -1 |
| EthernetPriorityMark | int[-1:] | W | Ethernet priority code (as defined in 802.1D) to be used for outgoing RTP packets for this profile. A value of -1 indicates the default value is to be used.<br>If either the VLANIDMark or EthernetPriorityMark are greater than zero, then the outgoing frames MUST be tagged. Otherwise, the outgoing frames MAY be tagged or untagged. | -1 |
| TelephoneEventPayloadType | unsignedInt [0:128] | W | Payload type to be used for RTP telephone events.<br>This parameter indicates the payload type to be used for DTMF events if RFC 2833 transmission of DTMF information is in use. | - |
| .VoiceService.{i}.VoiceProfile.{i}.RTP.RTCP. | object | - | Voice profile parameters related to RTCP.<br>If this object is supported, the capability VoiceService.{i}.Capabilities.RTCP MUST be equal to true.<br>Applicable only for a VoIP endpoint. | - |
| Enable | boolean | W | Enable or disable RTCP. | - |
| TxRepeatInterval | unsignedInt [1:] | W | Transmission repeat interval, in milliseconds. | - |
| LocalCName | string(64) | W | Local Cname (canonical name). | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| VoiceService.{i}.VoiceProfile.{i}.RTP.SRTP. | object | - | Voice profile parameters for secure voice transmission via SRTP.<br>If this object is supported, the capability VoiceService.{i}.Capabilities.SRTP MUST be equal to true.<br>Applicable only for a VoIP endpoint. | - |
| Enable | boolean | W | Enable or disable the use of SRTP.<br>If RTCP is enabled, a true value of this parameter also implies the use of SRTCP. | - |
| KeyingMethods | string(256) | W | Comma-separated list of keying methods that may be used. The value MUST be a subset of those listed in the parameter VoiceService.{i}.-Capabilities.SRTPKeyingMethods. | [See note 7] |
| EncryptionKeySizes | string(256) | W | Comma-separated list of encryption key sizes that may be used. The value MUST be a subset of those listed in the parameter VoiceService.{i}.-Capabilities.SRTPEncryptionKeySizes. | [See note 8] |
| VoiceService.{i}.VoiceProfile.{i}.RTP.-Redundancy | object | - | Voice profile parameters for RTP payload redundancy as defined by RFC 2198.<br>If this object is supported, the capability VoiceService.{i}.Capabilities.Redundancy MUST be equal to true.<br>Applicable only for a VoIP endpoint. | - |
| Enable | boolean | W | Enable or disable the use of RTP payload redundancy as defined by RFC 2198. | False |
| PayloadType | unsignedInt [0:127] | W | The Payload Type of RTP packet using RFC 2198. Values should be within the range of dynamic Payload Types (96-127). | 0 |
| BlockPayloadType | unsignedInt [0:127] | - | Block Payload Type of redundancy packet. | 0 |
| FaxAndModemRedundancy | int[-1:5] | W | Specifies the redundancy number for fax and modem pass-through data transmissions.<br>A non-negative value indicates that RFC 2198 is to be used for fax and modem pass-through data. The value indicates the number of redundant copies to be transmitted (the total number transmitted is one plus this value).<br>A value of -1 indicates RFC 2198 is not to be used for fax and modem pass-through data.<br>If the optional parameter ModemRedudancy is present, then FaxAndModemRedudancy applies *only* to fax transmissions, but not to modem transmissions. | -1 |
| ModemRedundancy | int[-1:5] | W | Specifies the redundancy number for modem pass-through data transmissions.<br>A non-negative value indicates that RFC 2198 is to be used for modem pass-through data. The value indicates the number of redundant copies to be transmitted (the total number transmitted is one plus this value).<br>A value of -1 indicates RFC 2198 is not to be used for modem pass-through data. | -1 |
| DTMFRedundancy | int[-1:5] | W | Specifies the redundancy number for DTMF transmissions. | -1 |

[7] By default this parameter MUST have the value of VoiceDevice.{i}.Capabilities.SRTPKeyingMethods.
[8] By default this parameter MUST have the value of VoiceDevice.{i}.Capabilities.SRTPEncryptionKey-Sizes.

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | A non-negative value indicates that RFC 2198 is to be used for DTMF. The value indicates the number of redundant copies to be transmitted (the total number transmitted is one plus this value).<br>A value of -1 indicates RFC 2198 is not to be used for DTMF. | |
| VoiceRedundancy | int[-1:5] | W | Specifies the redundancy number for general voice transmissions.<br>A non-negative value indicates that RFC 2198 is to be used for voice. The value indicates the number of redundant copies to be transmitted (the total number transmitted is one plus this value).<br>A value of -1 indicates RFC 2198 is not to be used for voice. | -1 |
| MaxSessionsUsingRedundancy | unsignedInt | W | The maximum number of sessions using RFC 2198 payload redundancy simultaneously in this VoiceProfile.<br>A value of zero indicates no explicit limit on the number of sessions using redundancy. | 0 |
| .VoiceService.{i}.VoiceProfile.{i}.-NumberingPlan. | object | - | This object contains information related the numbering plan.<br>This object is applicable only if the device supports a dialing mechanism for which a number plan is needed (for example, a device with an explicit Dial button may not need to be aware of the dialing plan) and if the device does not already support a numbering plan mechanism for this profile (e.g., in-band via MGCP).<br>If this object is supported, the capability VoiceService.{i}.Capabilities.NumberingPlan MUST be equal to true.<br>Applicable only for a VoIP endpoint. | - |
| MinimumNumberOfDigits | unsignedInt [1:40] | W | This is the minimum number of digits that must be collected before an outgoing request (e.g., a SIP INVITE) can be initiated.<br>If "End of Dialing" (refer to the definition of the InterDigitTimer) occurs before the minimum number of digits has been reached then the number will be considered incomplete and no request will be initiated.<br>In practice, searching the "PrefixInfo" list should only commence once the minimum number of digits (as specified by this parameter) has been received. | - |
| MaximumNumberOfDigits | unsignedInt [1:40] | W | This is the maximum number of digits that may be collected before an outgoing request (e.g., a SIP INVITE) must be initiated. Any additional dialed digits will be ignored. This parameter is only used in the case that no match in the "PrefixInfo" list has been found. | - |
| InterDigitTimerStd | unsignedInt [1:50000] | W | This timer is the maximum allowable time (expressed in milliseconds) between the dialing of digits. This timer is restarted every time a digit is dialed. Expiration of this timer indicates "End of Dialing". | - |
| InterDigitTimerOpen | unsignedInt [1:50000] | W | This timer is the maximum allowable time (expressed in milliseconds) between the dialing of digits once the minimum number of digits defined on a prefix based has been reached.<br>This timer is only applicable to "open numbering", where the exact number of digits for a prefix is not | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | known. | |
| InvalidNumberTone | unsignedInt | W | The tone that should be provided to the user when the number dialed is determined to be invalid.<br>The value corresponds to EntryID of an entry in the table VoiceService.{i}.VoiceProfile.{i}.Tone.-Description. A value of zero, or any value that does not match a valid EntryID, results in no tone played.<br>If VoiceService.{i}.Capabilities.ToneGeneration is equal to false, no tone is played regardless of the value of this parameter. | - |
| PrefixInfoMaxEntries | unsignedInt | - | This is the maximum number of instances of the PrefixInfo object that can be supported. | - |
| PrefixInfoNumberOfEntries | unsignedInt | - | Indicates the number of instances of PrefixInfo. | - |
| .VoiceService.{i}.VoiceProfile.{i}.-NumberingPlan.PrefixInfo.{i}. | object | W | Each entry in this table contains information related to an individual prefix in the numbering plan. The number of prefixes is defined by the PrefixInfoNumberOfEntries paramter.<br>It is anticipated that once the minimum number of digits has been received, the VoIP device will search this prefix list every time a new digit is received. If no new entry is found, then the object that was previously found will be used instead.<br>If this table is supported, the parameters Prefix-InfoMaxEntries and PrefixInfoNumberOfEntries in the parent object MUST be present. | -[9] |
| PrefixRange | string(42) | W | This is a string representation of a range of prefixes. Each prefix consists of a "From" part consisting of 1 to n digits (string representation) followed by an optional "To" part consisting of exactly one digit prefixed by a "-" symbol.<br>It should be noted that only the characters "0-9", "*": and "#" can be used to represent the "From" and "To" parts of the prefix range.<br>A further constraint is that the "To" digit MUST always be numerically greater than the last digit of the "From" part.<br>Examples:<br>02<br>031-5<br>032<br>0325<br>*#34<br>#22 | <Empty> |
| PrefixMinNumberOfDigits | unsignedInt [1:40] | W | This is the minimum number of allowable digits for the prefix range. Once the minimum number of digits has been reached, the "InterDigitTimer-Open" will be used instead of the "InterDigitTimer-Std".<br>If the minimum number of digits has been reached and the inter-digit timer expires, an outgoing request should be initiated. | 1 |
| PrefixMaxNumberOfDigits | unsignedInt [1:40] | W | This is the maximum number of allowable digits for the prefix range. Once the number of digits received reaches this value an outgoing request should be initiated. | 1 |
| NumberOfDigitsToRemove | unsignedInt | W | If this parameter has a non-zero value, the | 0 |

[9] The defaults given for this object apply only to explicit creation of an instance of this object and not to automatic creation of instances of this object due to creation of a parent object.

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | [0:40] | | specified number of digits will be removed from the internal digit buffer (which contains the dialed digits) from the position specified by the "PosOf-DigitToRemove" parameter.<br>Subsequently a search of the PrefixInfo list for a matching prefix using the modified number should be performed. Note that this parameter does not have any impact on the number sent in the outgoing request - but is instead only used for searching within the Numbering Plan.<br>This parameter has no effect if it is set to 0.<br>This parameter is provided to handle Carrier override and other codes that may prefix standard numbers and to ensure that the correct "End of Dialing" can be specified without significant data duplication. | |
| PosOfDigitsToRemove | unsignedInt [0:40] | W | This parameter is used in conjunction with the NumberOfDigitsToRemove parameter. It specifies the position within the internal digit buffer from which the digits are to be removed. | 0 |
| DialTone | unsignedInt | W | The tone to be played by the VoIP device when the user has dialed exactly the same digits as defined in the prefix. The VoIP device will cease playing the tone once an additional digit has been dialed.<br>The value corresponds to EntryID of an entry in the table VoiceService.{i}.VoiceProfile.{i}.Tone.-Description. A value of zero, or any value that does not match a valid EntryID, results in no tone played.<br>If VoiceService.{i}.Capabilities.ToneGeneration is equal to false, no tone is played regardless of the value of this parameter. | 0 |
| FacilityAction | string(64) | W | This is a string representing a Facility Action implemented by the VoIP device.<br>Appendix A lists a set of defined values for this string.<br>The parameter MAY instead indicate a vendor-specific FacilityAction, which MUST use the format defined in [3].<br>An empty or unrecognized string (i.e., a Facility Action not supported by the CPE) should be treated as a normal outgoing request. | <Empty> |
| FacilityActionArgument | string(256) | W | Optional argument associated with the specified FacilityAction. The interpretation of the argument is dependent on the specific FacilityAction. Where used, the value is specified in Appendix A in the definition of the corresponding FacilityAction value. | <Empty> |
| .VoiceService.{i}.VoiceProfile.{i}.Tone. | object | - | This object defines the contents of the tones and announcements generated locally by the VoIP device.<br>If this object is supported, the capability VoiceService.{i}.Capabilities.ToneGeneration MUST be equal to true.<br>Applicable only for a VoIP endpoint. | - |
| EventNumberOfEntries | unsignedInt | - | Indicates the number of entries in the Event table in the Tone object. | - |
| DescriptionNumberOfEntries | unsignedInt | - | Indicates the number of entries in the Description table in the Tone object. | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| PatternNumberOfEntries | unsignedInt | - | Indicates the number of entries in the Pattern table in the Tone object. | - |
| .VoiceService.{i}.VoiceProfile.{i}.Tone.-Event.{i}. | object | - | Table of events for which a tone is defined. The table is pre-populated with the list of events for which the CPE supports definition of tones.<br>If this table is supported, the parameter Event-NumberOfEntries in the parent object MUST be present. | - |
| Function | string(64) | - | The event for which the tone is to apply. Enumeration of:<br>"Busy"<br>"Confirmation"<br>"Dial"<br>"MessageWaiting"<br>"OffHookWarning"<br>"RingBack"<br>"ReOrder"<br>"Stutterdial"<br>"CallWaiting1"<br>"CallWaiting2"<br>"CallWaiting3"<br>"CallWaiting4"<br>"AlertingSignal"<br>"SpecialDial"<br>"SpecialInfo"<br>"Release"<br>"Congestion"<br>"UserDefined1"<br>"UserDefined2"<br>"UserDefined3"<br>"UserDefined4"<br>The parameter MAY instead indicate a vendor-specific event name, which MUST use the format defined in [3]. | - |
| ToneID | unsignedInt | W | The EntryID of the entry in the Description table for the tone to be associated with the given event.<br>A value of zero indicates no tone is to be played for this event. | - |
| .VoiceService.{i}.VoiceProfile.{i}.Tone.-Description.{i}. | object | W | Each entry in this table defines the contents of an individual tone.<br>If ability to add, delete, and modify entries in this table is supported, the capability VoiceService.{i}.-Capabilities.ToneDescriptionsEditable MUST be equal to true.<br>If this table is supported, the parameter DescriptionNumberOfEntries in the parent object MUST be present. | -[10] |
| EntryID | unsignedInt [1:] | - | Unique identifier of this tone. Assigned by the CPE upon creation of the entry. | - |
| ToneEnable | boolean | W | Enables or disables the tone entry. If a disabled | False |

[10] The defaults given for this object apply only to explicit creation of an instance of this object and not to automatic creation of instances of this object due to creation of a parent object.

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | tone entry is referenced, the result is that no tone is played. | |
| ToneName | string(64) | W | Name of the tone.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true. | <Empty> |
| TonePattern | unsignedInt | W | The EntryID of the entry in the Pattern table that begins the tone pattern for this tone.<br>If the tone is specified by a tone file instead of a tone pattern, this parameter MUST be set to zero.<br>This parameter is applicable only if VoiceService.-{i}.Capabilities.PatternBasedToneGeneration is equal to true.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true. | 0 |
| ToneFile | string(256) | W | This is the file name of a tone file that has been downloaded to the CPE. The download may have occurred via the TR-069 Download mechanism or by some other means.<br>If the tone is specified by a tone pattern instead of a tone file, this parameter MUST be empty.<br>This parameter is applicable only if VoiceService.-{i}.Capabilities.FileBasedToneGeneration is equal to true.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true. | <Empty> |
| ToneRepetitions | unsignedInt [0:255] | W | The default number of times the data in the ToneFile should be repeated. If the value 0 (zero) is specified then the Tone should be played indefinitely.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true. | 0 |
| ToneText | string(64) | W | The text to be displayed by on the screen of the VoIP device when the tone is played and no specific error message has been provided.<br>This parameter is applicable only for VoIP devices capable text display. | <Empty> |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| .VoiceService.{i}.VoiceProfile.{i}.Tone.-Pattern.{i}. | object | W | Each entry in the table defines a single phase in an overall tone pattern. Each phase identifies the entry that corresponds to the next phase.<br>Each entry in the table refers to the entry that corresponds to the next phase of the pattern. The table MAY be set up such that entries form loops, or MAY end after a finite sequence.<br>If this object is supported, the capability VoiceService.{i}.Capabilities.PatternBasedTone-Generation MUST be equal to true, and the parameter PatternNumberOfEntries in the parent object MUST be present.<br>If ability to add, delete, and modify entries in this table is supported, the capability VoiceService.{i}.-Capabilities.ToneDescriptionsEditable MUST be equal to true. | -[11] |
| EntryID | unsignedInt [1:] | W | Identifier of a tone-pattern entry. The value MUST be unique within this table. (If two entries have the same value, only the first instance is used.)<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true.<br>Note: when ToneDescriptionsEditable is true, this parameter is editable so that the NextEntryID values for each table entry can be pre-assigned for a series of associated table entries rather than requiring the ACS to set the value according to an ID assigned dynamically upon creation of each entry. | - |
| ToneOn | boolean | W | Whether or not a tone is on during this phase of the pattern. If the value is false, the frequency and power parameters in this entry MUST be ignored.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true. | False |
| Frequency1 | unsignedInt [0:4095] | W | First tone frequency in hertz.<br>A value of zero indicates this tone component is not used.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true. | 0 |
| Power1 | int | W | First tone power level in units of 0.1 dBm0.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true. | 0 |
| Frequency2 | unsignedInt [0:4095] | W | Second tone frequency in hertz.<br>A value of zero indicates this tone component is not used.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true. | 0 |
| Power2 | int | W | Second tone power level in units of 0.1 dBm0.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true. | 0 |

[11] The defaults given for this object apply only to explicit creation of an instance of this object and not to automatic creation of instances of this object due to creation of a parent object.

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| Frequency3 | unsignedInt [0:4095] | W | Third tone frequency in hertz.<br>A value of zero indicates this tone component is not used.<br>This parameter is required to be editable only if the parameter is supported for reading and VoiceService.{i}.Capabilities.ToneDescriptions-Editable is equal to true. | 0 |
| Power3 | int | W | Third tone power level in units of 0.1 dBm0.<br>This parameter is required to be editable only if the parameter is supported for reading and VoiceService.{i}.Capabilities.ToneDescriptions-Editable is equal to true. | 0 |
| Frequency4 | unsignedInt [0:4095] | W | Fourth tone frequency in hertz.<br>A value of zero indicates this tone component is not used.<br>This parameter is required to be editable only if the parameter is supported for reading and VoiceService.{i}.Capabilities.ToneDescriptions-Editable is equal to true. | 0 |
| Power4 | int | W | Fourth tone power level in units of 0.1 dBm0.<br>This parameter is required to be editable only if the parameter is supported for reading and VoiceService.{i}.Capabilities.ToneDescriptions-Editable is equal to true. | 0 |
| ModulationFrequency | unsignedInt [0:4095] | W | Modulation frequency in hertz.<br>A value of zero indicates this tone component is not used.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true. | 0 |
| ModulationPower | int | W | Modulation power level in units of 0.1 dBm0.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true. | 0 |
| Duration | unsignedInt | W | The duration of this phase of the tone pattern, in milliseconds.<br>A value of zero indicates an unlimited duration.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true. | 0 |
| NextEntryID | unsignedInt | W | The EntryID for the next phase of the tone pattern, after the specified Duration of this phase has completed.<br>A value of zero indicates that the tone pattern is to terminate after the current phase is completed.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Tone-DescriptionsEditable is equal to true. | 0 |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| .VoiceService.{i}.VoiceProfile.{i}.ButtonMap. | object | - | This object is provided to permit the purpose of the CPE buttons and function keys to be defined via the ACS.<br>Support of this object is appropriate only for a device that has programmable buttons in its user interface.<br>If this object is supported, the capability VoiceService.{i}.Capabilities.ButtonMap MUST be equal to true.<br>Applicable only for a VoIP endpoint. | - |
| NumberOfButtons | unsignedInt | - | Indicates the number of Button objects | - |
| .VoiceService.{i}.VoiceProfile.{i}.ButtonMap.-Button.{i}. | object | - | Each entry in this table specifies the purpose of each programmable CPE button / function key and whether the user has permission to reprogram the button. | - |
| ButtonName | string(16) | - | Name of the Button. | - |
| FacilityAction | string(64) | W | This is an optional parameter that should only be specified for buttons related to a particular Facility Action (e.g., invocation of "Call Return") implemented by the VoIP device.<br>Appendix A lists a set of defined values for this string.<br>The parameter MAY instead indicate a vendor-specific FacilityAction, which MUST use the format defined in [3].<br>An empty or unrecognized string (i.e. a Facility Action not supported by the CPE) should be treated as no Facility Action to be taken.<br>Note that If this parameter is specified (non-empty) then the QuickDialNumber parameter (see below) should be an empty string. | - |
| FacilityActionArgument | string(256) | W | Optional argument associated with the specified FacilityAction. The interpretation of the argument is dependent on the specific FacilityAction. Where used, the value is specified in Appendix A in the definition of the corresponding FacilityAction value. | <Empty> |
| QuickDialNumber | string(40) | W | This is a string representing a quick dial destination number. Only the characters '0-9', '*' and '#' can be used.<br>Note that If this parameter is specified (non-empty) then the FacilityAction parameter (see above) should be an empty string. | - |
| ButtonMessage | string(64) | W | This string represents the message to be displayed on the screen when the button or function key is pressed. | - |
| UserAccess | boolean | W | This parameter indicates whether the user has permission to program the button or function key. If this parameter is set to TRUE then the FacilityAction, QuickDialNumber and ButtonMessage parameters MUST all be empty. | - |
| .VoiceService.{i}.VoiceProfile.{i}.FaxT38 | object | - | T.38 Fax information for devices that support T.38 relay.<br>If this object is supported, the capability VoiceService.{i}.Capabilities.FaxT38 MUST be equal to true.<br>Applicable only to a VoIP endpoint. | - |
| Enable | boolean | W | Enable or disable the use of T.38. | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| BitRate | unsignedInt | W | Maximum data rate for fax. Enumeration of the following values:<br>2400<br>4800<br>7200<br>9600<br>12000<br>14400<br>33600 | - |
| HighSpeedPacketRate | unsignedInt | W | The rate at which high speed data will be sent across the network, in milliseconds. Enumeration of the following values:<br>10<br>20<br>30<br>40 | - |
| HighSpeedRedundancy | unsignedInt<br>[0:3] | W | Specifies the packet-level redundancy for high-speed data transmissions (i.e., T.4 image data). The value MUST be in the range 0 through 3. | - |
| LowSpeedRedundancy | unsignedInt<br>[0:5] | W | Specifies the packet-level redundancy for low-speed data transmissions (i.e., T.30 handshaking information). The value MUST be in the range 0 through 5. | - |
| TCFMethod | string | W | The method with which data is handled over the network. Enumeration of:<br>"Local"<br>"Network" | - |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}. | object | W | Object associated with a distinct voice line. Support for adding and removing lines is conditional on whether the CPE supports more than one line in total as indicated by VoiceService.{i}.Capabilities.MaxLineCount. By default, on creation of a given VoiceProfile, a single Line object MUST be present, initially in the disabled state.<br>Applicable only for a VoIP endpoint. | - |
| Enable | string | W | Enables or disables this line, or places it into a quiescent state. Enumeration of:<br>"Disabled"<br>"Quiescent"<br>"Enabled"<br>On creation, a line MUST be in the Disabled state.<br>In the Quiescent state, in-progress sessions remain intact, but no new sessions are allowed. Support for the Quiescent state in a CPE is optional. If this parameter is set to "Quiescent" in a CPE that does not support the Quiescent state, it MUST treat it the same as the Disabled state (and indicate Disabled in the Status parameter). | False |
| DirectoryNumber | string(32) | W | Directory number associated with this line. May be used to identify the line to the user.<br>In case of H.323 signaling, this MUST be an E.164 number. | <Empty> |
| Status | string | - | Indicates the status of this line. Enumeration of:<br>"Up" | "Disabled" |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | "Initializing"<br>"Registering"<br>"Unregistering"<br>"Error"<br>"Testing"<br>"Quiescent"<br>"Disabled" | |
| CallState | string | - | Indicates the call state for this line. Enumeration of:<br>"Idle"<br>"Calling"<br>"Ringing"<br>"Connecting"<br>"InCall"<br>"Hold"<br>"Disconnecting" | "Idle" |
| PhyReferenceList | string(32) | W | A comma separated list of Physical Interface Identifiers that this Line is associated with. Each item corresponds to the value of the InterfaceID parameter in a particular instance of the VoiceService.{i}.PhyInterface.{i}. table. | - |
| RingMuteStatus | boolean | - | Whether or not ringing has been locally muted. Applicable only if the line is associated with a single telephony device for which ring can be muted. | - |
| RingVolumeStatus | unsignedInt [0:100] | - | Percent value of current ringer volume level. Applicable only if the line is associated with a single telephony device for which the ringer volume can be controlled. | - |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.SIP. | object | - | Voice line parameters that are specific to SIP call signaling. | - |
| AuthUserName | string(128) | W | Username used to authenticate the connection to the server. | <Empty> |
| AuthPassword | string(128) | W | Password used to authenticate the connection to the server.<br>When read, this parameter returns an empty string, regardless of the actual value. | <Empty> |
| URI | string(389) | W | URI by which the user agent will identify itself for this line.<br>If empty, the actual URI used in the SIP signaling SHOULD be automatically formed by the CPE as:<br>"sip:UserName@Domain"<br>Where UserName is username given for this line (VoiceService.{i}.VoiceProfile.{i}.Line.{i}.SIP.Auth-UserName), and Domain is the domain given for this profile (VoiceService.{i}.VoiceProfile.{i}.SIP.-UserAgentDomain). If this domain parameter is empty, then the IP address of the CPE SHOULD be used for the domain.<br>If URI is non-empty, but is a SIP or SIPS URI that contains no "@" character, then the actual URI used in the SIP signaling SHOULD be automatically formed by the CPE by appending this parameter with an "@" character followed by the domain given for this profile (VoiceService.{i}.-VoiceProfile.{i}.SIP.UserAgentDomain). If this domain parameter is empty, then the IP address | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | of the CPE SHOULD be used for the domain. | |
| SIPEventSubscribeNumberOfElements | unsignedInt | - | Indicates the number of EventSubscribe objects. | - |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.SIP. EventSubscribe.{i} | object | - | Table of SIP Events automatically populated by the CPE with each of the SIP event subscriptions in the table VoiceService.{i}.VoiceProfile.{i}.SIP. EventSubscribe.{i}. This table allows specification of the authentication credentials needed for each event subscription.<br>If this table is supported, the capability VoiceService.{i}.Capabilities.SIP.Event-Subscription MUST be equal to true and the parameter SIPEventSubscribeNumberOfElements in the parent object MUST be present. | - |
| Event | string(32) | - | SIP event name corresponding to the value given in the table VoiceService.{i}.VoiceProfile.{i}.SIP. EventSubscribe.{i}. | - |
| AuthUserName | string(128) | W | Username used to authenticate the connection to the event notify server. | <Empty> |
| AuthPassword | string(128) | W | Password used to authenticate the connection to the event notify server.<br>When read, this parameter returns an empty string, regardless of the actual value. | <Empty> |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.-MGCP. | object | - | Voice line parameters that are specific to MGCP call signaling. | - |
| LineName | string(32) | W | Used to identify the line when using MGCP signaling. If empty, the CPE SHOULD use the default names "aaln/1", etc. | <Empty> |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.H323. | object | - | Voice line parameters that are specific to H.323 call signaling. | - |
| H323ID | string(32) | W | The H323ID assigned to the line. | <Empty> |
| .VoiceService.{i}.VoiceProfile.{i}.Line.-{i}.Ringer. | object | - | This object defines the ring sequences generated by the VoIP device.<br>If this object is supported, the capability VoiceService.{i}.Capabilities.RingGeneration MUST be equal to true. | - |
| EventNumberOfEntries | unsignedInt | - | Number of entries in the Event table in the Ringer object. | - |
| DescriptionNumberOfEntries | unsignedInt | - | Number of entries in the Description table in the Ringer object. | - |
| PatternNumberOfEntries | unsignedInt | - | Number of entries in the Pattern table in the Ringer object. | - |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.-Ringer.Event.{i}. | object | - | Table of events for which a ring pattern is defined. The table is pre-populated with the complete list of events for which the CPE supports definition of ring patterns.<br>If this table is supported, the parameter Event-NumberOfEntries in the parent object MUST be present. | - |
| Function | string(64) | - | The event for which the ring pattern is to apply. Enumeration of:<br>"Default"<br>"RingSplash"<br>The parameter MAY instead indicate a vendor-specific event name, which MUST use the format defined in [3]. | - |
| RingID | unsignedInt | W | The EntryID of the entry in the Description table for the ring to be associated with the given event. | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | A value of zero indicates ringing is to be disabled for this event. | |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.-Ringer.Description.{i}. | object | W | Each entry in this table defines the contents of an individual ring specification.<br>If ability to add, delete, and modify entries in this table is supported, the capability VoiceService.{i}.-Capabilities.RingDescriptionsEditable MUST be equal to true.<br>If this table is supported, the parameter DescriptionNumberOfEntries in the parent object MUST be present. | -[12] |
| EntryID | unsignedInt [1:] | - | Unique identifier of this ring description. Assigned by the CPE upon creation of the entry. | - |
| RingEnable | boolean | W | Enables or disables the ring description entry. If a disabled ring description entry is referenced, the result is that no ring is played. | False |
| RingName | string(64) | W | Name of the ring description.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Ring-DescriptionsEditable is equal to true. | <Empty> |
| RingPattern | unsignedInt | W | The EntryID of the entry in the Pattern table that begins the ring pattern for this ring description.<br>If the ring is specified by a ring file instead of a ring pattern, this parameter MUST be set to zero.<br>This parameter is applicable only if VoiceService.-{i}.Capabilities.PatternBasedRingGeneration is equal to true.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Ring-DescriptionsEditable is equal to true. | 0 |
| RingFile | string(256) | W | This is the file name of a ring file that has been downloaded to the CPE. The download may have occurred via the TR-069 Download mechanism or by some other means.<br>If the ring is specified by a ring pattern instead of a ring file, this parameter MUST be empty.<br>This parameter is applicable only if VoiceService.-{i}.Capabilities.FileBasedRingGeneration is equal to true.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Ring-DescriptionsEditable is equal to true. | <Empty> |

[12] The defaults given for this object apply only to explicit creation of an instance of this object and not to automatic creation of instances of this object due to creation of a parent object.

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.-Ringer.Pattern.{i}. | object | W | Each entry in the table defines a single phase in an overall ring pattern. Each phase identifies the entry that corresponds to the next phase.<br>Each entry in the table refers to the entry that corresponds to the next phase of the pattern. The table MAY be set up such that entries form loops, or MAY end after a finite sequence.<br>If this object is supported, the capability VoiceService.{i}.Capabilities.PatternBasedRing-Generation MUST be equal to true and the parameter PatternNumberOfEntries in the parent object MUST be present.<br>If ability to add, delete, and modify entries in this table is supported, the capability VoiceService.{i}.-Capabilities.RingPatternEditable MUST be equal to true. | -[13] |
| EntryID | unsignedInt [1:] | W | Identifier of a ring-pattern entry. The value MUST be unique within this table. (If two entries have the same value, only the first instance is used.)<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Ring-PatternEditable is equal to true.<br>Note: when RingPatternEditable is true, this parameter is editable so that the NextEntryID values for each table entry can be pre-assigned for a series of associated table entries rather than requiring the ACS to set the value according to an ID assigned dynamically upon creation of each entry. | - |
| RingerOn | boolean | W | True indicates the ringer is to be on for the specified period. False indicates the ringer is to be off for the specified period.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Ring-PatternEditable is equal to true. | False |
| Duration | unsignedInt | W | The duration of this phase of the ring pattern, in milliseconds.<br>A value of zero indicates an unlimited duration.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Ring-PatternEditable is equal to true. | 0 |
| NextEntryID | unsignedInt | W | The EntryID for the next phase of the ring pattern, after the specified Duration of this phase has completed.<br>A value of zero indicates that the ring pattern is to terminate after the current phase is completed.<br>This parameter is required to be editable only if the parameter VoiceService.{i}.Capabilities.Ring-PatternEditable is equal to true. | 0 |
| .VoiceService.{i}.VoiceProfile.{i}.Line.-{i}.CallingFeatures. | object | - | Voice line parameters related to optional endpoint based calling features. | - |
| CallerIDEnable | boolean | W | Enable or disable the transmission of caller ID information on outgoing calls. | - |
| CallerIDNameEnable | boolean | W | Enable or disable the transmission of caller ID name information on outgoing calls. | - |
| CallerIDName | string(256) | W | String used to identify the caller. | - |

[13] The defaults given for this object apply only to explicit creation of an instance of this object and not to automatic creation of instances of this object due to creation of a parent object.

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| CallWaitingEnable | boolean | W | Enable or disable call waiting in the endpoint. This parameter should not be present if the CPE does not support endpoint managed call waiting. | - |
| CallWaitingStatus | string | - | Status of endpoint managed call waiting, if supported. Enumeration of:<br>"Disabled"<br>"Idle"<br>"SecondaryRinging"<br>"SecondaryConnecting"<br>"SecondaryConnected"<br>This parameter should not be present if the CPE does not support endpoint managed call waiting. | - |
| MaxSessions | unsignedInt | W | Indicates the maximum number of simultaneous sessions that may be conferenced together by the endpoint. This value SHOULD be less than the value of VoiceService.{i}.Capabilities.Max-SessionsPerLine. This parameter should not be present if the CPE does not support endpoint managed conference calling. | - |
| ConferenceCallingStatus | string | - | Status of endpoint managed conference calling, if supported. Enumeration of:<br>"Disabled"<br>"Idle"<br>"SecondaryCalling"<br>"SecondaryConnecting"<br>"SecondaryConnected"<br>"InConferenceCall"<br>This parameter should not be present if the CPE does not support endpoint managed conference calling. | - |
| ConferenceCallingSessionCount | unsignedInt | - | Number of active sessions on this line. This parameter has the following interpretation:<br>0 = No call in progress<br>1 = Single call in progress<br>>1 = Conference call in progress<br>This parameter should not be present if the CPE does not support endpoint managed conference calling. | - |
| CallForwardUnconditionalEnable | boolean | W | Enable or disable call forwarding by the endpoint. This parameter should not be present if the CPE does not support endpoint based call forwarding. | - |
| CallForwardUnconditionalNumber | string(32) | W | Directory number to which all incoming calls to this line should be forwarded if CallForward-UnconditionalEnable is true. This parameter should not be present if the CPE does not support endpoint based call forwarding | - |
| CallForwardOnBusyEnable | boolean | W | Enable or disable call forwarding-on-busy by the endpoint. This parameter should not be present if the CPE does not support endpoint based call forwarding. | - |
| CallForwardOnBusyNumber | string(32) | W | Directory number to which all incoming calls to this line should be forwarded if CallForwardOn-BusyEnable is true and the line is busy. This parameter should not be present if the CPE does not support endpoint based call forwarding | - |
| CallForwardOnNoAnswerEnable | boolean | W | Enable or disable call forwarding-on-no-answer by the endpoint. This parameter should not be | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | present if the CPE does not support endpoint based call forwarding. | |
| CallForwardOnNoAnswerNumber | string(32) | W | Directory number to which all incoming calls to this line should be forwarded if CallForwardOnNo-AnswerEnable is true and there is no local answer. This parameter should not be present if the CPE does not support endpoint based call forwarding | - |
| CallForwardOnNoAnswerRingCount | unsignedInt | W | Number of rings before considering there to be no answer for call forwarding-on-no-answer. This parameter should not be present if the CPE does not support endpoint based call forwarding | - |
| CallTransferEnable | boolean | W | Enable or disable call transfer by the endpoint. This parameter should not be present if the CPE does not support endpoint based call transfer. | - |
| MWIEnable | boolean | W | Enable or disable Message Waiting Indication by the endpoint. This parameter should not be present if the CPE does not support MWI. | - |
| MessageWaiting | boolean | - | Indicates whether or not a message is currently waiting on this line as known by the CPE. This parameter should not be present if the CPE does not support MWI. | - |
| AnonymousCallBlockEnable | boolean | W | Enable or disable Anonymous Call Block capability in the endpoint. This parameter should not be present if the CPE does not support endpoint based Anonymous Call Block capability. | - |
| AnonymousCalEnable | boolean | W | Enable or disable Anonymous Call capability in the endpoint. This parameter should not be present if the CPE does not support endpoint based Anonymous Call capability. | - |
| DoNotDisturbEnable | boolean | W | Enable or disable Do Not Disturb capability in the endpoint. This parameter should not be present if the CPE does not support endpoint based Do Not Disturb capability. | - |
| CallReturnEnable | boolean | W | Enable or disable Call Return capability in the endpoint. This parameter should not be present if the CPE does not support endpoint based Call Return capability. | - |
| RepeatDialEnable | boolean | W | Enable or disable Repeat Dial capability in the endpoint. This parameter should not be present if the CPE does not support endpoint based Repeat Dial capability. | - |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Voice-Processing. | object | - | Voice line parameters related to voice processing capabilities. | - |
| TransmitGain | int | W | Gain in units of 0.1 dB to apply to the transmitted voice signal prior to encoding. This gain is a modifier of the default transmit-gain, which is unspecified. | - |
| ReceiveGain | int | W | Gain in units of 0.1 dB to apply to the received voice signal after decoding. This gain is a modifier of the default receive-gain, which is unspecified. | - |
| EchoCancellationEnable | boolean | W | Enable or disable echo cancellation for this line. | - |
| EchoCancellationInUse | boolean | - | Indication of whether or not echo cancellation is currently in use for this line. | - |
| EchoCancellationTail | unsignedInt | - | Tail length in milliseconds of the echo canceller associated with this line (whether or not it is currently in use). | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| .VoiceService.{i}.VoiceProfile.{i}.Line.-{i}.Codec. | object | - | This object indicates the state of the transmit and receive codec for this voice line instance. | - |
| TransmitCodec | string(64) | - | The codec currently in use for the outgoing voice stream. Enumeration from the list of available codecs as given in the VoiceService.{i}.-Capabilities.Codecs table. | - |
| ReceiveCodec | string(64) | - | The codec currently in use for the incoming voice stream. Enumeration from the list of available codecs as given in the VoiceService.{i}.-Capabilities.Codecs table. | - |
| TransmitBitRate | unsignedInt | - | Codec bit rate in bits per second for the codec currently in use for the outgoing voice stream. | - |
| ReceiveBitRate | unsignedInt | - | Codec bit rate in bits per second for the codec currently in use for the incoming voice stream. | - |
| TransmitSilenceSuppression | boolean | - | Whether or not silence suppression is in use for the outgoing voice stream. | - |
| ReceiveSilenceSuppression | boolean | - | Whether or not silence suppression is in use for the incoming voice stream. | - |
| TransmitPacketizationPeriod | unsignedInt | - | Current outgoing packetization period in milliseconds. | - |
| .VoiceService.{i}.VoiceProfile.{i}.Line.-{i}.Codec.List.{i}. | object | - | Table to describe the set of codecs enabled for use with this line. Each entry in this table refers to a distinct combination of codec and bit rate. When a Line is created, this object MUST be populated with the set of supported codecs matching the VoiceService.{i}.Capabilities.Codecs table. The ACS MAY restrict and/or prioritize the codec support for this profile using this object.<br>Applicable only for a VoIP endpoint. | - |
| EntryID | unsignedInt [1:] | - | Unique identifier for each entry in this table. The value MUST match that of the corresponding entry in the VoiceService.{i}.Capabilities.Codecs table. | - |
| Codec | string(64) | - | Identifier of the codec type. The value MUST match that of the corresponding entry in the VoiceService.{i}.Capabilities.Codecs table. | - |
| BitRate | unsignedInt | - | Bit rate, in bits per second. The value MUST match that of the corresponding entry in the VoiceService.{i}.Capabilities.Codecs table. | - |
| PacketizationPeriod | string(64) | W | Comma-separate list of supported packetization periods, in milliseconds, or continuous ranges of packetization periods as defined in VoiceService.-{i}.Capabilities.Codecs.PacketizationPeriod.<br>The set of packetization periods may be restricted by modifying the value of this parameter to a more restricted set of values than is listed in VoiceService.{i}.Capabilities.Codecs.-PacketizationPeriod. The CPE MUST ignore any values or portions of ranges outside of those specified in VoiceService.{i}.Capabilities.Codecs.-Packetization.Period. | - |
| SilenceSuppression | boolean | W | Indicates support for silence suppression for this codec. If silence suppression is supported, it can be disabled for this codec/bit-rate by setting this parameter to false. | - |
| Enable | boolean | W | Enable or disable the use of this combination of codec parameters. | True |
| Priority | unsignedInt [1:] | W | Indicates the priority for this combination of codec parameters, where 1 is the highest priority. Where the priority differs between entries in this table, the CPE SHOULD use the highest priority | 1 |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| | | | (lowest numbered) entry among those supported by the remote endpoint and consistent with the available bandwidth. Where the priorities are equal among multiple entries, the CPE MAY apply a local criterion for choosing among them. | |
| .VoiceService.{i}.VoiceProfile.{i}.Line.-{i}.Session.{i}. | object | - | Information on each active session associated with this voice line instance. | - |
| SessionStartTime | dateTime | - | The time that the session started, in UTC. | - |
| SessionDuration | unsignedint | - | Duration time of the current session, in seconds. | - |
| FarEndIPAddress | string | - | The IP address of far end VoIP device. | - |
| FarEndUDPPort | unsignedInt [0:65535] | - | The UDP port used for current RTP session in the far end device. | - |
| LocalUDPPort | unsignedInt [0:65535] | - | The local UDP port used for current RTP session. | - |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Stats. | object | - | Statistics for this voice line instance. | - |
| ResetStatistics | boolean | W | When set to one, resets the statistics for this voice line. Always False when read. | - |
| PacketsSent | unsignedInt | - | Total number of RTP packets sent for this line. | - |
| PacketsReceived | unsignedInt | - | Total number of RTP packets received for this line. | - |
| BytesSent | unsignedInt | - | Total number of RTP payload bytes sent for this line. | - |
| BytesReceived | unsignedInt | - | Total number of RTP payload bytes received for this line. | - |
| PacketsLost | unsignedInt | - | Total number of RTP packets that have been lost for this line. | - |
| Overruns | unsignedInt | - | Total number of times the receive jitter buffer has overrun for this line. | - |
| Underruns | unsignedInt | - | Total number of times the receive jitter buffer has underrun for this line. | - |
| IncomingCallsReceived | unsignedInt | - | Total incoming calls received. | - |
| IncomingCallsAnswered | unsignedInt | - | Total incoming calls answered by the local user. | - |
| IncomingCallsConnected | unsignedInt | - | Total incoming calls that successfully completed call setup signaling. | - |
| IncomingCallsFailed | unsignedInt | - | Total incoming calls that failed to successfully complete call setup signaling. | - |
| OutgoingCallsAttempted | unsignedInt | - | Total outgoing calls attempted. | - |
| OutgoingCallsAnswered | unsignedInt | - | Total outgoing calls answered by the called party. | - |
| OutgoingCallsConnected | unsignedInt | - | Total outgoing calls that successfully completed call setup signaling. | - |
| OutgoingCallsFailed | unsignedInt | - | Total outgoing calls that failed to successfully complete call setup signaling. | - |
| CallsDropped | unsignedInt | - | Total calls that were successfully connected (incoming or outgoing), but dropped unexpectedly while in progress without explicit user termination. | - |
| TotalCallTime | unsignedInt | - | Cumulative call duration in seconds. | - |
| ServerDownTime | unsignedInt | - | The number of seconds the CPE is unable to maintain a connection to the server. SHOULD not include time in which overall network connectivity is unavailable. Applies only to SIP. | - |
| ReceivePacketLossRate | unsignedInt [0:100] | - | Current receive packet loss rate in percent, calculated as defined in section 6.4 of [6] | - |
| FarEndPacketLossRate | unsignedInt [0:100] | - | Current far end receive packet lost rate in percent, calculated as defined in section 6.4 of [6]. | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| ReceiveInterarrivalJitter | unsignedInt | - | Current receive interarrival jitter in microseconds. Calculated from J(i) as defined in section 6.4 of [6], with units converted to microseconds. | - |
| FarEndInterarrivalJitter | unsignedInt | - | Current Interarrival jitter in microseconds as reported from the far-end device via RTCP. Calculated from J(i) as defined in section 6.4 of [6], with units converted to microseconds. | - |
| RoundTripDelay | unsignedInt | - | Current round trip delay in microseconds calculated as defined in section 6.4 of [6]. | - |
| AverageReceiveInterarrivalJitter | unsignedInt | - | Average receive interarrival jitter in microseconds since the beginning of the current call. Calculated as the average of D(i,j) as defined in section 6.4 of [6], with units converted to microseconds. | - |
| AverageFarEndInterarrivalJitter | unsignedInt | - | Average far-end interarrival jitter in microseconds since the beginning of the current call. Calculated as the average of the interarrival jitter values reported by the far-end, with units converted to microseconds. | - |
| AverageRoundTripDelay | unsignedInt | - | Average round trip delay in microseconds since the beginning of the current call. Average of the RoundTripDelay statistic accumulated each time the delay is calculated. | - |
| .VoiceService.{i}.PhyInterface.{i}. | object | - | Each instance is associated with a distinct physical FXS (Foreign eXchange Station) port. Instances of this object are statically created by the CPE.<br>Applicable only for a VoIP Endpoint. | - |
| PhyPort | string(2) | - | The physical port number on the device. | - |
| InterfaceID | unsignedInt [1:] | - | The unique identifier of the physical port. This value MAY be used in the PhyReferenceList parameter in the Line object to indicate which physical ports are associated with a line. | - |
| Description | string(32) | - | A description of the physical port. | - |
| .VoiceService.{i}.PhyInterface.{i}.Tests. | object | - | Voice port tests.<br>If this object is supported, the capability VoiceService.{i}.Capabilities.VoicePortTests MUST be equal to true. | - |
| TestState | string | W | Indicates the current test state. Enumeration of:<br>“None”<br>“Requested”<br>“Complete”<br>“Error_TestNotSupported”<br>Value MAY be set to Requested to initiate a diagnostic test. When writing, the only allowed value is Requested. To ensure the use of the proper test parameters (the writable parameters in this object), the test parameters MUST be set either prior to or at the same time as (in the same SetParameterValues) setting the TestState to Requested.<br>When requested, the CPE SHOULD wait until after completion of the communication session with the ACS before starting the test.<br>When the test initiated by the ACS is completed (successfully or not), the CPE MUST establish a new connection to the ACS to allow the ACS to view the results, indicating the Event code "8 DIAGNOSTICS COMPLETE" in the Inform message. | - |

| Name[1] | Type | Write[2] | Description | Default[3] |
|---|---|---|---|---|
| TestSelector | string(64) | W | Indicates which test to perform. Enumeration of:<br>"PhoneConnectivityTest"<br>The phone connectivity test indicates that the CPE should determine if one or more phones associated with this physical port are properly connected. This test is appropriate only for CPE that connect to phones of any type.<br>The parameter MAY instead indicate a vendor-specific test, which MUST use the format defined in [3]. For example:<br>"X_EXAMPLE-COM_MyTest" | - |
| PhoneConnectivity | boolean | - | Indicates whether or not at least one phone associated with this physical port is properly connected. This parameter is applicable only if the PhoneConnectivityTest is supported. | - |

## 3.1 Notification Requirements

CPE MUST support Active Notification (see [2]) for all parameters defined in the VoiceService data model with the exception of those parameters listed in Table 2. For only those parameters listed Table 2, the CPE MAY reject a request by an ACS to enable Active Notification via the SetParameterAttributes RPC by responding with fault code 9009 as defined in [2] (Notification request rejected).

CPE MUST support Passive Notification (see [2]) for all parameters defined in the VoiceService data model, with no exceptions.

**Table 2 – Parameters for which Active Notification MAY be denied by the CPE**

| Parameter[14] |
|---|
| .VoiceService.{i}.Capabilities. |
| MaxProfileCount |
| MaxLineCount |
| MaxSessionsPerLine |
| MaxSessionCount |
| SignalingProtocols |
| Regions |
| RTCP |
| SRTP |
| SRTPKeyingMethods |
| SRTPEncryptionKeySizes |
| RTPRedundancy |
| DSCPCoupled |
| EthernetTaggingCoupled |
| PSTNSoftSwitchOver |
| FaxT38 |
| FaxPassThrough |
| ModemPassThrough |
| ToneGeneration |
| ToneDescriptionsEditable |
| PatternBasedToneGeneration |

[14] The name of a Parameter referenced in this table is the concatenation of the root object name as defined in [3], the object name shown in the yellow header, and the individual Parameter name.

| **Parameter**[14] |
|---|
| FileBasedToneGeneration |
| ToneFileFormats |
| RingGeneration |
| RingDescriptionsEditable |
| PatternBasedRingGeneration |
| RingPatternEditable |
| FileBasedRingGeneration |
| RingFileFormats |
| DigitMap |
| NumberingPlan |
| ButtonMap |
| VoicePortTests |
| .VoiceService.{i}.Capabilities.SIP. |
| Role |
| Extensions |
| Transports |
| URISchemes |
| EventSubscription |
| ResponseMap |
| TLSAuthenticationProtocols |
| TLSAuthenticationKeySizes |
| TLSEncryptionProtocols |
| TLSEncryptionKeySizes |
| TLSKeyExchangeProtocols |
| .VoiceService.{i}.Capabilities.MGCP. |
| Extensions |
| .VoiceService.{i}.Capabilities.H323. |
| FastStart |
| H235AuthenticationMethods |
| .VoiceService.{i}.Capabilities.Codecs.{i}. |
| EntryID |
| Codec |
| BitRate |
| PacketizationPeriod |
| SilenceSuppression |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}. |
| CallState |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Session.{i}. |
| SessionStartTime |
| SessionDuration |
| FarEndIPAddress |
| FarEndUDPPort |
| LocalUDPPort |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Stats. |
| PacketsSent |
| PacketsReceived |

| **Parameter**[14] |
|---|
| BytesSent |
| BytesReceived |
| PacketsLost |
| Overruns |
| Underruns |
| IncomingCallsReceived |
| IncomingCallsAnswered |
| IncomingCallsConnected |
| IncomingCallsFailed |
| OutgoingCallsAttempted |
| OutgoingCallsAnswered |
| OutgoingCallsConnected |
| OutgoingCallsFailed |
| CallsDropped |
| TotalCallTime |
| ServerDownTime |
| ReceivePacketLossRate |
| FarEndPacketLossRate |
| ReceiveInterarrivalJitter |
| FarEndInterarrivalJitter |
| RoundTripDelay |
| AverageReceiveInterarrivalJitter |
| AverageFarEndInterarrivalJitter |
| AverageRoundTripDelay |

# 4 Profile Definitions

This section specifies the profiles defined for version 1.0 of the Voice Service data model. The use of profiles for this data model follows the definition and usage conventions described in [3].

## 4.1 Notation

The following abbreviations are used to specify profile requirements:

| Abbreviation | Description |
|---|---|
| R | Read support is REQUIRED. |
| W | Both Read and Write support is REQUIRED. |
| P | The object is REQUIRED to be present. |
| C | Creation and deletion of the object via AddObject and DeleteObject is REQUIRED. |

## 4.2 Endpoint Profile

Table 3 defines the Endpoint:1 profile for the VoiceService:1 object. The minimum required version for this profile is VoiceService:1.0.

**Table 3 – Endpoint:1 profile definition for VoiceService:1**

| Name | Requirement |
|---|---|
| .VoiceService.{i}. | P |
| VoiceProfileNumberOfEntries | R |
| .VoiceService.{i}.Capabilities. | P |
| MaxProfileCount | R |
| MaxLineCount | R |
| MaxSessionsPerLine | R |
| MaxSessionCount | R |
| SignalingProtocols | R |
| Regions | R |
| RTCP | R |
| SRTP | R |
| RTPRedundancy | R |
| DSCPCoupled | R |
| EthernetTaggingCoupled | R |
| PSTNSoftSwitchOver | R |
| FaxT38 | R |
| FaxPassThrough | R |
| ModemPassThrough | R |
| ToneGeneration | R |
| RingGeneration | R |
| NumberingPlan | R |
| ButtonMap | R |
| VoicePortTests | R |
| .VoiceService.{i}.Capabilities.Codecs.{i}. | P |
| EntryID | R |
| Codec | R |
| BitRate | R |

| Name | Requirement |
|---|---|
| PacketizationPeriod | R |
| SilenceSuppression | R |
| .VoiceService.{i}.VoiceProfile.{i}. | PC[15] |
| Enable | W |
| Reset | W |
| NumberOfLines | R |
| Name | W |
| SignalingProtocol | W |
| MaxSessions | W |
| DTMFMethod | W |
| DTMFMethodG711 | W |
| .VoiceService.{i}.VoiceProfile.{i}.RTP. | P |
| LocalPortMin | W |
| LocalPortMax | W |
| DSCPMark | W |
| TelephoneEventPayloadType | W |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}. | PC[16] |
| Enable | W |
| Status | R |
| CallState | R |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Codec. | P |
| TransmitCodec | R |
| ReceiveCodec | R |
| TransmitBitRate | R |
| ReceiveBitRate | R |
| TransmitSilenceSuppression | R |
| ReceiveSilenceSuppression | R |
| TransmitPacketizationPeriod | R |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Codec.List.{i}. | P |
| EntryID | R |
| Codec | R |
| BitRate | R |
| PacketizationPeriod | W |
| SilenceSuppression | W |
| Enable | W[17] |
| Priority | W[17] |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Session.{i}. | P |
| SessionStartTime | R |
| SessionDuration | R |
| FarEndIPAddress | R |
| FarEndUDPPort | R |

[15] Support for creation and deletion of Profiles is REQUIRED only if more than one Profile is supported as indicated by VoiceService.{i}.Capabilities.MaxProfileCount.
[16] Support for creation and deletion of Lines is REQUIRED only if more than one Line is supported as indicated by VoiceService.{i}.Capabilities.MaxLineCount.
[17] This parameter is REQUIRED to be writable only if there is more than one entry in this table.

| Name | Requirement |
|---|---|
| LocalUDPPort | R |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Stats. | P |
| ResetStatistics | W |
| PacketsSent | R |
| PacketsReceived | R |
| BytesSent | R |
| BytesReceived | R |
| PacketsLost | R |
| Overruns | R |
| Underruns | R |
| IncomingCallsReceived | R |
| IncomingCallsAnswered | R |
| IncomingCallsConnected | R |
| IncomingCallsFailed | R |
| OutgoingCallsAttempted | R |
| OutgoingCallsAnswered | R |
| OutgoingCallsConnected | R |
| OutgoingCallsFailed | R |
| CallsDropped | R |
| TotalCallTime | R |

## 4.3 SIPEndpoint Profile

The SIPEndpoint:1 profile for the VoiceService:1 object is defined as the union of the Endpoint:1 profile and the additional requirements defined in Table 4. The minimum required version for this profile is VoiceService:1.0.

**Table 4 – SIPEndpoint:1 profile definition for VoiceService:1**

| Name | Requirement |
|---|---|
| .VoiceService.{i}.Capabilities.SIP. | P |
| Role | R |
| Extensions | R |
| Transports | R |
| URISchemes | R |
| EventSubscription | R |
| ResponseMap | R |
| .VoiceService.{i}.VoiceProfile.{i}.SIP. | P |
| ProxyServer | W |
| ProxyServerPort | W |
| ProxyServerTransport | W |
| RegistrarServer | W |
| RegistrarServerPort | W |
| RegistrarServerTransport | W |
| UserAgentDomain | W |
| UserAgentPort | W |
| UserAgentTransport | W |
| OutboundProxy | W |

| Name | Requirement |
|---|---|
| OutboundProxyPort | W |
| Organization | W |
| RegistrationPeriod | W |
| RegisterExpires | W |
| UseCodecPriorityInSDPResponse | W |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.SIP. | P |
| AuthUserName | W |
| AuthPassword | W |
| URI | W |

## 4.4 MGCPEndpoint Profile

The MGCPEndpoint:1 profile for the VoiceService:1 object is defined as the union of the Endpoint:1 profile and the additional requirements defined in Table 5. The minimum required version for this profile is VoiceService:1.0.

**Table 5 – MGCPEndpoint:1 profile definition for VoiceService:1**

| Name | Requirement |
|---|---|
| .VoiceService.{i}.Capabilities.MGCP. | P |
| Extensions | R |
| .VoiceService.{i}.VoiceProfile.{i}.MGCP. | P |
| CallAgent1 | W |
| CallAgentPort1 | W |
| CallAgent2 | W |
| CallAgentPort2 | W |
| RetranIntervalTimer | W |
| MaxRetranCount | W |
| RegisterMode | W |
| LocalPort | W |
| Domain | W |
| User | W |
| AllowPiggybackEvents | W |
| SendRSIPImmediately | W |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.MGCP. | P |
| LineName | W |

## 4.5 H323Endpoint Profile

The H323Endpoint:1 profile for the VoiceService:1 object is defined as the union of the Endpoint:1 profile and the additional requirements defined in Table 6. The minimum required version for this profile is VoiceService:1.0.

**Table 6 – H323Endpoint:1 profile definition for VoiceService:1**

| Name | Requirement |
|---|---|
| .VoiceService.{i}.Capabilities.H323. | P |
| FastStart | R |
| H235AuthenticationMethods | R |

| Name | Requirement |
|---|---|
| .VoiceService.{i}.VoiceProfile.{i}.H323. | P |
| Gatekeeper | W |
| GatekeeperPort | W |
| GatekeeperID | W |
| TimeToLive | W |
| H235Authentication | W |
| AuthPassword | W |
| SendersID | W |
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}.H323. | P |
| H323ID | W |

## 4.6 TAEndpoint Profile

The TAEndpoint:1 profile (for a terminal adapter endpoint) for the VoiceService:1 object is defined as the union of the Endpoint:1 profile and the additional requirements defined in Table 7. The minimum required version for this profile is VoiceService:1.0.

### Table 7 – TAEndpoint:1 profile definition for VoiceService:1

| Name | Requirement |
|---|---|
| .VoiceService.{i}.VoiceProfile.{i}.Line.{i}. | P |
| PhyReferenceList | W |
| .VoiceService.{i}.PhyInterface.{i}. | P |
| PhyPort | R |
| InterfaceID | R |

## Normative References

The following documents are referenced by this specification.

[1] RFC 2119, *Key words for use in RFCs to Indicate Requirement Levels*, http://www.ietf.org/rfc/rfc2119.txt

[2] TR-069, *CPE WAN Management Protocol*, DSL Forum Technical Report

[3] TR-106, *Data Model Template for TR-069-Enabled Devices*, DSL Forum Technical Report

[4] RFC 3261, *SIP: Session Initiation Protocol*, http://www.ietf.org/rfc/rfc3261.txt.

[5] RFC 3435, *Media Gateway Control Protocol (MGCP) Version 1.0*, http://www.ietf.org/rfc/rfc3435.txt.

[6] RFC 3550, *RTP: A Transport Protocol for Real-Time Applications*, http://www.ietf.org/rfc/rfc3550.txt.

# Appendix A. Facility Actions

A defined set of Facility Action values is given in Table 8. Facility Actions are referenced in the VoiceService data model in the objects VoiceService.{i}.VoiceProfile.{i}.NumberingPlan.PrefixInfo and VoiceService.{i}.VoiceProfile.{i}.ButtonMap.Button.

**Table 8 – Defined Facility Actions**

| Facility Action Name | Description |
|---|---|
| AA_REGISTER | Register an "Abbreviated Address" and corresponding destination. Abbreviated Addressing permits a user to define short numbers (within a particular range) to represent commonly used destinations. |
| AA_ERASE | Remove an "Abbreviated Address" and corresponding. |
| AA_INTERROGATE | Interrogate the status of an "Abbreviated Address". |
| CA_ACTIVATE | Make this call an anonymous call. |
| CCBS_ACTIVATE | Activation of Call Completion to Busy Subscriber. Call completion to Busy Subscriber is a facility that permits the user to request an automatic call back when a currently busy destination becomes available. |
| CCBS_DEACTIVATE | Deactivation of Call Completion to Busy Subscriber. |
| CCBS_INTERROGATE | Interrogation of the Call Completion to Busy Subscriber status and destination. |
| CCNR_ACTIVATE | Activation of Call Completion on No Reply. Call completion on no reply is a facility that permits the user to request an automatic call back when activity (eg a phone call is made) is detected from a destination from which there is currently no reply. |
| CCNR_DEACTIVATE | Deactivation of Call Completion on No Reply. |
| CCNR_INTERROGATE | Interrogation of the Call Completion on No Reply status and destination. |
| CFB_REGISTER | Register the destination for Call Forwarding on Busy. If Call Forwarding on Busy is activated, the call will be forwarded to the specified destination of the VoIP device is "busy". |
| CFB_ACTIVATE | Activate Call Forwarding on Busy to the registered destination. |
| CFB_DEACTIVATE | Deactivate Call Forwarding on Busy to the registered destination. |
| CFB_ERASE | Erase the registered Call Forwarding on Busy Destination. |
| CFB_INTERROGATE | Interrogate the registered Call Forwarding on Busy Destination. |
| CFNR_REGISTER | Register the destination for Call Forwarding on No Reply. If Call Forwarding on No Reply is activated, the call will be forwarded to the specified destination if there is no "answer". |
| CFNR_ACTIVATE | Activate Call Forwarding on No Reply to the registered destination. |
| CFNR_DEACTIVATE | Deactivate Call Forwarding on No Reply to the registered destination. |
| CFNR_ERASE | Erase the registered Call Forwarding on No Reply Destination. |
| CFNR_INTERROGATE | Interrogate the registered Call Forwarding on No Reply Destination. |
| CFNR_TIMER | Set the Call Forwarding on No Reply timer. |
| CFT_ACTIVATE | Activate Call Forwarding Timed. Call Forwarding Timed is a facility that permits the user to forward calls to particular destinations depending on the time of day. |
| CFT_DEACTIVATE | Deactivate Call Forwarding Timed. |
| CFT_INTERROGATE | Interrogate the status of Call Forwarding Timed. |

| Facility Action Name | Description |
|---|---|
| CFU_ACTIVATE | Activate Call Forwarding Unconditional to the registered destination. |
| CFU_DEACTIVATE | Deactivate Call Forwarding Unconditional to the registered destination. |
| CFU_REGISTER | Register the destination for Call Forwarding Unconditional. . If Call Forwarding Unconditional is activated, the call will always be unconditionally forwarded to the specified destination. |
| CFU_ERASE | Erase the registered Call Forwarding Unconditional Destination. |
| CFU_INTERROGATE | Interrogate the registered Call Forwarding Unconditional Destination. |
| CLIR_ACTIVATE | Activate Calling Line Identification Restriction (i.e. your calling number will be restricted). |
| CLIR_DEACTIVATE | Deactivate Calling Line Identification Restriction. |
| CLIR_INTERROGATE | Interrogate the Calling Line Identification status. |
| CW_ACTIVATE | Activate Call Waiting. |
| CW_DEACTIVATE | Deactivate Call Waiting. |
| CW_INVOKE | Used for answering a waiting call or switching between calls. |
| OCB_ACTIVATE | Activate Outgoing Call Barring. Outgoing call barring is a facility that permits the user to bar calls to certain destinations—usually depending on the cost of a call. |
| OCB_DEACTIVATE | Deactivate Outgoing Call Barring. |
| OCB_INTERROGATE | Interrogate the Outgoing Call Barring status. |
| PSO_ACTIVATE | Switch the call to a PSTN FXO line. Applies only if the value of the capability VoiceService.{i}.Capabilities.PSTNSoftSwitchOver is true. |
| PW_SET | Set the password used for controlling access to the Facility Actions. |
| SCF_ACTIVATE | Activate Selective Call Forwarding. Selective Call Forwarding is a facility that permits the user to forward calls to different destinations depending on the calling number. |
| SCF_DEACTIVATE | Deactivate Selective Call Forwarding. |
| SCF_INTERROGATE | Interrogate the Selective Call Forwarding status. |
| SCREJ_ACTIVATE | Activate Selective Call Rejection. Selective Call Rejection is a facility that permits the user to selectively reject calls depending on the calling number. |
| SCREJ_DEACTIVATE | Deactivate Selective Call Rejection. |
| SCREJ_INTERROGATE | Interrogate the Selective Call Rejection status. |
| SR_ACTIVATE | Activate Selective Ringing. Selective Ringing is a facility that permits the user to specify the generation of different ring tones depending on the calling number. |
| SR_DEACTIVATE | Deactivate Selective Ringing. |
| SR_INTERROGATE | Interrogate the Selective Ringing status. |
| TU_ACTIVATE | Send the dialed number as a TEL URI. Applies only for calls made using the SIP signaling protocol, and only if "tel" is listed among the supported URI formats in VoiceService.{i}.Capabilities.SIP.URISchemes.<br>For this Facility Action, the FacilityActionArgument has the following definition:<br>If empty, the TEL URI is to be represented as a global number.<br>If non-empty, the TEL URI is to be represented as a local number, and the value of the FacilityActionArgument is to be interpreted as the phone-context. |

# Appendix B. Downloading Tone and Ringer Files

An ACS MAY make use of the TR-069 Download method to download Tone and/or Ringer files for use in tone generation and ringer definitions. File-based tone generation support in a CPE is indicated by the capability VoiceService.{i}.Capabilities.FileBasedToneGeneration. File-based ring generation support in a CPE is indicated by the capability VoiceService.{i}.Capabilities.FileBasedRingGeneration.

For using the TR-069 Download method to download tone or ringer files to the CPE, Table 9 indicates the FileType argument that MUST be used for the Download.

**Table 9 – Download FileType Arguments**

| File Type | Download FileType Argument |
|---|---|
| Tone file | "4 Tone File" |
| Ringer file | "5 Ringer File" |

Files MAY either be downloaded individually, or one or more files of the same type (tone or ringer) MAY be bundled in the TR-069 Signed Package Format.

The actual file format of the individual tone or ringer files MUST match one of the available file formats listed in the capabilities VoiceService.{i}.Capabilities.ToneFileFormats for tone files, and VoiceService.-{i}.Capabilities.RingFileFormats for ring files.