Reference Guide

TR-069 & TR-104

CPE WAN Management Protocol (CWMP)

Version 6.6

HD VoIP Sounds Better

AudioCodes

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Notice

This document explains how to configure the AudioCodes gateway using TR-069 and TR-104 protocols.

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Documentation Feedback

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1 Introduction

This document describes the CPE WAN Management Protocol, intended for communication between a CPE and Auto-Configuration Server (ACS). The CPE WAN Management Protocol defines a mechanism that encompasses secure auto-configuration of a CPE, and also incorporates other CPE management functions into a common framework.

The CPE WAN Management Protocol is intended to support a variety of functionalities to manage a collection of CPE, including the following primary capabilities:

- Auto-configuration and dynamic service provisioning
- Software/firmware image management
- Software module management
- Status and performance monitoring
- Diagnostics

Note: CWMP is supported by the Multi-Service Business Router (MSBR) product line as well as the MediaPack MP-1xx product line.

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2 Overview

The following describes the TR-069 and TR-104 models.

2.1 TR-069 Data Model

TR-069 is a specification published by Broadband Forum (www.broadband-forum.org) entitled CPE WAN management protocol (CWMP). It defines an application layer protocol for remote management of end-user devices.

TR-069 uses a bi-directional SOAP/HTTP protocol for communication between the customer premises equipment (CPE) and the Auto Configuration Servers (ACS). For MSBR devices, the TR-069 connection to the ACS can be done on the LAN or WAN interface.

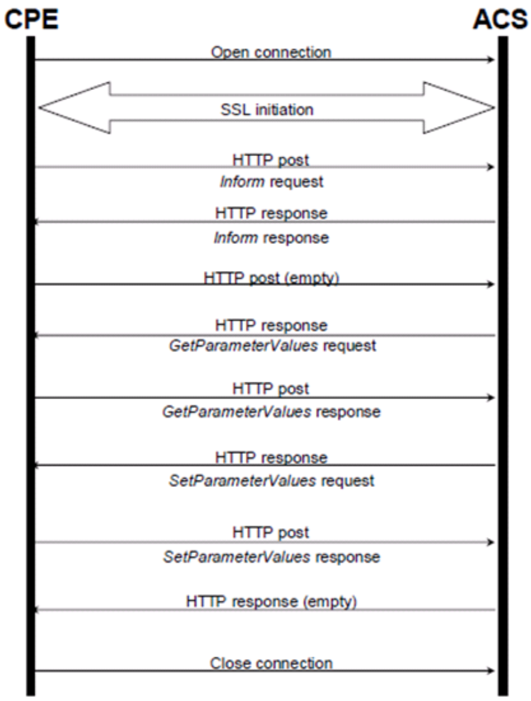
The protocol stack looks as follows:

Figure 2-1: TR-069 Protocol Stack

CPE/ACS Management Application
RPC Methods
SOAP
HTTP
SSL/TLS
TCP/IP

Communication is typically established by the CPE; hence, messages from CPE to ACS are typically carried in HTTP requests, and messages from ACS to CPE in HTTP responses.

Figure 2-2: TR-069 Session Example

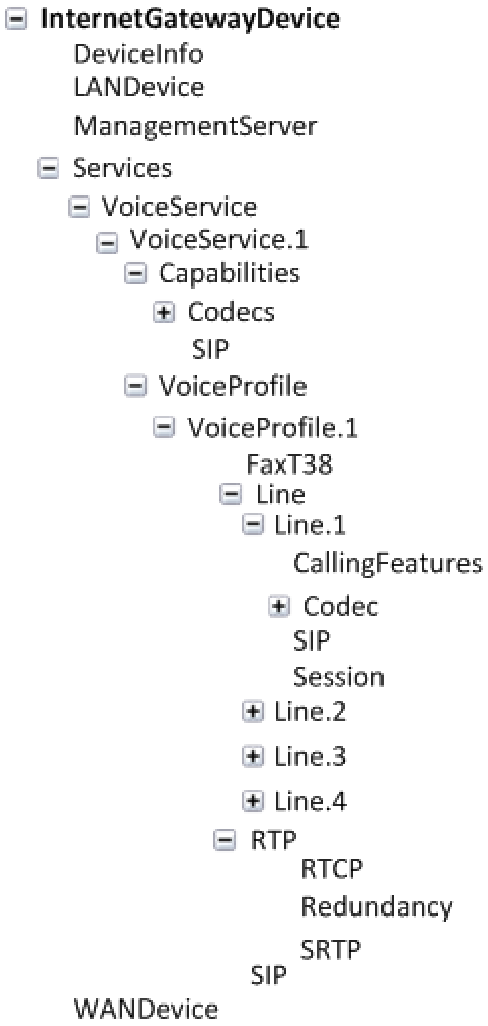


Communication between ACS and CPE is defined via Remote Procedure Call (RPC) methods. TR-069 defines a generic mechanism by which an ACS can read or write parameters to configure a CPE and monitor CPE status and statistics. It also defines the mechanism for file transfer and firmware/software management. However, it does not define individual parameters; these are defined in separate documents, as described below.

Some of the RPC methods are Configuration File Download, Firmware upgrade, Get Parameter Value, Set Parameter Value, Reboot, and the upload and download files.

TR-106 defines the “data model” template for TR-069 enabled devices. The Data Model consists of objects and parameters hierarchically organized in a tree with a single Root Object, typically named *Device*. Arrays of objects are supported by appending a numeric index to the object name (e.g. ABCService.1 in the example below); such objects are called “multi-instance objects”.

Figure 2-3: TR-069 Model Data Example



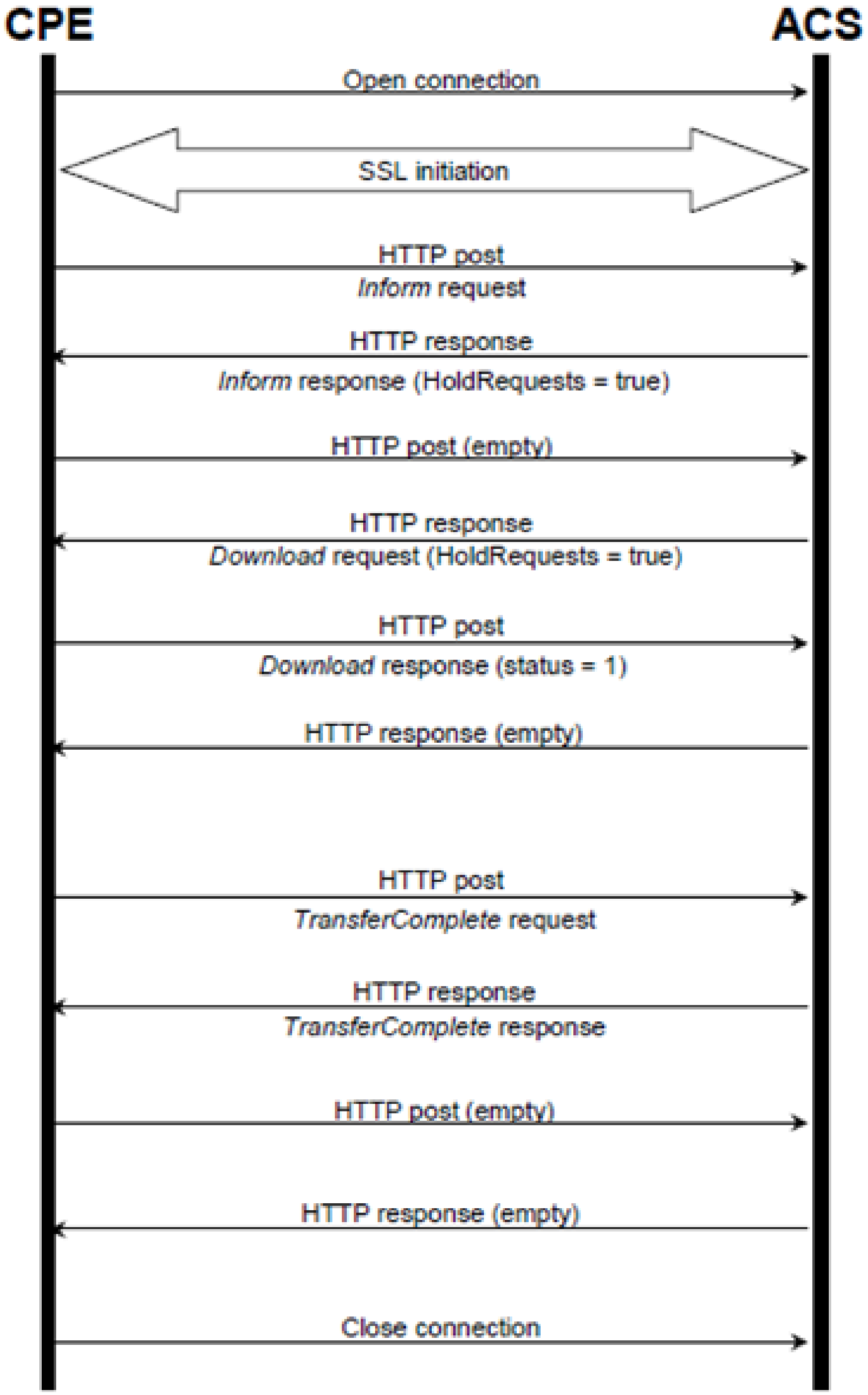
The following is a list of some of the TR-069 methods:

- **CPE Methods:**
  - GetRPCMethods: Used by the CPE or ACS to discover the set of methods supported by the Server or CPE it is in communication with.
  - SetParameterValues: Used by the ACS to modify the value of CPE parameter(s).
  - GetParameterValues: Used by the ACS to obtain the value of CPE parameter(s).
  - GetParameterNames: Used by the ACS to discover the parameters accessible on a particular CPE.
  - SetParameterAttributes: Used by the ACS to modify attributes associated with CPE parameter(s).
  - GetParameterAttributes: Used by the ACS to read the attributes associated with CPE parameter(s).
  - AddObject: Used by the ACS to create a new instance of a multi-instance object—a collection of parameters and/or other objects for which multiple instances are defined.
  - DeleteObject: Removes a particular instance of an object.
  - Download: Used by the ACS to cause the CPE to download the following file(s) from a designated location:
    - Firmware Upgrade Image (File Type = 1) - cmp file.
    - Vendor Configuration File (File Type = 3) - output of `show running-config` CLI command, which includes Data (for MSBR devices only) and Voice configuration.

    The CPE responds to the Download method, indicating successful or unsuccessful completion via one of the following:
    - A DownloadResponse with the Status argument set to zero (indicating success), or a fault response to the Download request (indicating failure).
    - A TransferComplete message sent later in the same session as the Download request (indicating either success or failure). In this case, the Status argument in the corresponding DownloadResponse has a value of one.

- A TransferComplete message sent in a subsequent session (indicating success or failure). In this case, the Status argument in the corresponding DownloadResponse has a value of one.

Figure 2-4: Download Method Execution Example



- Upload: Used by the ACS to cause the CPE to upload (to the ACS) the following files to a designated location:
  - Vendor Configuration File (File Type = 1 or 3): Output of `show running-config` CLI command, which includes Data (for MSBR devices only) and Voice configuration. For File Type 3 (where index is included – see below) only one instance of the file is supported.
  - Vendor Log File (File Type = 2 or 4): “Aggregated” log file. For File Type 2, the last file is supported. For File Type 4 (where index is included – see below), multiple files is supported.

  The CPE responds to the Upload method, indicating successful or unsuccessful completion via the UploadResponse or TransferComplete method.

  For a complete description of the Upload method, refer to TR-069 Amendment 3 section A.4.1.5.
- Reboot: Reboots the CPE. The CPE sends the method response and completes the remainder of the session prior to rebooting.
- X_0090F8_CommandResponse: Runs CLI commands.

■ **ACS Methods:**

- Inform: A CPE must call this method to initiate a transaction sequence whenever a connection to an ACS is established.
- TransferComplete: Informs the ACS of the completion (either successful or unsuccessful) of a file transfer initiated by an earlier Download or Upload method call.

2.2 TR-104 Data Model

TR-104 defines the data model for the provisioning of a voice-over-IP (VoIP) CPE device by an Auto-Configuration Server (ACS) using the mechanism defined in TR-069.

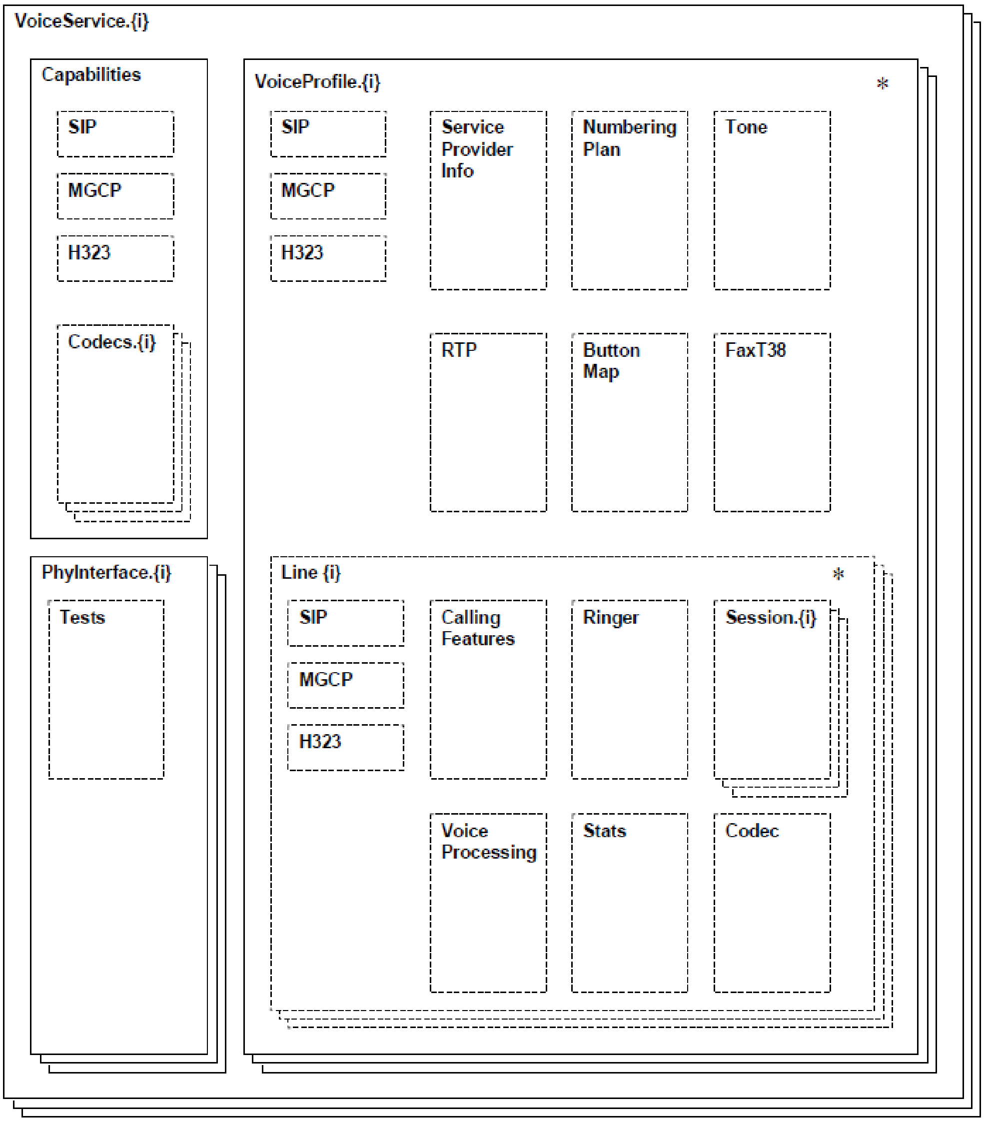
The following profiles are implemented by the device:

- Endpoint Profile
- SIP Endpoint Profile

2.2.1 Major Elements

The following diagram provides an overview of all objects defined in TR-104:

Figure 2-5: Objects Defined in TR-104



TR-104 describes functionality of a VoIP device that has one or more FXS lines.

> **Note:** TR-104 is only supported for FXS interfaces.

The device configuration model differs considerably from the model described in TR-104. For example, instead of the *VoiceProfile* object that corresponds to a group of phone lines, a separate *TelProfile* and/or IP Profile objects are used. TR-104 uses an implicit routing model while the device supports explicit routing tables that may be used to implement much more complicated call routings.

To “bridge” between these two very different configuration models, the following is assumed regarding the way the device is configured:

- Default Tel Profile (0) is used for all Hunt Groups (FXS Lines).
- “IP to Tel” Routing for phone lines (FXS ports) is implemented via the *PSTNPrefix* table. For each phone line (FXS port), a separate and unique entry in this table is created.
- “Tel to IP” routing is performed by the default *ProxySet (0)* and when the *IsProxySet* parameter is set to "1" (enabled). The first IP address entry in the Proxy Set configuration is used.
- Default IP Profile (0) is used for all routing rules.
- Outbound Proxy (if needed) is implemented via a single line in the *PSTNPrefix* table with ‘*’ wildcard for source and destination prefix.

2.2.2 VoiceService

VoiceService is a container "service" object as defined in TR-106.

The device implements a single instance of this object – VoiceService.1 only if it is equipped with the FXS ports.

2.2.2.1 VoiceService.{i}.Capabilities.Codecs

Codecs table describes the set of supported codecs. The table is read-only. Line.{i}.Codec.List table is used instead to customize list and parameters of coders assigned to the specific phone line.

The device will populate this table with *all supported* coders (as per CoderName_Type and CoderName_Rate ENUMs, taking into account the current DSP template). Each entry will contain "default" parameters (e.g. p-time) that correspond to the specific coder.

2.2.2.2 VoiceService.{i}.VoiceProfile

VoiceProfile corresponds to one or more phone lines (FXS ports) that share the same basic configuration.

The device implements a single instance of the VoiceProfile.1 object.

2.2.2.3 VoiceService.{i}.VoiceProfile.{i}.Line

The Line object corresponds to a single phone line (FXS port).

The device implements an instance of this object for each phone line (FXS port) configured in Hunt Group (TrunkGroup) table. Add/remove operations will be supported to allow configuration/removal of specific FXS port (See details in the table below).

2.2.2.4 VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Codec

The Codec object describes voice coder currently used by the specific phone line. In addition to that it provides a list of coders (Codec.List.{i}) enabled on the specific line and the ability to modify this configuration.

The device implements a global Coder table that is shared by all phone lines (FXS ports). It allows "per-line" customization of coders via Coder Group table. However, use of this functionality requires use of different Tel Profiles for different Hunt Groups (FXS port) and number of coder groups that may be configured is limited to 4.

Map Codec.List.{i} table directly to Coders (CoderName) table. This essentially means that all lines share the same configuration and configuration change for one line immediately affects all other lines.

2.2.2.5 VoiceService.{i}.VoiceProfile.{i}.Line.{i}.SIP

The Line.{i}.SIP object contains username/password assigned to a specific phone line (FXS port).

The device will map this object to corresponding entry in Authentication Table.

2.2.3 Endpoint Profile

The device complies with Version 1 of Endpoint profile (Endpoint:1) as defined in TR-104 Section 4.2.

Table 2-1: Endpoint Profile Table

Name	Type	Write	Description	Comment
InternetGatewayDevice. Services.VoiceService.{i}.	object	–	The top-level object.	A single instance of VoiceService object will be created.
VoiceProfileNumberOfEntries	unsignedInt	–	Defines the number of instances of VoiceProfile.	1
.VoiceService.{i}.Capabilities.	object	–	Defines the overall capabilities of the VoIP CPE.	
MaxProfileCount	unsignedInt	–	Defines the maximum total number of distinct voice profiles supported.	1
MaxLineCount	unsignedInt	–	Defines the maximum total number of lines supported across all profiles. This parameter is applicable only for a VoIP endpoint.	Total number of FXS interfaces (e.g. 4).
MaxSessionsPerLine	unsignedInt	–	Defines the 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. This parameter is applicable only for a VoIP endpoint.	1
MaxSessionCount	unsignedInt	–	Defines the 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 MaxSessionsPerLine.)	same value as MaxLineCount

Name	Type	Write	Description	Comment
SignalingProtocols	string(256)	–	Defines the comma-separated list of signaling protocols supported. Each item is an enumeration of: "SIP" "MGCP" "MGCP-NCS" "H.248" "H.323" Each entry MAY be appended with a version indicator in the form "/X.Y". For example: "SIP/2.0" The list MAY include vendor-specific protocols, which MUST be in the format defined in [3]. For example: "X_EXAMPLE-COM_MyProt"	"SIP"
Regions	string(256)	–	Defines the 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. An empty list indicates that the CPE does not support region-based customization via the Region parameter in the VoiceService.{i}.VoiceProfile.{i} object.	<empty>
RTCP	boolean	–	Defines support for RTCP. A true value indicates support for the object VoiceService.{i}.VoiceProfile.{i}.RTP.RTCP. This parameter is applicable only for a VoIP endpoint.	True
SRTP	boolean	–	Defines support for SRTP. A true value indicates support for the object VoiceService.{i}.VoiceProfile.{i}.RTP.SRTP. A true value also indicates that the SRTPKeyingMethods and SRTPEncryptionKeySizes parameters in this object are present. This parameter is applicable only for a VoIP endpoint.	True

Name	Type	Write	Description	Comment
SRTPKeyingMethods	string(256)	–	Defines comma-separated list of keying protocols supported by this endpoint for SRTP. Each item is an enumeration of: "Null" "Static" "SDP" "IKE" This list MAY include vendor-specific keying methods, which MUST use the format defined in [3]. This parameter is applicable only if the SRTP parameter in this object is equal to true. **Note:** This parameter is NOT part of *EndpointProfile*, but it's mandatory when SRTP is supported.	"Static"
SRTPEncryptionKeySizes	string(256)	–	Defines comma-separated list of unsigned integers, each represented a supported SRTP encryption key size. This parameter is applicable only if the SRTP parameter in this object is equal to true.	according to SRTPofferedSuites parameter possible values, the only supported value is "128"
RTPRedundancy	boolean	–	Defines support for RTP payload redundancy as defined in RFC 2198. A true value indicates support for VoiceService.{i}.VoiceProfile.{i}.RTP.Redundancy. This parameter is applicable only for a VoIP endpoint.	True
DSCPCoupled	boolean	–	Defines a true value that indicates that the CPE is constrained such that transmitted call control packets use the same DSCP marking as transmitted RTP packets. If the value is true, the CPE MUST NOT support the DSCPMark parameter for call control. This parameter is applicable only for a VoIP endpoint.	False

Name	Type	Write	Description	Comment
EthernetTaggingCoupled	boolean	–	Defines a true value that 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. 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). This parameter is applicable only for a VoIP endpoint.	False
PSTNSoftSwitchOver	boolean	–	Defines a true value that indicates the CPE is capable of supporting the PSO_Activate Facility Action, which allows a call to be switched to a PSTN FXO (Foreign eXchange Office) line. This parameter is applicable only for a VoIP endpoint.	False
FaxT38	boolean	–	Defines support for T.38 fax. A true value indicates support for the object VoiceService.{i}.VoiceProfile.{i}.FaxT38. This parameter is applicable only for a VoIP endpoint.	True
FaxPassThrough	boolean	–	Defines support for fax pass-through. A true value indicates support for the parameter VoiceService.{i}.VoiceProfile.{i}.FaxPassThrough. This parameter is applicable only for a VoIP endpoint.	True
ModemPassThrough	boolean	–	Defines support for modem pass-through. A true value indicates support for the parameter VoiceService.{i}.VoiceProfile.{i}.ModemPassThrough. This parameter is applicable only for a VoIP endpoint.	True

Name	Type	Write	Description	Comment
ToneGeneration	boolean	–	Defines support for tone generation. A true value indicates support for the object VoiceService.{i}.VoiceProfile.{i}.Tone. A true value also indicates that the ToneDescriptionsEditable, PatternBasedToneGeneration, and FileBasedToneGeneration parameters in this object are present. This parameter is applicable only for a VoIP endpoint.	False
RingGeneration	boolean	–	Defines support for ring generation. A true value indicates support for control of ring generation via the VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Ringer object. A true value also indicates that the RingDescriptionsEditable, PatternBasedRingGeneration, and FileBasedRingGeneration parameters in this object are present. This parameter is applicable only for a VoIP endpoint.	False
NumberingPlan	boolean	–	Defines support for a configurable numbering plan. A true value indicates support for a configurable numbering plan via the VoiceService.{i}.VoiceProfile.{i}.NumberingPlan object. This parameter is applicable only for a VoIP endpoint.	False
ButtonMap	boolean	–	Defines support for a configurable button map. A true value indicates support for a configurable button map via the VoiceService.{i}.VoiceProfile.{i}.ButtonMap object. This parameter is applicable only for a VoIP endpoint.	False
VoicePortTests	boolean	–	Defines support for remotely accessible voice-port tests. A true value indicates support for the VoiceService.{i}.PhyInterface.{i}.Tests object. This parameter is applicable only for a VoIP endpoint.	False

Name	Type	Write	Description	Comment
.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. Applicable only for a VoIP endpoint.	This table will be populated with entries for each supported coder (as per CoderName_Type enum and applicable bit-rates). If possible we should limit list of supported coders to those supported in current DSP template only. For each entry, default coder parameters (e.g. p-time) will be specified. Note: This table is NOT mapped to Coders (CoderName) table. The latter is mapped to Line.{i}.Codec.List table.
EntryID	unsignedInt [1:]	–	Defines a unique identifier for each entry in this table.	
Codec	string(64)	–	Defines an identifier of the type of codec. Enumeration of: "G.711MuLaw" "G.711ALaw" "G.726" "G.729" "G.729a" "G.729e" "G.728" "G.723.1" "G.722" "G.722.1" "G.722.2" "GSM-FR" "GSM-HR" "GSM-EFR" "iLBC" The parameter MAY instead be a vendor-specific codec, which MUST be in the format defined in [3]. For example: "X_EXAMPLE-COM_MyCodec"	
BitRate	unsignedInt	–	Defines a Bit Rate in bits per second. The value MUST be among the values appropriate for the specified codec.	

Name	Type	Write	Description	Comment
PacketizationPeriod	string(64)	–	Defines a comma-separated 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: “20” indicates a single discrete value. “10, 20, 30” indicates a set of discrete values. “5-40” indicates a continuous inclusive range. “5-10, 20, 30” indicates a continuous range in addition to a set of discrete values 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	C	Support for creation and deletion of Profiles is REQUIRED only if more than one Profile is supported as indicated by VoiceService.{i}.Capabilities.MaxProfileCount.	A single instance of VoiceProfile object will be created. Add/remove operations will NOT be supported.
Enable	string	W	Enables or disables all lines in this profile, or places it into a quiescent state. Enumeration of: “Disabled” “Quiescent” “Enabled” On creation, a profile MUST be in the Disabled state. 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.	Only ‘Enabled’ state will be supported.
NumberOfLines	unsignedInt	–	Defines the number of instances of Line within this VoiceProfile. Applicable only for a VoIP endpoint.	Number of entries in Hunt Group (TrunkGroups) table of type FXS.
Name	string(64)	W	Defines a human-readable string to identify the profile instance.	‘Default Profile’ Write operation will not be supported.

Name	Type	Write	Description	Comment
SignalingProtocol	string(64)	W	Defines the protocol to be used for this profile. A single protocol selected from among the available protocols indicated in VoiceService.{i}.Capabilities .SignalingProtocols.	“SIP”
MaxSessions	unsignedInt	W	Defines the 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.)	We will assume that a single session is supported per line – hence we will use the same value as for NumberOfLines.
DTMFMethod	string(64)	W	Defines the method by which DTMF digits MUST be passed. Enumeration of: “InBand” “RFC2833” “SIPInfo” If the parameter DTMFMethodG711 is non-empty, then this parameter applies only when the current codec is not G.711. The value “SIPInfo” is applicable only if the SignalingProtocol is SIP. This parameter is applicable only for a VoIP endpoint.	RxDTMFOption==0 && TxDTMFOption==1 → “SIPInfoNortel” RxDTMFOption==0 && TxDTMFOption==3 → “SIPInfo” RxDTMFOption==0 && TxDTMFOption==2 → “Notify” RxDTMFOption==3 && TxDTMFOption==4 → “RFC2833” RxDTMFOption==0 && TxDTMFOption==0 && DTMFTransportType==2 → “InBand” Any other combination → “Other” (read-only) **Note:** These are _global_ parameters. They may be overwritten by setting corresponding parameter in IPProfile. If we manage to “link” TelProfile and IPProfile together (see …) , we’ll use parameters from IPProfile; otherwise global parameters will be used.

Name	Type	Write	Description	Comment
DTMFMethodG711	string(64)	W	Defines the method by which DTMF digits MUST be passed if the current codec is G.711. Enumeration of: “InBand” “RFC2833” “SIPInfo” 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. The value “SIPInfo” is applicable only if the SignalingProtocol is SIP. This parameter is applicable only for a VoIP endpoint.	We support “empty” values only.
.VoiceService.{i}.VoiceProfile.{i}.RTP.	object	–	Voice profile parameters related to the voice stream sent via RTP. Applicable only for a VoIP endpoint.	
LocalPortMin	unsignedInt [0:65535]	W	Defines the base of port range to be used for incoming RTP streams for this profile.	BaseUDPport (or use Media Realm’s Port Range Start, if Media Realm is defined)
DSCPMark	unsignedInt [0:63]	W		
TelephoneEventPayloadType	unsignedInt [0:128]	W	Defines the payload type to be used for RTP telephone events. This parameter indicates the payload type to be used for DTMF events if RFC 2833 transmission of DTMF information is in use.	According to RFC 2833 PayloadType.
.VoiceService.{i}.VoiceProfile.{i}.Line.{i}.	object	C	Defines the support for creation and deletion of Lines is REQUIRED only if more than one Line is supported as indicated by VoiceService.{i}.Capabilities.MaxLineCount.	An instance of this object is created for each Hunt Group / TrunkGroup entry of type FXS. Add operation will be supported only if some “unused” FXS line exists. It will create a new entry in Hunt Group / TrunkGroup table. Remove operation will delete corresponding entry from Hunt Group table.

Name	Type	Write	Description	Comment
Enable	string	W	Enables or disables this line, or places it into a quiescent state. Enumeration of: "Disabled" "Quiescent" "Enabled" On creation, a line MUST be in the Disabled state. 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).	Quiescent state is not supported. Disabled state is implemented "at TR069 level only" (the entry will NOT exist in Hunt Group table).
Status	string	–	Indicates the status of this line. Enumeration of: "Up" "Initializing" "Registering" "Unregistering" "Error" "Testing" "Quiescent" "Disabled"	The following statuses are supported: - Up - Disabled
CallState	string	–	Indicates the call state for this line. Enumeration of: "Idle" "Calling" "Ringing" "Connecting" "InCall" "Hold" "Disconnecting"	Need to query SIP database to get this info.
.VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Codec.	object	–	This object indicates the state of the transmit and receive codec for this voice line instance.	**Note:** This object shows the data about currently established media session (i.e. the data should be taken from MediaEngine).
TransmitCodec	string(64)	–	Defines 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)	–	Defines 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	–	Defines the codec bit rate in bits per second for the codec currently in use for the outgoing voice stream.	

Name	Type	Write	Description	Comment
ReceiveBitRate	unsignedInt	–	Defines the codec bit rate in bits per second for the codec currently in use for the incoming voice stream.	
TransmitSilenceSuppression	boolean	–	Defines whether or not silence suppression is in use for the outgoing voice stream.	
ReceiveSilenceSuppression	boolean	–	Defines whether or not silence suppression is in use for the incoming voice stream.	
TransmitPacketizationPeriod	unsignedInt	–	Defines the 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. Applicable only for a VoIP endpoint.	The table will be populated with all supported coders (similar to Capabilities.Codecs table). Each “Enabled” object will be mapped to the entry in Coders (CoderName) table. Index in Coders table will be determined according to the object’s Priority parameter. “Disabled” objects will exist “at TR-069 level only”.
EntryID	unsignedInt [1:]	–	Defines the unique identifier for each entry in this table. The value MUST match that of the corresponding entry in the VoiceService.{i}.Capabilities .Codecs table.	Similar to Capabilities.Codecs
Codec	string(64)	–	Defines the Identifier of the codec type. The value MUST match that of the corresponding entry in the VoiceService.{i}.Capabilities .Codecs table.	Initial value - similar to Capabilities.Codecs; when “Enabled” – CoderName_Type
BitRate	unsignedInt	–	Defines the Bit rate in bits per second. The value MUST match that of the corresponding entry in the VoiceService.{i}.Capabilities .Codecs table.	Initial value - similar to Capabilities.Codecs; ; when “Enabled” – CoderName_Rate

Name	Type	Write	Description	Comment
PacketizationPeriod	string(64)	W	Defines the comma-separated list of supported packetization periods, in milliseconds, or continuous ranges of packetization periods as defined in VoiceService.{i}. Capabilities.Codecs.PacketizationPeriod. 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.	Initial value – similar to Capabilities.Codecs; when “Enabled” – CoderName_PacketInterval Can be modified by user.
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	Initial value – similar to Capabilities.Codecs; when “Enabled” – CoderName_SCE Can be modified by user.
Enable	boolean	W	This parameter is REQUIRED to be writable only if there is more than one entry in this table.	When set to ‘true’ corresponding entry in Coders (CoderName) table is created (with index that corresponds to Priority parameter). ***When set to ‘<u>false’</u> corresponding entry is removed from the Coders table***. **Note:** in DR add 2 options: Either not able option to put DISABLE to this param, or add new field of Admin-State to Coder Tables
Priority	unsignedInt [1:]	W	This parameter is REQUIRED to be writable only if there is more than one entry in this table.	Will be used to determine index of entry in Coders (CoderName) table. When value is changed and Enable==’true’ existing Coders entry will be removed and a new entry will be created instead (with a new index).
.VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Session.{i}.	object	–	Information on each active session associated with this voice line instance.	Information about currently active call – should be taken from SIP DB.
SessionStartTime	date time	–	Defines the time that the session started, in UTC.	
SessionDuration	unsignedint	–	Defines the duration time of the current session, in seconds.	
FarEndIPAddress	string	–	Defines the IP address of far end VoIP device.	

Name	Type	Write	Description	Comment
FarEndUDPPort	unsignedInt [0:65535]	–	Defines the UDP port used for current RTP session in the far end device.	
LocalUDPPort	unsignedInt [0:65535]	–	Defines the local UDP port used for current RTP session.	
.VoiceService.{i}.VoiceProfile.{i}.Line.{i}.Stats.	object	–	Statistics for this voice line instance.	Will be mapped to corresponding PMs (if such exist), SIP or VE counters. We may need to create new PMs to support all needed counters.
ResetStatistics	boolean	W	Defines when set to one, resets the statistics for this voiceline. Always False when read.	
PacketsSent	unsignedInt	–	Defines the total number of RTP packets sent for this line.	
PacketsReceived	unsignedInt	–	Defines the total number of RTP packets received for this line.	
BytesSent	unsignedInt	–	Defines the total number of RTP payload bytes sent for this line.	
BytesReceived	unsignedInt	–	Defines the total number of RTP payload bytes received for this line.	
PacketsLost	unsignedInt	–	Defines the total number of RTP packets that have been lost for this line.	
Overruns	unsignedInt	–	Defines the total number of times the receive jitter buffer has overrun for this line.	
Underruns	unsignedInt	–	Defines the total number of times the receive jitter buffer has underrun for this line.	
IncomingCallsReceived	unsignedInt	–	Defines the total incoming calls received.	
IncomingCallsAnswered	unsignedInt	–	Defines the total incoming calls answered by the local user.	
IncomingCallsConnected	unsignedInt	–	Defines the total incoming calls that successfully completed call setup signaling.	
IncomingCallsFailed	unsignedInt	–	Defines the total incoming calls that failed to successfully complete call setup signaling.	
OutgoingCallsAttempted	unsignedInt	–	Defines the total outgoing calls attempted.	
OutgoingCallsAnswered	unsignedInt	–	Defines the total outgoing calls answered by the called party.	

Name	Type	Write	Description	Comment
OutgoingCallsConnected	unsignedInt	–	Defines the total outgoing calls that successfully completed call setup signaling.	
OutgoingCallsFailed	unsignedInt	–	Defines the total outgoing calls that failed to successfully complete call setup signaling.	
CallsDropped	unsignedInt	–	Defines the total number calls that were successfully connected (incoming or outgoing), but dropped unexpectedly while in progress without explicit user termination.	
TotalCallTime	unsignedInt	–	Defines the cumulative call duration in seconds.	

2.2.3.1 Endpoint Profile Extensions

The following parts of TR-104 are not part of Endpoint profile (Endpoint:1) however are applicable to the device due to one of the following:

- are implied from the way we implement VoiceService.1.Capabilities object.
- were explicitly requested by potential customer

Table 2-2: Endpoint Profile Extensions Table

Name	Type	Write	Description	Comment
.VoiceService.{i}.VoiceProfile.{i}.	object	–		
FaxPassThrough	string	W	Specifies the behavior of the CPE for passthrough of fax data. Enumeration of: “Disable” “Auto” “Force” The value “Disable” prevents the CPE from switching to a fax pass-through mode. The value “Auto” allows the CPE to automatically detect fax data to determine whether or not to switch to a fax pass-through mode. The value “Force” forces the CPE to switch to a fax pass-through mode regardless of whether fax signaling is detected. If this parameter is supported, the capability VoiceService.{i}.Capabilities.FaxPassThrough MUST be equal to true. This parameter is appropriate only for a VoIP endpoint.	According to IsFaxUsed and FaxTransportMode parameters: **Read:** (IsFaxUsed==2) || ((IsFaxUsed==0) && (FaxTransportMode==2)) →”Auto” Otherwise → “Disabled” **Write:** “Auto” → IsFaxUsed=2 “Disable”→ IsFaxUsed=0, FaxTransportMode=0 (unless FaxT38.Enable == true) Note: the same values may be taken from “default” IPProfile 0.
ModemPassThrough	string	W	Specifies the behavior of the CPE for passthrough of modem data. Enumeration of: “Disable” “Auto” “Force” The value “Disable” prevents the CPE from switching to a modem pass-through mode.	According to V21ModemTransportType, V22ModemTransportType, V23ModemTransportType, V32ModemTransportType, V34ModemTransportType and BellModemTransportType parameters:

Name	Type	Write	Description	Comment
			The value “Auto” allows the CPE to automatically detect modem data to determine whether or not to switch to a modem pass-through mode. The value “Force” forces the CPE to switch to a modem pass-through mode regardless of whether modem signaling is detected. If this parameter is supported, the capability VoiceService.{i}.Capabilities.ModemPassThrough MUST be equal to true. This parameter is appropriate only for a VoIP endpoint.	**Read:** If all VxxModemTransportType == 2 → “Auto” otherwise → “False” **Write:** “Auto” → all VxxModemTransportType = 2 “Disable” → all VxxModemTransportType = 0
.VoiceService.{i}.VoiceProfile.{i}.RTP.RTCP.	object	–	Voice profile parameters related to RTCP. If this object is supported, the capability VoiceService.{i}.Capabilities.RTCP MUST be equal to true. Applicable only for a VoIP endpoint.	Neither of the below parameters are configurable via WEB/CLI/INI file.
Enable	boolean	W	Enables or disables RTCP.	Internal parameter: RtcpInterval==0
TxRepeatInterval	unsignedInt [1:]	W	Defines the Transmission repeat interval, in milliseconds	Internal parameter: RtcpInterval
VoiceService.{i}.VoiceProfile.{i}.RTP.SRTP.	object	–	Defines the Voice profile parameters for secure voice transmission via SRTP. If this object is supported, the capability VoiceService.{i}.Capabilities.SRTP MUST be equal to true. Applicable only for a VoIP endpoint.	
Enable	boolean	W	Enables or disables the use of SRTP. If RTCP is enabled, a true value of this parameter also implies the use of SRTCP.	EnableMediaSecurity
KeyingMethods	string(256)	W	Defines a 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.	“Static” Write operation will not be supported.
EncryptionKeySizes	string(256)	W	Defines a 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.	According to SRTPofferedSuites parameter. But since we support only “128” – this is the only possible value.
VoiceService.{i}.VoiceProfile.{i}.RTP.Redundancy	object	–	Defines Voice profile parameters for RTP payload redundancy as defined by RFC 2198. If this object is supported, the capability VoiceService.{i}.Capabilities.Redundancy MUST be equal to true. Applicable only for a VoIP endpoint.	
Enable	boolean	W	Enables or disables the use of RTP payload redundancy as defined by RFC 2198.	RTPRedundancyDepth
PayloadType	unsignedInt [0:127]	W	Defines the Payload Type of RTP packet using RFC 2198. Values should be within the range of dynamic Payload Types (96-127).	RFC2198PayloadType
FaxAndModemRedundancy	int[-1:5]	W	Specifies the redundancy number for fax and modem pass-through data transmissions. 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).	FaxRelayRedundancyDepth

Name	Type	Write	Description	Comment
			A value of -1 indicates RFC 2198 is not to be used for fax and modem pass-through data. If the optional parameter ModemRedudancy is present, then FaxAndModemRedudancy applies only to fax transmissions, but not to modem transmissions.	
VoiceService.{i}.VoiceProfile.{i}.FaxT38	object	–	Defines T.38 Fax information for devices that support T.38 relay. If this object is supported, the capability VoiceService.{i}.Capabilities.FaxT38 MUST be equal to true. Applicable only to a VoIP endpoint.	
Enable	boolean	W	Enables or disables the use of T.38.	According to IsFaxUsed and FaxTransportMode parameters: **Read**: ((IsFaxUsed==1) || (IsFaxUsed==3) || ((IsFaxUsed==0) && (FaxTransportMode==1)) → true otherwise → false **Write:** true → IsFaxUsed=1 false → IsFaxUsed=0, FaxTransportMode=0 (unless FaxPassThrough == "Auto") **Note:** The same values may be taken from "default" IPProfile 0.
BitRate	unsignedInt	W	Defines the maximum data rate for fax. Enumeration of the following values: 2400 4800 7200 9600 12000 14400 33600	mapped to FaxRelayMaxRate
HighSpeedPacketRate	unsignedInt	W	Defines the rate at which high speed data will be sent across the network, in milliseconds. Enumeration of the following values: 10 20 30 40	Mapped to FaxModemBypassBasicRTPPacketInterval
HighSpeedRedundancy	unsignedInt [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.	FaxRelayRedundancyDepth
LowSpeedRedundancy	unsignedInt [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.	FaxRelayEnhancedRedundancyDepth
.VoiceService.{i}.VoiceProfile.{i}.Line.{i}.	object	C	Defines support for creation and deletion of Lines is REQUIRED only if more than one Line is supported as indicated by VoiceService.{i}.Capabilities.MaxLineCount.	
DirectoryNumber	string(32)	W	Defines the directory number associated with this line. May be used to identify the line to the user. In case of H.323 signaling, this MUST be an E.164 number.	TrunkGroup_FirstPhoneNumber (from corresponding entry in Hun Group table)
.VoiceService.{i}.VoiceProfile.{i}.Line.{i}.CallingFeatures.	object	–	Defines Voice line parameters related to optional endpoint based calling features.	

Name	Type	Write	Description	Comment
CallerIDNameEnable	boolean	W	Enables or disables the transmission of caller ID information on outgoing calls.	This and the next parameters will be mapped to Caller Display Information table (CallerDisplayInfo) CallerDisplayInfo_IsCidRestricted
CallerIDName	string(256)	W	Defines a string used to identify the caller.	CallerDisplayInfo_DisplayString

2.2.4 SIP Endpoint Profile

The device complies with Version 1 of SIP Endpoint profile (SIPEndpoint:1) as defined in TR-104 section 4.3.

Table 2-3: SIP Endpoint Profile Table

Name	Type	Write	Description	Comment
.VoiceService.{i}.Capabilities.SIP.	object	–	Defines SIP-specific capabilities. Applicable only if SIP is among the list of supported protocols.	
Role	string	–	Defines the role of this VoIP CPE. Enumeration of: “UserAgent” “BackToBackUserAgents” “OutboundProxy” 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.	Always set to “UserAgent”
Extensions	string(256)	–	Defines a comma-separated list of SIP extension methods supported. SIP extension methods MUST be in the form of the method name in upper case. The list MAY include vendor-specific extensions, which MUST use the format defined in [3]. Examples: “REFER” “INFO” “X_EXAMPLE-COM_MyExt”	“REFER, INFO” SUBSCRIBE , etc..
Transports	string(256)	–	Defines a comma-separated list of SIP transport protocols supported. Each entry is an enumeration of: “UDP” “TCP” “TLS” “SCTP” The list MAY include vendor-specific transports, which MUST use the format defined in [3].	According to SIPTransportType UDP/TCP/TLS
URISchemes	string(256)	–	Defines a 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: “tel, fax”	“” (empty)
EventSubscription	boolean	–	Defines 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.	False
ResponseMap	boolean	–	Defines support for SIP response map. A true value indicates support for the VoiceService.{i}.VoiceProfile.{i}.SIP.ResponseMap object. This parameter is applicable only for a VoIP endpoint.	False
.VoiceService.{i}.VoiceProfile.{i}.SIP.	object	–	Defines Voice profile parameters that are specific to SIP user agents.	
ProxyServer	string(256)	W	Defines a Host name or IP address of the SIP proxy server. All SIP signaling traffic MUST be sent to the host indicated by this parameter and the port indicated by the ProxyServerPort parameter unless OutboundProxy	ProxyIp_IpAddress from the 1st entry in “default” Proxy Set (0); note that ProxyIp_IpAddress contains _both_ hostname and port –

Name	Type	Write	Description	Comment
			parameter is non-empty or a different route was discovered during normal operations SIP routing operation. 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 different proxy host was discovered dynamically during normal SIP routing operations.	hence some manipulation is needed. During “write” operation ensure that IsProxySet parameter is set to 1). During “read” operation, if IsProxySet parameter is set to 0, return empty string.
ProxyServerPort	unsignedInt [0:65535]	W	Defines the Destination Port to be used in connecting to the SIP server.	ProxyIp_IpAddress from the 1st entry in “default" Proxy Set (0); note that ProxyIp_IpAddress contains _both_ hostname and port – hence some manipulation is needed.
ProxyServerTransport	string	W	Defines the 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: “UDP” “TCP” “TLS” “SCTP”	According to ProxyIp_TransportType from the 1st entry in “default" Proxy Set (0
RegistrarServer	string(256)	W	Defines a Host name or IP address of the SIP registrar server. 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, ProxyServerPort, and ProxyServerTransport), ignoring all of the registrar server parameters (RegistrarServer, RegistrarServerPort and RegistrarServerTransport).	RegistrarIP **Note:** RegistrarIP may contain both hostbname and port – hence some manipulation is needed.
RegistrarServerPort	unsignedInt [0:65535]	W	Defines the Destination port to be used in connecting to the SIP registrar server. 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 (ProxyServer, ProxyServerPort, and ProxyServerTransport), ignoring all of the registrar server parameters (RegistrarServer, RegistrarServerPort and RegistrarServerTransport).	RegistrarIP **Note:** RegistrarIP may contain both hostbname and port – hence some manipulation is needed
RegistrarServerTransport	string	W	Defines the 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: “UDP” “TCP” “TLS” “SCTP” If the RegistrarServer parameter is empty,	According to RegistrarTransportType

Name	Type	Write	Description	Comment
			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, ProxyServerPort, and ProxyServerTransport), ignoring all of the registrar server parameters (RegistrarServer, RegistrarServerPort and RegistrarServerTransport).	
UserAgentDomain	string(256)	W	Defines a CPE domain string. If empty, the CPE SHOULD use its IP address as the domain.	ProxyName
UserAgentPort	unsignedInt [0:65535]	W	Defines a port used for incoming call control signaling.	LocalSIPPort or TCPLocalSIPPort or TLSLocalSIPPort – according to SIPTransportType
UserAgentTransport	string	W	Defines the Transport protocol to be used for incoming call control signaling. Must be chosen from among the transports supported, as indicated by VoiceService.{i}.Capabilities.SIP.Transports. Enumeration of: "UDP" "TCP" "TLS" "SCTP"	SIPTransportType
OutboundProxy	string(256)	W	Defines the 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.	Outbound Proxy will be mapped to the 1st entry in PREFIX table that (if exists) must look as follows: [PREFIX] FORMAT PREFIX_Index = PREFIX_DestinationPrefix, PREFIX_DestAddress, PREFIX_SourcePrefix, PREFIX_ProfileId, PREFIX_MeteringCode, PREFIX_DestPort, PREFIX_SrcIPGroupID, PREFIX_DestHostPrefix, PREFIX_DestIPGroupID, PREFIX_SrcHostPrefix, PREFIX_TransportType, PREFIX_SrcTrunkGroupID; PREFIX 0 = *, 10.8.211.180, *, 0, 255, 0, -1, , -1, , -1, -1; [\PREFIX]
OutboundProxyPort	unsignedInt [0:65535]	W	Defines the 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.	PREFIX_DestPort (see above)
RegistrationPeriod	unsignedInt [1:]	W	Defines the Period over which the user agent must periodically register, in seconds.	RegistrationTime
RegisterExpires	unsignedInt [1:]	W	Defines the Register Request Expires header value, in seconds.	RegistrationTime (the same value as RegistrationPeriod)
.VoiceService.{i}.VoiceProfile.{i}.Line.{i}.SIP.	object	–	Defines the Voice line parameters that are specific to SIP call signaling.	
AuthUserName	string(128)	W	Defines the Username used to	This and the next

Name	Type	Write	Description	Comment
			authenticate the connection to the server.	parameter will be mapped to the corresponding entry in Authentication table. Authentication_UserId On write, we will ensure that AuthenticationMode is set to 3 (PerFXS). On read, if AuthenticationMode is not set to 3 (PerFXS), empty string will be returned.
AuthPassword	string(128)	W	Defines the Password used to authenticate the connection to the server. When read, this parameter returns an empty string, regardless of the actual value.	Authentication_UserPassword On write, we will ensure that AuthenticationMode is set to 3 (PerFXS). On read, if AuthenticationMode is not set to 3 (PerFXS), empty string will be returned.

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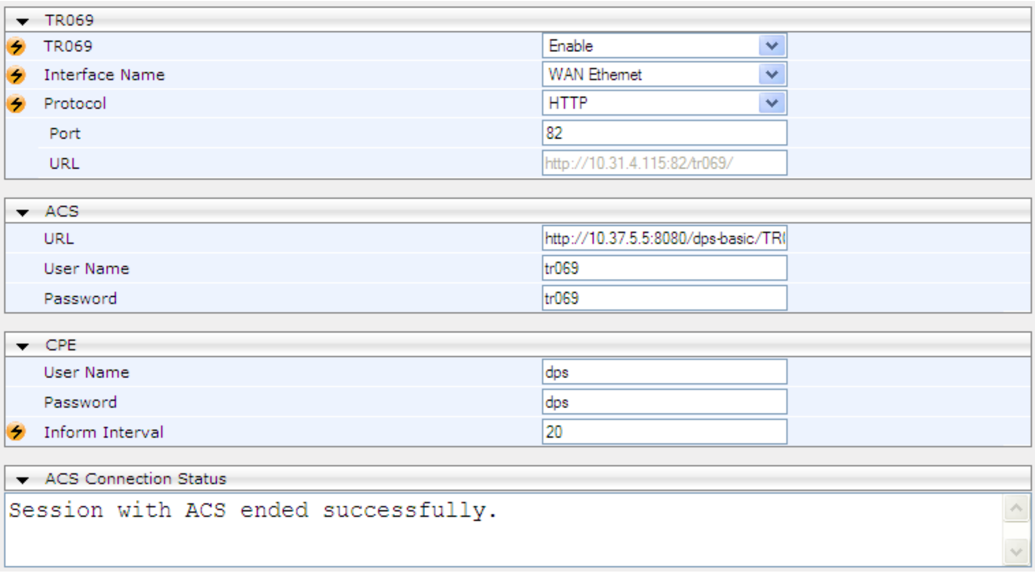
3 Configuring TR-069

The CWMP/TR-069 Settings page is used to enable and configure TR-069.

➢ **To configure TR-069:**

1. Open the CWMP/TR-069 Settings page (**Configuration** tab > **System** menu > **Management** submenu > **CWMP**).

Figure 3-1: CWMP/TR-069 Settings Page



2. Configure the parameters as required.
3. Click **Submit** to apply your changes.
4. Save the changes to flash memory

The TR-069 parameters are described in the table below.

Table 3-1: TR-069 Parameters

Parameter	Description
Web: TR069 CLI: service **[TR069ServiceEnable]**	Enables TR-069 management. ▪ **[0]** Disable (Default) ▪ **[1]** Enable **Note:** For this parameter to take effect, a device reset is required.
Web: Connection Interface CLI: interface-type **[TR069NetworkSource]**	Defines the device's network interface for the TR-069 connection. ▪ **[0]** LAN ▪ **[1]** WAN (default) **Note:** The parameter is applicable only to the MSBR product line**.**
Web: Protocol CLI: protocol **[TR069Protocol]**	Defines the protocol used for the TR-069 connection. ▪ [0] HTTP (default) ▪ [1] HTTPS
Web: Port CLI: port **[TR069HTTPPort]**	Defines the local HTTP/S port used for TR-069. The valid range is 0 to 65535. The default is 82. **Note:** For this parameter to take effect, a device reset is required.

Parameter	Description
Web: URL CLI: acl-url **[TR069AcsUrl]**	Defines the URL address of the Auto Configuration Servers (ACS) to which the device connects. For example, http://10.4.2.1:10301/acs/. By default, no URL is defined.
Web: User Name CLI: acs-user-name **[TR069AcsUsername]**	Defines the login username that the device uses for authenticated access to the ACS. The valid value is a string of up to 256 characters. By default, no username is defined.
Web: Password CLI: acs-password **[TR069AcsPassword]**	Defines the login password that the device uses for authenticated access to the ACS. The valid value is a string of up to 256 characters. By default, no password is defined.
Web: URL CLI: connection-request-url **[TR069ConnectionRequestUrl]**	Defines the URL for the ACS connection request. For example, http://10.31.4.115:82/tr069/.
Web: User Name CLI: connection-request-user-name **[TR069ConnectionRequestUsername]**	Defines the connection request username used by the ACS to connect to the device. The valid value is a string of up to 256 characters. By default, no username is defined.
Web: Password CLI: connection-request-password **[TR069ConnectionRequestPassword]**	Defines the connection request password used by the ACS to connect to the device. The valid value is a string of up to 256 characters. By default, no password is defined.
Web: Inform Interval CLI: inform-interval **[TR069PeriodicInformInterval]**	Defines the inform interval (in seconds) at which the device periodically communicates with the ACS. Each time the device communicates with the ACS, the ACS sends a response indicating whether or not the ACS has an action to execute on the device. The valid value is 0 to 4294967295. The default is 60.
[TR069RetryinimumWaitInterval]	Defines the minimum interval (in seconds) that the device waits before attempting again to communicate with the ACS after the previous communication attempt failure. The valid value is 1 to 65535. The default is 5.
CLI: debug-mode **[TR069DebugMode]**	Defines the debug mode level, which is the type of messages sent to the Syslog server. The valid value is between 0 and 3, where 0 (default) means no debug messages are sent and 3 is all message types are sent.

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