Global Knowledge Services

Touchstone Release 11.1

Firmware Guide

STANDARD Revision 2.0

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Touchstone® TS11.1 Firmware Guide
STANDARD 2.0

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Chapter 1

Overview

Touchstone devices provide the subscriber connection to the HFC IP network.

Touchstone devices running TS11.1 firmware comply to the following standards:

- DOCSIS 3.1
- PacketCable 1.5 and PacketCable 2.0

About This Manual

This manual describes Touchstone® TS11.1 firmware.

Some features described in this manual may not be fully tested and supported in your specific firmware release version. Where possible, features supported only by specific versions are indicated in this manual. See the *Release Notes/Letter of Operational Considerations* accompanying your firmware for further details.

Audience

This manual assumes that you have a basic understanding of DOCSIS and PacketCable standards, and a working knowledge of cable data and telephony networks.

In This Manual

This manual contains the following chapters:

- Chapter 1, “Overview,” describes the Touchstone firmware and documentation, standards compliance, and load variants.
- Chapter 2, “DOCSIS Provisioning,” describes provisioning the Cable Modem (CM) component of Touchstone devices.
- Chapter 3, “NCS Voice Provisioning,” describes provisioning NCS telephony services on Touchstone devices.
- Chapter 4, "Provisioning ARRIS SIP Loads," describes features and services available for ARRIS SIP loads.
- Chapter 5, “Provisioning PacketCable 2.0 SIP Loads,” describes special features and services available for SIP PC20 loads.
- Chapter 6, “Operations,” describes monitoring, fault detection, and alerting mechanisms.
- Chapter 7, “Administration,” describes performance statistics and maintaining system reliability.
- Chapter 8, “Maintenance,” describes firmware updates, diagnostics, and troubleshooting features.

- Chapter 9, "References," describes calling features, default ring cadences, and tones for each supported country code.

Supported Hardware

TS11.1 supports the following Touchstone models.

- TM3402A, TM3402B

Firmware Functionality

Touchstone TS11.1 firmware provides the following functionality:

- Supports Touchstone Model 34 products.
- Compatibility with DOCSIS 3.1.
- Supports mixed-mode provisioning (for example, DOCSIS 3.1 OFDM downstreams and DOCSIS 3.0 upstreams).
- Interoperability with ARRIS and other CMTS products.
- Supports up to 32 downstream DOCSIS 3.0 bonded channels and up to eight upstream bonded channels.
- Supports up to two 192 MHz OFDM bonded downstream channels and up to two OFDMA upstream channels.
- North American DOCSIS 3.1 loads support both 42/108 MHz and 85/108 MHz splits.
- Euro-DOCSIS 3.1 loads support 85/108 MHz and 204/258 MHz splits.
- Supports NCS, ARRIS SIP, and PC2.0 SIP telephony.
- Supports Ethernet interfaces to personal computers.
- Enhanced web-based troubleshooting interface.
- IPv4 and IPv6 addressing.
- Enhanced power management.

Firmware Download Center

ARRIS provides the ability to download firmware updates over the Internet, using the ARRIS Software/Firmware Delivery Tool. Benefits include an archive of released loads, and email notification of updated loads.

To obtain an account, contact ARRIS Technical Support.

High-level Changes from DOCSIS 3.0

DOCSIS 3.1 implements several significant improvements and changes.

- Hierarchical QoS
- Low Density Parity Code (LDPC) FEC enables use of higher-order modulation levels, including: 1024, 2048, 4096-QAM
- New downstream PHY:
  - New modulation scheme: Orthogonal Frequency Division Multiplexing (OFDM)

  - Many closely-spaced orthogonal subcarrier signals carry parallel data streams
  - Each sub-carrier uses a QAM modulation scheme at a low symbol rate
- New upstream PHY:
  - New modulation scheme: Orthogonal Frequency Division Multiple Access (OFDMA), a multi-user version of OFDM
  - Assigns groups of subcarriers to individual users
  - Allows simultaneous low data rate transmission from several users
- Larger PDU sizes
- CM reports downstream cable plant subcarrier mean error rate (MER)

To take advantage of DOCSIS 3.1 throughput enhancements, network upgrades may be needed. See the ARRIS white paper, *Preparing for DOCSIS 3.1: A Methodical Approach to Assessing Network Readiness* (*http://www.arris.com/globalassets/resources/white-papers/preparing-for-docsis-3-1.pdf*), for more information.

Standards Compliance

This section outlines Touchstone DOCSIS® and PacketCable™ compliance, and describes ARRIS-proprietary extensions to the standards.

Standard Functionality

Touchstone devices running version TS11.1 firmware comply with the following standards:

- DOCSIS 3.1
- PacketCable 1.5
- PacketCable 2.0

Optional Functionality

TS11.1 supports the following optional functionality specified by DOCSIS and PacketCable standards:

- Support for 10 ms and 20 ms packetization rates
- Support for up to 32 Upstream Service Flows
- Support for analog Fax/Modems, including automatic tone detection, echo cancellation disable and switching to the G.711 CODEC

DOCSIS Specifications

All DOCSIS specifications are available at the *DOCSIS web site* (*http://www.cablemodem.com/specifications/*).

- *DOCSIS 3.0 Radio Frequency Interface Specification*, CM-SP-RFIv2.0-I11-060206
- *DOCSIS 3.1 Operations Support System Interface Specification*, CM-SP-CM-OSSIv3.1
- *DOCSIS 3.1 Security Specification*, CM-SP-SECv3.1

- *DOCSIS 3.1 Physical Layer Specification*, CM-SP-PHYv3.1
- *DOCSIS 3.1 MAC and Upper Layer Protocols Interface Specification*, CM-SP-MULPIv3.1
- *DOCSIS 1.1 Baseline Privacy Plus Interface Specification*, SP-BPI+-I12-050812
- *DOCSIS 1.1 Cable Modem to Customer Premise Equipment Interface Specification*, CM-SP-CMCIv3.0-I01-080320
- *Layer 2 Virtual Private Networks*, CM-SP-L2VPN-I09-100611

CableLabs IPv6 Specifications

- *CableLabs Assigned Names and Numbers*, CL-SP-CANN-I01-070119
- *CableLabs DHCP Options Registry*, CL-SP-CANN-DHCP-Reg-I01-070119

DOCSIS 3.1 Security Overview

TS11.1 supports a subset of the DOCSIS 3.1 Security Specification. In practical terms, this means:

- CLI access is *ssh* only, requires the Password of the Day for access, and highly restricted on production units. Debug devices (identified by an orange enclosure) use the arristi password and have full CLI access.
- All WebGUI access uses HTTPS, rather than unencrypted HTTP.
- Debug loads are no longer provided. Instead, separate debug devices provide debugging capabilities.

Load Name Extensions

The complete load name format is **NN**.*vv.vv.XXX_date_ss.cp.ll.special.i.dcs.sb*, where:

- **NN**.*vv.vv* is the firmware version:
  - TS11.01 for TM3402
- *XXX* is the revision number
- *date* indicates the date (in MMDDYY format) when the load was built

The following extensions are used to further identify hardware and signaling support in the load.

ss

The SDK release number. For example, **70** for release 7.0.

cp

Telephony signaling type supported:

- **NCS**
- **PC15** (PacketCable 1.5 NCS)
- **PC20** (PacketCable 2.0 SIP)
- **SIP** (ARRIS SIP)

ll

Load type. Currently supported load types are:

- **03**: TM3402

special

Tags indicating special functionality (for example, **MAC14**).

i

A bitmask indicating the images included in the load:

- bit 0: ARM
- bit 1: Atom
- bit 2: UEFI
- bit 3: undefined

dcs

One of:

- **NA**: DOCSIS 3.0 load, signed with a North American certificate
- **EU**: DOCSIS 3.0 load, signed with a European certificate
- **D31**: DOCSIS 3.1 load

sb

Security indicator, one of:

- **.simg**: Secure Boot image
- **.img**: DOCSIS only signed image
- **.bin**: unsigned image

Example

The following load name:

```
TS11.01.123_121216_20.NCS.03.7.D31.simg
```

is a DOCSIS 3.1 load for a TM3402, supporting NCS telephony.

DOCSIS Provisioning

All Touchstone devices have a DOCSIS-compliant cable modem (CM) component. This chapter provides information for DOCSIS provisioning and ARRIS extensions.

General Provisioning Information

This section provides a general overview of provisioning-related information.

Service Flow Limitations

Touchstone devices support up to 32 service flows:

- Best Effort: 8
- UGS and UGS-AD: 24

OUI Ranges

TS11.1 uses 0000CA as the Vendor ID in DHCP messages. ARRIS issues periodic Field Bulletins with the most current listings.

Configuring ToD Offset

Time of Day changes, including for Daylight Savings Time shifts, normally occur during DHCP Renew operations. It may be necessary to manually change the offset, especially in SIP deployments where the eDVA clock is used as a source for Calling Line Presentation (CLIP) information.

1. To change the ToD offset, set the **arrisCmDoc30SetupTODTimeOffset** MIB object to the desired offset (in seconds). Valid range: **−43200** (−12 hours) to **46800** (+13 hours).

Note: This object is only accessible through the CM IP address.

When a DHCP exchange occurs (Renew, Rebind, or initial), the offset specified in the DHCP exchange overrides the value set in this MIB object.

Setting the DST Policy

DST policies may vary due to national legislation or local preferences. Touchstone firmware provides control over local DST policy.

1. To change the DST policy, set the **arrisCmDoc30SetupDSTPolicy** object to a string with the following format:

   start=*month*/*day*/*weekday*/*hour***;end=***month*/*day*/*weekday*/*hour*

where...	is...
month	the month: 1 for January, to 12 for December.
day	The day: **–31** to **–1** to count backwards from the end of the month, **1** to **31** to count forward from the beginning of the month.
weekday	The day of the week that DST begins or ends: **1** for Monday, to **7** for Sunday, or **0** to ignore the weekday and use the exact date. If not zero, DST begins or ends on the specified weekday after the *date* if the date is positive, or before the *date* if negative.
hour	The hour at which DST begins or ends: **00** to **23**.

   Example:

   ```
   start=3/8/7/02;end=11/1/7/02
   ```

   Implements the U.S. DST policy in effect since March 2007: DST begins at 2 a.m. on the second Sunday in March and ends at 2 a.m. on the first Sunday in November.

Clearing the CPE List

When the Telephony Modem loses link on all LAN interfaces, the modem clears its CPE list. This allows subscribers allowed only one CPE device to swap computers without resetting the Telephony Modem or calling support.

Provisioning Considerations

Typically, you provision the network using a PacketCable-compliant provisioning server. The server provides both provisioning tools to create data files, and servers (DHCP, DNS, TFTP) to store and transfer firmware loads and provisioning data to both the CMTS and all attached cable modems and eDVAs.

For upgrade considerations, see *Upgrading Touchstone Firmware* (page 29).

Interface Index Scheme

Touchstone firmware uses **ifIndex** designations to provide proper administration of multiple Gateway interfaces. The following tables list the specified defaults.

CM Interface Table

The following is the default CM interface table in TS11.1. Note that not all Touchstone devices support the maximum number of downstreams, upstreams, or SSIDs. To see the capabilities of a particular device, walk the **ifTable**.

ifIndex	ifType	Description
1	`other`(1)	eRouter embedded interface
2	`docsCableMacLayer`(127)	RF MAC interface
3	`docsCableDownstream`(128)	RF Downstream interface 1
4	`docsCableUpstream`(129)	RF Upstream interface 1
5	`usb`(160)	USB interface (if supported)
6–9	`ethernetCsmacd`(6)	Ethernet ports 1–4
12	Removed (see note)	
16	`other`(1)	eDVA interface
48–78	`docsCableDownstream`(128)	RF Downstream interfaces 2-31
80–86	`docsCableUpstream`(129)	RF Upstream interfaces 2–8
200+	`ipforward`(142)	Logical LAN IP interfaces (see below)
300+	`ipforward`(142)	Logical WAN IP interfaces (see below)
10000	`ieee80211`(71)	2.4 GHz WiFi radio interface
10001–10016	`ieee80211`(71)	WiFi SSIDs 1–16 on 2.4 GHz radio
10100	`ieee80211`(71)	5.0 GHz WiFi radio interface
10101-10116	`ieee80211`(71)	WiFi SSIDs 1–16 on 5.0 GHz radio

Interfaces 200 through 207 define logical LAN subnets, as defined by DHCP. Subnets can tie together one or more physical interfaces; for example, the Ethernet ports and the subscriber's SSIDs. The **ifAdminStatus** for these interfaces is read-only.

Interfaces 300 through 302 define logical WAN subnets. Currently, **ifIndex** 300 is the CM interface as seen from the WAN, and 301 is the eRouter interface as seen from the WAN.

Interface Types

The IANAifType textual convention defines the interface types associated with entries in the **ifTable**. Interface types used in Touchstone products include:

ifType	Description
1	Other (DOCSIS and eRouter interfaces)
6	Ethernet
16	eDVA PacketCable interface
71	IEEE802.11 (wifi) interface
127	DOCSIS cable
MAC layer	
128	DOCSIS cable downstream
129	DOCSIS cable upstream
142	IP forwarding (used for logical subnets)
160	USB
198	eDVA telephony line (Voice over Cable)

Configuration File Provisioning Notes

CM and eDVA provisioning files, as described in DOCSIS and PacketCable specifications, use TLV (Type/Length/Value) objects to specify configuration parameters. This section provides information useful in provisioning Touchstone products through configuration files.

Support for TLV-41 (Downstream Channel Lists)

TLV-41 provides Downstream Channel List support. When provisioned in the CM configuration file, downstream channel lists provide the ability to specify an allowed range of downstream frequencies to use during downstream scanning operations.

When Downstream Channel Lists are specified in the configuration file, the CM does not use any frequencies outside of the provisioned range without specifically being directed to do so by the CMTS. Also, this list overrides the last operational channel value stored in NVRAM. If the CM (portion of the E-UE) loses sync with the CMTS, the CM retains the provisioned list of downstream channels provided in the configuration file, and uses them to search for a new downstream during subsequent MAC re-initialization and downstream scanning.

Full details on the operation of this feature and TLV-41 parameters can be found in Appendix C of the DOCSIS 2.0 RFI specification (SP-RFIv2.0-I11-060206).

Provisioning Considerations for European Loads

Provisioning changes are necessary to transition from a Touchstone North American load to the EURO load. The basic flow sequence remains the same; however, some OIDs change from the North American PacketCable MIBs to the IETF PacketCable MIBs. The objects that require OID changes are highlighted below.

Flow MTA15: SNMP Enrollment INFORM

pktcMtaDevProvisioningEnrollment and its contents:

- **sysDescr** (remains the same)
- **pktcMtaDevSwCurrentVers**
- **pktcMtaDevTypeIdentifier**
- **ifPhysAddress** (previously **pktcMtaDevMacAddress**)
- **pktcMtaDevCorrelationId**

Flow MTA19: SNMPv3 SET

- **pktcMtaDevConfigFile**
- **pktcMtaDevProvConfigHash**
- **pktcMtaDevProvConfigKey**

Flow MTA23: TFTP Configuration File Request

North American PacketCable configuration files must be updated to use Euro-PacketCable OIDs.

Flow MTA25: SNMP INFORM

pktcMtaDevProvisioningStatus and its contents:

- **pktcMtaDevConfigFile**
- **pktcMtaDevProvConfigHash**
- **pktcMtaDevProvConfigKey**

Persistence

Some settings are optionally persistent; that is, stored in non-volatile memory in the E-UE. To clear persistent settings, reset the E-UE to factory defaults (see see "*Reset to Factory Defaults* (page 286) for details).

Persistence has two purposes, with different behaviors:

Changing default behavior:

Special settings may be required to address subscriber-specific issues. A common case is to disable pulse dialing when wiring or CPE issues dial "phantom" digits. Instead of creating a special configuration file, set the **arrisMtaDevEndPntDialingMethod** object in an SNMP browser. This type of persistent setting overrides configuration file settings.

Some objects using this type of persistence provide an `ignore` value that clears the non-volatile memory setting. This can be used to restore a factory default value without resetting the entire E-UE to factory defaults.

Setting pre-provisioning behavior:

Some settings need to configure low-level hardware before the CM component downloads its configuration file. Changes are always persistent, whether made in the configuration file or an SNMP browser, and often require a reset for the change to take effect. Once the value is stored and the reset occurs, setting the same value (in the configuration file) does not affect further operation. Changing the setting in the configuration file would again cause a hardware reset next time the CM downloads its configuration file.

Cable Modem Interface Mask

TS11.1 supports the Cable Modem Interface Mask (CMIM). The CMIM is a field in the upstream classifiers that can be used to filter out traffic based on the CM interface receiving the packet. When this field is present in the US classifier, a packet matches the classifier only if the traditional fields match and the source interface of the packet is present in the CMIM of the classifier.

The CMIM is an encoded 2- or 4-byte mask where each bit represents the interface whose **ifIndex** matches the bit position. The short form of the mask omits bits 16 through 31. For example, the RF interface has an **ifIndex** of `2` which corresponds to bit 2 of the CMIM. For this mask, bit position 0 is the most significant bit of the most significant word. For example, a CMIM classifier intended to match all of the CPE ports (external interfaces) of a CM has a CMIM value setting bits 1 and 5-15, so an encoding of either `0x47FF` or `0x47FF0000` is valid. See *Interface Index Scheme* (page 21) for a list of valid **ifIndex** values.

DOCSIS 3.1 Security Considerations

TLV-81 and TLV-82 support secure software download in DOCSIS 3.1. Specifying certificates for secure download differs from DOCSIS 3.0 as follows:

- DOCSIS 3.0 supports a two-level certificate hierarchy: Manufacturer CVC (specified in TLV-32), and Co-signer CVC (specified in TLV-33).
- DOCSIS 3.1 supports the DOCSIS 3.0 method, but adds a three-level certificate chain: DOCSIS root CA certificate, CableLabs CVC sub-CA certificate, and the Manufacturer or Co-signer CVC.
    - Use TLV-81 to specify a Manufacturer CVC chain (sub-CA certificate and Manufacturer CVC).
    - Use TLV-82 to specify a Co-signer CVC chain (sub-CA certificate and Co-signer CVC).

TLV-81 and TLV-82 require the two certificates be formatted into a degenerate PKCS7 signedData structure.

ARRIS has DOCSIS 3.1 training that covers this and other topics. For more information, see ask.arris.com.

CM DHCP Interactions

When a Touchstone E-UE registers, the CM and eDVA make separate DHCP and TFTP requests.

The following are CM-side interactions with the DHCP server.

DHCP Option 51 Support

DHCP option 51, described in RFC 2132, allows a client device to request a particular lease time for its IP address. The option contains a 32-bit number specifying the requested lease time in seconds.

Touchstone firmware sends DHCP option 51 in DHCP Request messages during IP address renew and rebind operations.

DHCP Option 60 Support

TS11.1 uses DHCP option 60 (Vendor Class Identifier) in DHCP Discover messages to specify the DOCSIS support required. The option contains the string “`docsis 3.1`” to indicate DOCSIS 3.1 support.

Dual-Mode Operation

Some Touchstone devices are available in a dual-mode version that can configure its CM component for either DOCSIS or Euro-DOCSIS operation based on the type of downstream first detected.

When a dual-mode device ranges and registers for the first time, it stores the detected signal type in non-volatile memory. During subsequent reboots, it automatically scans for the stored signal type.

If a dual-mode device is moved to a plant with a different signal type, it uses several methods to detect and lock to a new signal. For example, if a dual-mode device had originally ranged and registered on a North American DOCSIS plant, and then was moved to a Euro-DOCSIS plant, it would use the following procedure:

1. Scan all cached frequencies, attempting to detect (in order):
   - QAM256 carrier using Annex A
   - QAM64 carrier using Annex A
   - QAM256 carrier using Annex B
   - QAM64 carrier using Annex B
2. Scan all preset frequencies, checking for carriers as above.
3. Perform up to three general scans, checking for carriers as above.
4. Reboot.

Forcing Provisioning Mode and Certificates

To force North American PacketCable provisioning on a dual-mode device that would otherwise default to Euro-PacketCable provisioning, set the **arrisCmDoc30SetupPacketCableRegion** object to **northAmerican**(0). When overridden, the device continues to use the European root certificate.

Configuration Files and Signed Loads

Use European CVCs in configuration files for Dual Mode units. This allows configuration files for a normal European unit to work for Dual Mode devices as well. If the firmware load for a Dual Mode unit is signed, it should be European signed.

Certificates

Dual Mode Telephony Modems are programmed with four certificates, a European and North American certificate each for the CM and eDVA.

When the Telephony Modem boots up, it checks the value of the Dual Mode Discovered Market (DMDM) stored in NVM, and uses the certificates that correspond to that region. If the DMDM value is uninitialized, which would be the case the first time the Telephony Modem is installed or after a factory reset, it uses European signed certificates.

DMDM Status

Use the **arrisCmDevDualModeDiscoveredMarket** MIB object to retrieve the current DMDM value. Non-Dual Mode Telephony Modems always return **0** for this object.

Overriding the MDD IPv4/IPv6 Selection

Use the **docsIf3CmMdCfgIpProvMode** object to override IPv4 or IPv6 selection in the MDD message. The allowed values are:

- **ipv4Only**(0): override the MDD setting and use IPv4.
- **ipv6Only**(1): override the MDD setting and use IPv6.
- **honorMdd**(4): (default) use the IP mode set in the MDD message.

When setting this object, always use index .2 (the CATV MAC interface). Other index settings may prevent the eDVA from registering.

CAUTION
Potentially service-affecting
Override settings can potentially prevent the eDVA from registering. For example, setting this object to **ipv6**(2) when no DHCPv6 server is available causes the eDVA to attempt to register as IPv6 only, ignoring any DHCPv4 servers. Since this setting is stored in non-volatile

memory, you must either change the value through SNMP or reset the Telephony Modem to factory defaults to clear the setting.

Note: In TS11.1 and newer loads, the ARRIS-proprietary object **arrisCmDoc30SetupMddIpModeOverride** provides the same functionality as the DOCSIS 3.0 object. Setting this object to `apm`(3) or `dpm`(4) is now equivalent to the default disable (or `honorMdd`) functionality. ARRIS recommends using only one of the two objects, as setting both objects in the configuration file with conflicting values may have unexpected results.

If you set this object with an SNMP browser, and the new setting differs from the current operation mode, the CM resets to apply the new setting.

This object setting persists across reboots. Restoring the Telephony Modem to factory defaults resets the value to `disable`(0).

Displaying the MDD Setting

You can view the current MDD setting and override using the “DHCP Parameters” troubleshooting page (select **DHCP** in the Advanced pages). This page displays both the MDD Override setting and the selected MDD mode.

The CM generates log messages in response to MDD settings or overrides:

- “MDD IP mode Set” — override disabled. The log message shows the mode set by the MDD message.
- “MDD IP mode Override” — override enabled. The log messages shows the mode set by the MIB object.

Configuring Extended Upstream Transmit Power

TS11.1 supports extended upstream transmit power capabilities, as defined in CM-SP-MULPIv3.0-I20-121113.

To enable extended upstream transmit power, set the **arrisCmDoc30SetupExtendedUpstreamTransmitPowerValue** object to the desired maximum value, in 0.25dB increments. Valid range: `205` to `244`, or `0` to disable.

Note: Not all Touchstone devices support the entire range of values. The default value of the **arrisCmDoc30SetupExtendedUpstreamTransmitPowerValue** object is the maximum supported for the device.

When enabled, Touchstone devices report extended upstream transmit power capabilities in the REG-REQ-MP message.

Upgrading Touchstone Firmware

Use this procedure to upgrade from previous versions of Touchstone firmware.

Action

Perform the following tasks as needed.

Upgrading the Firmware through Provisioning

Follow these steps to upgrade the Touchstone firmware load using a provisioning server.

1. Install the new firmware on the TFTP server.
2. Use the provisioning server to add or verify the following items in the cable modem configuration file:
   - **ManufacturerCVC** (the CVC, needed only for secure downloading)
   - **UpgradeFileName** (file name of the firmware load)
   - **UpgradeServer** (IP address of the server containing the load)
   - **SnmpMib = docsDevSwAdminStatus.0 2** (allowProvisioningUpgrade)
3. During the maintenance window, use your provisioning server or element manager to reset each Touchstone E-UE.

   The E-UEs download the new firmware, then reset.
4. Verify that the E-UE has the new load by checking the value of the **docsDevSwOperStatus** object (using an SNMP server).

   The value should read `completeFromProvisioning`(3).

Upgrading the Firmware through SNMP

Follow these steps to upgrade the Touchstone firmware load using an SNMP manager.

1. Using the provisioning server, add the ManufacturerCVC to the configuration file.
2. Using the SNMP manager, set the following **docsDevSoftware** objects:

   docsDevSwServerAddressType
   Set to `1` for IPv4 server addressing or `2` for IPv6 addressing.

   docsDevSwServerAddress
   The IP address of the server containing the load.

docsDevSwFilename
The file name of the load.

docsDevSwAdminStatus
Set to **`upgradeFromMgt`**(1).

The E-UE downloads the new firmware, then resets.

3. Verify that the E-UE has the new load by checking the value of the **docsDevSwOperStatus** object.

   The value of the object should read **`completeFromMgt`**(3).

Configuring Channel Bonding Characteristics

Channel bonding is set up in the CM configuration file. The MIB objects in this procedure provide some extra control over, and monitoring of, channel bonding.

Action

Follow these steps to configure channel bonding characteristics.

1. To disable downstream channel bonding (effectively forcing DOCSIS 2.0 behavior), set the **arrisCmDoc30SetupDsBonding** object to **`disable`**(0).

Note: The object setting takes effect at the next reboot, and persists across reboots. If this object is set to **`disable`**(0), the modem disables DOCSIS 3.0 operation until re-enabled.

2. To configure how the Telephony Modem handles partial service situations, proceed to *Recovery from Partial Service* (page 203).
3. To display the current bonding mode, query the **arrisCmDoc30BondingMode** object. This object contains a string showing the current DOCSIS operating mode and the number of bonded downstream and upstream channels.

   Examples:

```
DOCSIS3.0 4x1
DOCSIS2.0 1x1
```

IPv6 Provisioning Notes

This section describes provisioning modes and flows appropriate to IPv6 support.

IPv6 Provisioning Modes

Touchstone firmware supports the following provisioning modes for IPv6:

- SECURE (full PacketCable)
- BASIC.1/2 (PacketCable and ARRIS versions)

- HYBRID.1/2
- PacketCable Minus KDC

Single MAC provisioning is explicitly not supported.

Selecting an Addressing Mode

Touchstone firmware supports the DOCSIS 3.0 MAC Domain Descriptor (MDD) message, defined in the *MAC and Upper Layer Protocols Interface Specification*, CM-SP-MULPIv3.0-I20-121113, for selecting IPv4 or IPv6 addressing. The CM uses TLV 5.1 in the MDD to select the addressing mode as follows:

Value	Addressing Mode
0	IPv4
1	IPv6
2	Alternate Provisioning Mode (APM): try IPv6 first, then IPv4
3	Dual Provisioning Mode (DPM)

If the E-UE does not find an MDD during provisioning, it always selects IPv4 addressing.

The **docsIf3CmMdCfgIpProvMode** object can override the MDD and set IPv4 or IPv6 operation.

CAUTION
Potentially service-affecting
Use this feature carefully. An invalid setting could isolate the Telephony Modem from the network.

The supported values are:

- **`ipv4Only`**(0): force IPv4 addressing
- **`ipv6Only`**(1): force IPv6 addressing
- **`honorMdd`**(4): Use the MDD to determine the address type

The **arrisCmDoc30SetupMddIpModeOverride** MIB object can override the MDD and set IPv4 or IPv6 operation.

DHCP Behavior for IPv6 Provisioning

When the CM receives an MDD message that specifies IPv6 operation, it acquires its IP address according to the *MAC and Upper Layer Protocols Interface Specification*, CM-SP-MULPIv3.0-I20-121113. TS11.1 supports both SLAAC and DAD mechanisms.

TS11.1 supports the DHCPv6 options listed in the following table. For details, see the *DOCSIS 2.0 + IPv6 Cable Modem Technical Report*, CM-TR-DOCSIS2.0-IPv6-V01-080307.

Option #	Sub-Option	Name
1		Client Identifier option (DUID)
2		Server Identifier Option
3		IA_NA option (IPv6 address)
6		Option Request Option
14		Rapid Commit Option
19		Reconfigure Message option
20		Reconfigure Accept Option
17		Vendor-specific information option
	32	TFTP Server Addresses option
	33	Configuration File Name option
	34	Syslog Server Addresses option
	35	TLV5 Encoding
	36	DOCSIS Device Identifier option
	37	Time Protocol Servers option
	38	Time Offset option

TS11.1 supports the DHCP Reconfigure message described in RFC 3315. Upon receiving a DHCP Reconfigure message, the CM validates the message then acquires updated DHCP parameters from the server.

Provisioning File Notes

The configuration file must be specific to either IPv6 or IPv4. If the plant has mixed IPv4 and IPv6 CMs, each address type requires separate provisioning files.

For IPv6 configuration, addresses must be fully qualified and not compressed. For example, an IPv6 address of **2001:0200:0000:0000:0000:0000:0000:0022** cannot be entered as **2001:0200::0022**.

TLV-38 Enhancements

TS11.1 supports TLV-38 (Notification) sub-type 8 (SNMP notification IPv6 address). Specify this sub-TLV to send SNMP traps and informs to an IPv6-configured receiver.

Configuring the Diplexer

Some Touchstone DOCSIS 3.1 devices allow diplexer configuration, to adjust the upstream and downstream frequency ranges to accommodate the HFC plant.

The diplexer capabilities depend on the market as follows:

North American models:

- Upstream: 5 MHz to 42 MHz, or 5MHz to 85 MHz
- Downstream: 108 MHz to 1002 MHz (fixed)

European models:

- Upstream: 5 to 85 MHz, or 5 to 204 MHz
- Downstream: 108 to 1218 MHz, or 258 to 1218 MHz

To configure the diplexer:

1. Read the **arrisCmDoc30DiplexerFreqencyRanges** object to determine which frequency ranges the device supports. The result is a string, showing the upstream/downstream ranges for each band, similar to the following:
   `Band0: 5-85MHz/108-1002MHz; Band1: 5-42MHz/108-1002MHz`
2. Set the **arrisCmDoc30DiplexerControl** object to the desired band: **`0`** for Band 0, **`1`** (the default) for Band 1.
3. Reset the Touchstone device for the change to take effect.

   The setting persists across reboots, and is not affected by a reset to factory defaults.

NCS Voice Provisioning

All Touchstone Telephony Modems and Telephony Gateways provide telephony service through an eMTA (also known as eDVA) component. This chapter provides information for provisioning NCS telephony.

NCS Provisioning Considerations

About IPsec

IPsec (Internet Protocol Security) is a collection of Internet standards used to encrypt and authenticate IP packets, to provide message integrity and privacy. IPsec provides security at the network layer (all TCP and UDP packets, and layers above).

IPsec is controlled by setting the **pktcMtaDevCmsIpsecCtrl** object for each CMS that the eDVA can communicate with; you can include this object in the eDVA configuration file. The object is indexed by the CMS FQDN for North American loads. Set the object to **`true`**(1) to enable IPsec between the eDVA and a particular CMS, and **`false`**(2) to disable it.

Note: Touchstone E-UEs use only the IPsec ESP transport mode.

Call Management Servers

Touchstone firmware accepts up to 64 call management server IP addresses identified in the eDVA configuration file. Each call server DNS entry can have up to six IP addresses associated with it, so assigning multiple IP addresses to a CMS reduces the total number of unique servers that can be listed. Support for multiple CMSs allows for load balancing, where an eDVA can be redirected to use a CMS with a lighter load.

When IPsec is activated, Touchstone E-UEs store up to 10 security associations, limiting the number of CMSs that it can communicate with at any given time. However, by setting the “CMS Redirect” bit (0x00400000) in the CallP Feature Switch (see *CallP Feature Switch* (page 46)), and by listing up to 9 CMSs in the configuration file, the Telephony Modem can bypass the 10-CMS limit, and support redirection to any other CMS on the customer network.

Voice and Signaling Ports

TS11.1 firmware uses a random selection of ports in the range 49152 through 65535 for RTP- and RTCP-based voice communications. The port numbers can be modified using the **arrisMtaCfgRTPDynPortStart** and **arrisMtaCfgRTPDynPortEnd** objects.

By default, the eDVA uses port 2727 on the upstream, and port 2427 on the downstream, to send and receive signaling information. You can change the default port number in the eDVA configuration file. You can also change the transmit port by sending an NCS message from the call server once the eDVA is operating.

eDVA Interface Table

The default **ifTable** for the eDVA is:

ifIndex	ifType	Description
1	**`other`**(1)	DOCSIS Embedded Interface
9	**`voiceOverCable`**(198)	Telephony Line 1
10	**`voiceOverCable`**(198)	Telephony Line 2
11	**`voiceOverCable`**(198)	Telephony Line 3 (if available)
12	**`voiceOverCable`**(198)	Telephony Line 4 (if available)

Provisioning Modes

Touchstone firmware supports PacketCable provisioning modes.

PacketCable Provisioning Modes

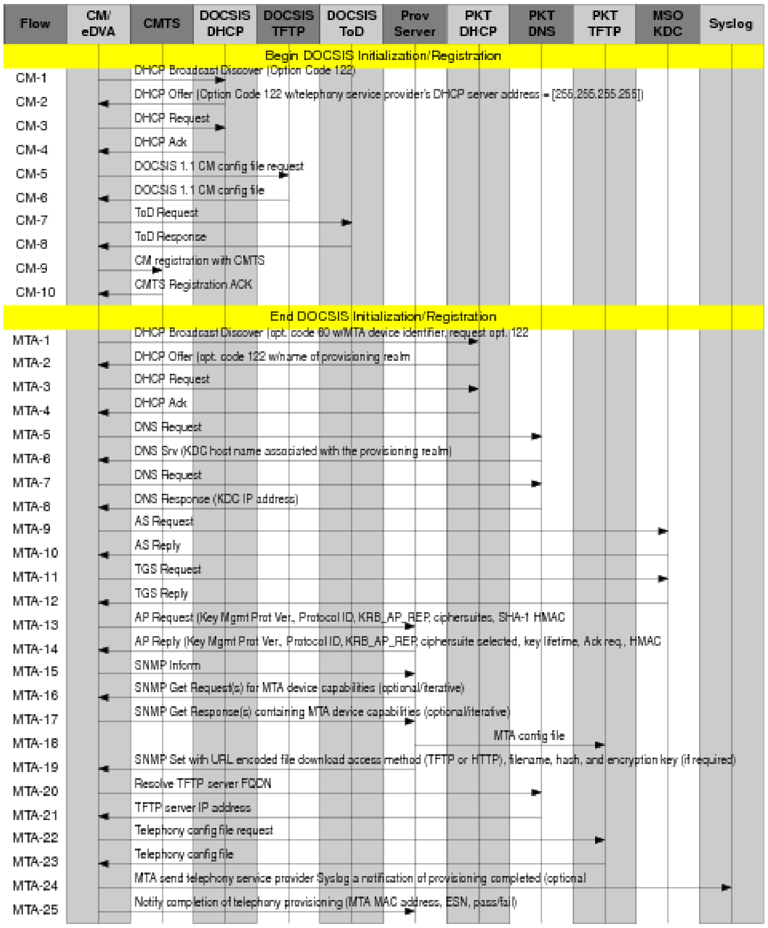
TS11.1 supports the standard PacketCable provisioning modes.

PacketCable SECURE (Full PacketCable)

This mode is also called “Full PacketCable,” and is the default provisioning mode. The data and telephony components have unique IP addresses, MAC addresses, and configuration files (i.e. two of each per E-UE). When the E-UE registers, it makes two separate DHCP and TFTP requests.

SNMP communication uses SNMPv3, sending an SNMPv3 INFORM. The E-UE and provisioning system support Kerberos mutual authentication and Kerberized SNMPv3 messaging.

IPsec is supported, and may be enabled or disabled using the **pktcMtaDevCmsIpsecCtrl** object (enabled by default). Media encryption (voice security) can be enabled on a per-call basis using NCS signaling (the LCO/SDP options) or disabled per eDVA using a feature switch. The feature switch is stored in NVRAM. The following diagram shows the full PacketCable event sequence. All events are included.

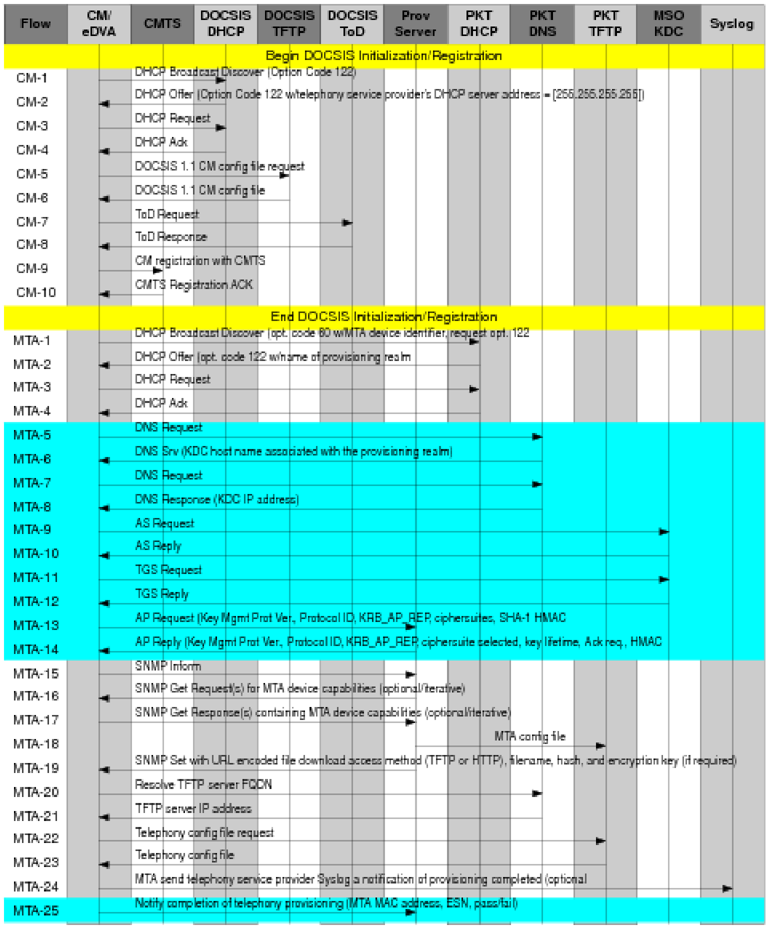


PacketCable HYBRID

Similar to the ARRIS-proprietary “PacketCable without KDC” provisioning mode. HYBRID flows are identical to the SECURE flow but remove the Kerberos message exchange, and use SNMPv2c instead of SNMPv3.

There are two HYBRID flows, HYBRID.1 and HYBRID.2; the primary difference between the two is that HYBRID.2 uses the “provisioning complete” SNMP INFORM. IPsec is disabled.

Media encryption can be controlled on a per-eDVA basis using a feature switch. The following diagram shows the PacketCable HYBRID event sequence. This sequence skips several events in the eDVA provisioning; the shaded steps below are skipped.



Note: Only HYBRID.1 skips step MTA-25.

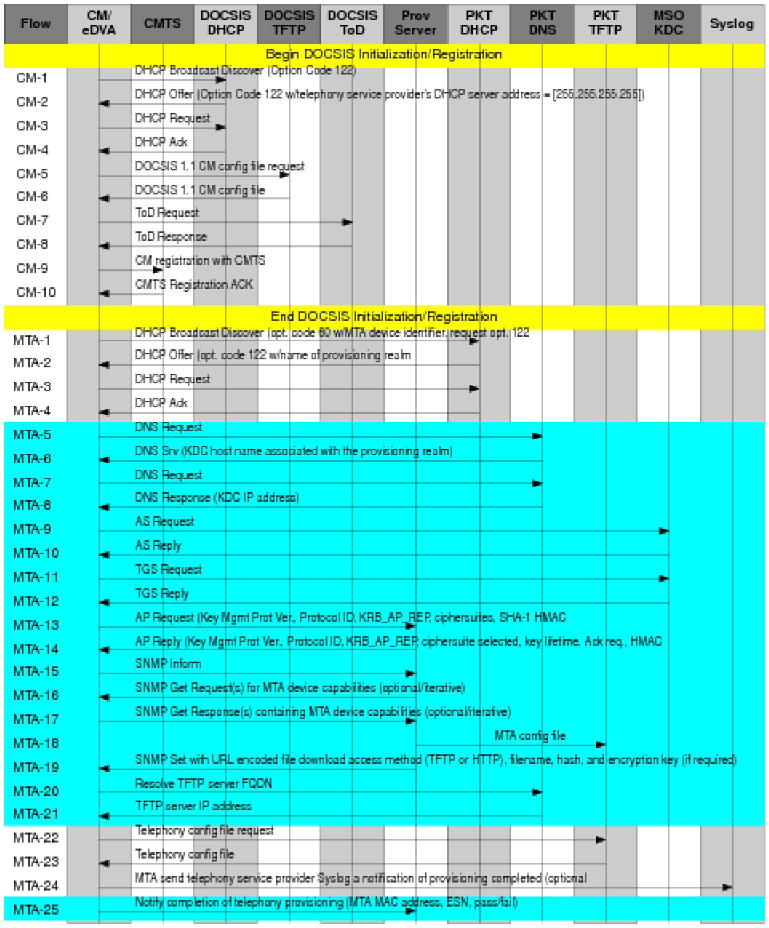
PacketCable BASIC

Simplified provisioning flows with no Kerberos or SNMPv3 security, and no SNMP enrollment using SNMP INFORM.

There are two BASIC flows, BASIC.1 and BASIC.2; the primary difference between the two is that BASIC.2 uses the "provisioning complete" SNMP INFORM.

When using a PacketCable BASIC mode, the downloaded configuration file must contain the MIB object **pktcMtaDevProvConfigHash**. The eDVA calculates the hash value of the provisioning file and verifies that the calculated hash and the hash value contained in the MIB object match. If they do not match, provisioning fails.

Note: PacketCable BASIC requires that the eDVA provisioning file contain the **pktcMtaDevProvConfigHash** object, with a value equal to the hash of the provisioning file. The eDVA calculates the hash and compares it to the value of this object; if the values do not match, provisioning fails. The following diagram shows the sequence for PacketCable BASIC.1. This sequence skips several steps in the eDVA provisioning. The PacketCable BASIC.2 sequence is nearly identical to BASIC.1, but does not skip the last step in the eDVA provisioning.



Verifying eDVA Provisioning and Endpoint Status

The **pktcMtaDevProvisioningState** object indicates the status of the eDVA initialization process. The MIB object **pktcNcsEndPntStatusError** indicates whether the endpoint has successfully registered with the call server.

ARRIS-proprietary Provisioning Modes

To improve compatibility with non-compliant equipment, ARRIS provides several provisioning modes for Touchstone E-UEs and firmware. Each mode has multiple options to enable and disable PacketCable features. To select a proprietary provisioning mode, set the **arrisMtaDevProvMethodIndicator** to the desired mode.

DHCP Support by Provisioning Mode

The following sections list DHCP parameters used by each provisioning mode.

Options Required in All Provisioning Modes

The following DHCP options are required in all CM and eDVA offers. The E-UE cannot function without a subnet mask and at least one router, DNS server, and Syslog server.

Option	Description
1	Subnet mask
3	IP address of the gateway router (one or more)
6	IP address of the DNS servers (one or more)
7	IP address of the log servers (one or more)

Note: In addition to the required options listed above, ARRIS strongly recommends including option 4 (IP address of the ToD server) in all CM and eDVA offers.

PacketCable Modes

These options are valid for Full PacketCable and PacketCable minus KDC provisioning modes. Note that options 122 and 177 are mutually exclusive (specify one or the other, not both). Option 177 is the default for PacketCable minus KDC.

CM DHCP Option 4: ToD Server IP Address

CM DHCP Option 122:

- SubOption 1: Service Provider’s Primary DHCP (required)
- SubOption 2: Service Provider’s Secondary DHCP (optional)

eDVA DHCP Option 122:

- SubOption 3: Service Provider’s SNMP Entity (required)
- SubOption 4: AS REQ/REP Exchange Backoff and Retry for SNMPv3 Key Management (optional)
- SubOption 5: AP REQ/REP Exchange Backoff and Retry for SNMPv3 Key Management (optional)
- SubOption 6: Kerberos Realm (FQDN) (Full PacketCable, Basic.1, Basic.2: required)
- SubOption 7: Authorization method (TGT for eDVA) (optional)
- SubOption 8: Provisioning Timer (minutes) (optional)
- SubOption 9: Security Ticket Invalidation (optional)

CM DHCP Option 177:

- SubOption 1: Service Provider’s Primary DHCP (required)
- SubOption 2: Service Provider’s Secondary DHCP (optional)

eDVA DHCP Option 177:

- SubOption 3: Service Provider's SNMP Entity (required)
- SubOption 4: Service Provider Network Primary DNS
- SubOption 5: Service Provider Network Secondary DNS
- SubOption 6: Kerberos Realm (FQDN)
- SubOption 7: Authorization method (TGT for eDVA)
- SubOption 8: Provisioning Timer (minutes)

Note: By default, the E-UE performs a "reinit" when it receives an SNMP Entity (sub-option 3) that differs from the original entity IP address. The **arrisMtaDevDhcpOptionOverride** MIB object allows you to provision the E-UE to accept a changed SNMP entity. This may be necessary for operation with certain DHCP servers that reassign SNMP entities for load-balancing.

eDVA DHCP Interactions

When a Touchstone E-UE registers, the CM and eDVA make separate DHCP and TFTP requests.

The Telephony Modem eDVA component sends various information to the provisioning server using the DHCP options described below.

DHCP Option 43 Support

TS11.1 sends DHCP option 43 (Vendor-Specific Information) in eDVA DHCP Discover messages with the following sub-options:

Sub-opt.	Name	Value
2	Device Type	"EDVA"
4	Serial Number	Varies (e.g. "997BNW87D747320")
5	HW Version	Hardware version of eDVA (e.g. "2.0")
6	SW Version	Firmware version (e.g. "9.1.115")
7	Boot ROM	Boot ROM version (e.g. "1.2.1.20")
8	Vendor ID	"0000CA"
9	Model Number	E-UE model number (e.g. "TM2472A")
10	Vendor Name	"Arris Interactive, L.L.C."
31	MTA MAC Addr.	Varies (e.g. "0015d004153d")
32	Correlation ID	Varies (must match the value of the **pktcMtaDevCorrelationId** object).

DHCP Option 51 Support

Touchstone firmware sends DHCP option 51 in DHCP Request messages during IP address renew and rebind operations.

DHCP Option 60 Support

TS11.1 uses DHCP option 60 in DHCP Discover messages to advertise PacketCable support. The option contains the string “pktc2.0” to indicate PacketCable 2.0 support.

The following table lists the sub-options sent with DHCP option 60 and their meanings:

Sub-option	Value	Description
1	0x02	PacketCable 2.0
2	0x02	Number of endpoints (2)
3	0x00	TGT support (no)
4	0x00	HTTP download file access method (no)
5	0x01	MTA-24 Event Syslog notification
9	0x01	NVRAM Ticket/Session Key Storage supported
10	0x01	Provisioning Event Reporting supported
11	0x0106090f	Supported CODECs:
		01 = other
		06 = PCMU
		09 = PCMA
		0F = telephone-event
13	0x01	Echo Cancellation supported
16	0x09	**ifIndex** of first phone line
18	0x0007	Supported provisioning flows: Secure, Hybrid, Basic
19	0x01	T.38 Version support (v0)
20	0x01	T.38 Error Correction support (redundancy)
21	0x01	RFC 2833 DTMF relay supported
22	0x01	Voice Metrics supported
23	0x02001f	CableLabs MIBs supported: EN-SIG, EN-MTA, EVENT-MIB, SIG-MIB, MTA-MIB
24	0x00	Multiple Grants per Interval (no)
25	0x00	V.152 support (no)
26	0x00	Certificate Bootstrapping (no)

Disabling Option 122 Sub-Option 3 Enforcement

Touchstone firmware can be configured to ignore the “SNMP Entity” (DHCP Option 122 Sub-Option 3) comparison checks during eDVA DHCP Renew and Rebind processing. This may be necessary if you want to allow the eDVA to direct Link Up and Link Down traps to an alternate trap server.

To configure this setting, set the **arrisMtaDevDhcpSubOpt3Immediate** object to `off`(1) to disable the override, or **on**(2) to enable the override.

Provisioning Quality of Service

DOCSIS and PacketCable standards specify the use of service flows to separate and prioritize voice and data traffic. Touchstone firmware provides two QoS modes.

See CM-SP-MULPIv3.1 for detailed descriptions of Service Flow types.

Full DQoS Mode

Touchstone firmware defaults to Dynamic Quality of Service (DQoS) provisioning. Full DQoS simplifies provisioning by requiring only that the primary Best Effort (BE) and MGCP (signaling) flows be provisioned. The firmware dynamically sets up and tears down UGS service flows, using a standard set of parameters designed for efficient use in DOCSIS-based networks, as needed. The CMS controls the bandwidth authorization as specified in the PacketCable *Dynamic Quality of Service* Specification, PKT-SP-DQOS1.5-I03-070412.

Full DQoS provides an added layer of security by authenticating eDVAs that contact the CMS during call setup. Each session is authorized; the session authorization uses a handle (the Gate-ID) assigned by the CMTS, passed to the CMS, and sent to the eDVA using an NCS message, to match requests with authorizations. Upon receiving call-signaling information, the eDVA passes the Gate-ID to the CMTS in a DSA/DSC message.

DSX QoS Mode

Touchstone firmware supports an ARRIS-proprietary feature that implements QoS using UGS flows for voice transmission using DOCSIS 1.1 DSx messaging. This functionality provides a level of QoS in a network where the CMS and CMTS do not support the PacketCable Full DQoS model.

DSx QoS functionality can be activated using a feature switch. When activated, the firmware sends the appropriate DSx messages needed to Add/Modify/Delete the UGS service flows. DSx messages flow only between the CMTS and the eDVA, and do not involve the CMS in any validation or requests for setting up or monitoring the UGS flows.

Note: When using this functionality with the ARRIS C4 CMTS, PacketCable authorization needs to be disabled. Contact your next level of support for instructions.

CODECs and Packetization Rates Supported

In addition to supporting the required G.711 CODEC, Touchstone firmware supports the following PacketCable™ optional and recommended CODECs:

- G.729, G.729a, G.729E

- G.722 (HD voice)
- telephone-event (RFC 2833 DTMF relay)

Touchstone SIP firmware loads offer CODECs based on the contents of the **pktcEUERSTBCallPrefCodecList** objec. If you offer G.729, you should also offer DTMF relay (`telephone-event`) as compressed CODECs can distort DTMF tones sent through the network.

Implementation of CODECs is defined by the *Codec and Media Specification*, PKT-SP-CODEC-MEDIA-I09-100527.

The supported packetization rates in TS11.1 are 10 and 20 ms.

G.729 CODEC Support

Touchstone firmware supports the G.729 CODEC to perform toll-quality voice compression. The G.729 CODEC is defined in *ITU-T Rec. G.729 Annex A, Reduced Complexity 8 kbit/s CS-ACELP Speech Codec*, November 1996, and can compress voice as low as 8 kb/s. Sometimes the G.729 CODEC is referred to as G.729A, since other variations such as G.729E have been subsequently defined.

G.729 CODEC Negotiation (SIP)

PC20 SIP firmware loads offer CODECs based on the contents of the **pktcEUERSTBCallPrefCodecList** object. The following is an example array string:

```
G729,PCMU,PCMA
```

This CODEC array would set G.729 as the default with DTMF relay enabled.

If you choose to offer G.729 in this array, you should also offer DTMF relay (`telephone-event`). To specify DTMF relay for PC20 SIP loads, set the **pktcEDVADtmfRelay** object to **`true`**(1). Do not add the string to the the **pktcEUERSTBCallPrefCodecList** object.

G.729 Fax and Modem Support

Since the G.729 CODEC highly compresses audio, it can not adequately support analog fax or modem transmission. Depending upon the call setup, the endpoint responds as follows when local fax or modem tones are detected:

- If a fax tone is detected, T.38 Fax Relay is allowed by the Call Agent, and the use of T.38 is negotiated between endpoints, then a T.38 Start event is generated and the fax is transmitted using T.38 Fax Relay.
- Otherwise, if either PCMU or PCMA is allowed by the Call Agent and negotiated with the far end as a backup CODEC, the endpoint automatically switches to PCMU or PCMA respectively.
- Otherwise, the endpoint simply notifies the fax or modem detection event to the Call Agent, if specifically requested to do so. Then, the Call Agent can modify the connection to use either PCMU or PCMA.

Using PCMU or PCMA as a backup CODEC to G.729 provides the following advantages:

- Upon detection of fax or modem tones, the endpoint can quickly switch to PCMU or PCMA without waiting for specific instructions from the Call Agent to do so. This helps to prevent fax or modem failure which may occur if the endpoint has to wait too long for instructions to switch to PCMU or PCMA.
- When PCMU or PCMA is negotiated as a backup CODEC, and DQoS is in use, the CMTS reserves bandwidth between the Telephony Modem and CMTS when the call is initially set up. If the Telephony Modem needs to switch to PCMA or PCMU the necessary bandwidth is already reserved.

G.729 Bandwidth Considerations

There is one disadvantage to specifying PCMA or PCMU as a backup CODEC: when DQoS is in use, the CMTS always reserves enough bandwidth for PCMA or PCMU even when G.729 is being used. This extra bandwidth is wasted during normal voice calls, but may be used for Best Effort data traffic.

Note: When only G.729 is allowed (i.e. PCMU and PCMA are not allowed as a backup CODEC), the CMTS reserves only the bandwidth needed to carry the call. However, there is no guarantee that extra bandwidth would be available if a switch to PCMU or PCMA became necessary, which would result in a failed fax or modem call.

CODECs and Supported Lines

This section describes how low bit-rate CODECs and T.38/SuperG3 FAX relay interact.

Engineering Rules

The following rules apply to Telephony Modems and Telephony Gateways supporting DOCSIS 3.0 or newer features:

- Two instances of T.38/SuperG3 fax relay total for lines 1-2; T.38/SG3 can be supported on any line
- One instance of low bit-rate (LBR) CODEC for lines 1-2; low bit-rate CODECs can be supported on any line

The following table shows CODEC support for each line and call leg on a 2-line eDVA.

Line	1		2	
Leg	1	2	1	2
	T.38/SG3		T.38/SG3	
	LBR	G.711	LBR	G.711
	G.711	G.711	G.711	G.711
	T.38/SG3		LBR	G.711

Feature Switches

Touchstone firmware provides various feature switches to enable or disable various functionality. Some feature switches provide improved interoperability with non-PacketCable compliant equipment, while others provide features that may violate DOCSIS or PacketCable standards.

CallP Feature Switch

Touchstone firmware provides an ARRIS-specific object, **ppCfgMtaCallpFeatureSwitch**, used to configure the Telephony Modem for the specific sub-set of PacketCable features supported by the selected network configuration. This allows the flexibility to interoperate with other vendors by providing the ability to enable or disable the proper functionality. The default is full PacketCable compatibility.

The feature switch is a 32-bit value, where each bit enables or disables a certain feature. Most of these values should only be changed with the guidance of ARRIS technical support, but some flags may be changed as necessary.

The following is a list of CallP Feature Switches that can be adjusted at your discretion. The default value is **`0x0`**.

Note: SIP loads support a subset of the available bit values. The following table indicates whether each bit value is available only for NCS or for both NCS and SIP loads.

Bit	Description
0x00000001	Disable NCS Piggyback Messages (NCS only) Set this bit to disable transmission of NCS piggybacked messages (that is, sending more than one NCS message in a UDP packet). This may be required by Call Agents that do not properly handle piggybacked NCS messages. **Note**: NCS redirection may not function properly when piggybacked messages are disabled.
0x00000002	Prevent Endpoint Lockstep Quarantine Mode (NCS only) Set this bit to prevent endpoints from entering the lockstep quarantine mode. This may be required by Call Agents that leave the endpoint in lockstep mode. When this bit is clear, the gateway must receive a new Notification Request command after sending a Notify command. Until this happens, the endpoint is in a lockstep state, and events that occur and are to be detected are simply stored in the quarantine buffer until receiving the Notification Request command.
0x00000004	Show OOS instead of Idle for unprovisioned lines Set this bit to return **`oos`**(0) instead of **`idle`**(1) in the **arrisMtaDevLineCardState** object for unprovisioned lines.
0x00000008	T.38 Capability Descriptor (NCS only) Set this bit to reduce the SDP capability descriptor to only send T.38-related information.
0x00000040	Automatic OSI (NCS only) Set this bit to automatically apply OSI to both the originating and terminating sides of a call. To fully enable this functionality, set bit **`0x20000000`** as well.
0x00000080	Omit MPTIME parameter in returned SDP (NCS only) Set this bit to omit the “a=mptime” and “a=ptime” parameters in the eDVA’s SDP message.
0x00000100	Enable DOCSIS 2.0 backward compatibility for DQoS (NCS and SIP) Set this bit to enable DOCSIS 3.0 or newer Telephony Modems to properly interoperate with a DOCSIS 2.0 CMTS.
0x00000800	Force use of DQoS (NCS only) Set this bit to force the Telephony Modem to require Dynamic Quality of Service, rejecting service otherwise. The default behavior allows non-DQoS service.
0x00001000	Nuera RFC 2833 messaging without request using payload 127 (NCS only) Set this bit to instruct the eDVA to generate RFC 2833 events with a payload type of 127 without specifically being instructed to do so (for compatibility with the Nuera RDT). Clear this bit (the default) to generate RFC 2833 events using NCS signaling and SDP exchange.
0x00004000	DSx/Access only DQoS (NCS and SIP) Set this bit to use DSx/Access-only DQoS only between the CM and CMTS.
0x00008000	Disable endpoint from sending provisional responses (NCS only) Set this bit to disable sending provisional responses to the CMS if execution of

Bit	Description
	the CRCX or MDCX commands takes additional time to execute. The ARRIS eDVA sends this provisional response if DQoS is to be performed, due to the extra amount of time it takes to set up bandwidth between the CM and CMTS. Once the provisional response is issued, the CMS should stop retransmitting the command. When the eDVA has completed the transaction, it transmits a “final” response back to the CMS. This “final” response must be acknowledged by the CMS; otherwise, the eDVA retransmits it until it is acknowledged. The default is to send provisional responses in accordance with PacketCable ECN MGCP-N-02218. Set this bit to provide compatibility with CMS vendors that are not capable of supporting provisional responses.
0x00010000	Payload Header Suppression (NCS and SIP) Set this bit to allow Payload Header Suppression of voice packets between the CM and CMTS. **Note**: This bit only affects PHS for RTP voice packet streams. Its setting does not affect PHS for DOCSIS data packets, which is controlled through the DOCSIS MIB. Conversely, DOCSIS MIB settings do not affect PHS for RTP voice packets.
0x00080000	LUCENT RFC-2833 messaging without request using payload 94 (NCS only) Some CMS vendors use RFC 2833 to have the eDVA pass detected telephony events (e.g. offhook, onhook, digits) to a PSTN gateway in-band, similar to ABCD robbed-bit signaling. For compatibility with Lucent iMerge, set this bit to instruct the eDVA to generate RFC 2833 events with a payload type of 94 during call setup, without specifically being instructed to do so. Clear this bit (the default) to generate RFC 2833 events using NCS signaling and SDP exchange.
0x00100000	Allow AES encryption for RTP/RTCP (NCS only) Clear this bit (the default) to allow the negotiation of voice encryption, which is a requirement of PacketCable, and is enabled by default on a per call basis. Set this bit to disable this feature, and reduce the size of the eDVA’s SDP message since encryption parameters are not included. **Note**: If the CMS (and far end) can handle the increased size of the SDP with AES encryption enabled, then set this bit to 0, as the far end is capable of negotiating voice security parameters (including NULL encryption) and the CMS will instruct the eDVA on whether to use encryption or not on a per-call basis.
0x00400000	NCS Redirect without IPsec (NCS only) Set this bit to allow a CMS to redirect the eDVA to another CMS that is not provisioned in the eDVA CMS table, allowing the eDVA to communicate with the CMS without attempting to establish an IPsec association first. Setting this bit is for redirect cases only; the eDVA does not respond to call servers not provisioned in the eDVA’s CMS table.

Bit	Description
0x00800000	Add brackets around IP for eDVA FQDN (NCS only) Set this bit to enable the eDVA to send an NCS message with a bracketed IP address as part of the eDVA FQDN when communicating with the call server. For call servers that use IP address information instead of FQDNs, the brackets surrounding the IP address are mandatory. The following example shows the messaging format with this feature switch enabled. `aaln/1@[10.10.13.11] MGCP 1.0 NCS 1.0`
0x01000000	Send DTMF digits via RFC 2833 with operator-specified payload without request (NCS only) Set this bit to instruct the eDVA to generate RFC 2833 DTMF events using a payload type defined in the **ppCfgRfc2833DigitPayloadType** MIB object without being instructed to do so by the CMS. The default payload type value is 101.
0x08000000	Use alternate (non-sequential) Caller ID delivery order (NCS and SIP) Set this bit to have the eDVA present Caller ID in an alternate (non-sequential) order. This may be required for compliance with some CPE devices that expect Caller ID information to be presented in a non-standard format. Clear this bit (the default) to use a sequential order based on the parameter type (date/time, then number, then name).
0x10000000	Delay DLCX against connection on on-hook only line (for VMWI) (NCS only) Set this bit to delay processing of a Delete Connection message from the CMS for a line that is on-hook, to allow any queued RTP packets in the DSP jitter buffer to be played out. In the case of VMWI, this prevents the DSP connection from being closed while delivering FSK signals for VMWI. The default setting for this bit is cleared (0).
0x20000000	Enable Automatic OSI (NCS only) Set this bit to have the eDVA automatically generate OSI (Open Switch Interval) to the CPE upon far end termination of a call (i.e. the eDVA receives a DLCX command for the last connection on the endpoint). The **arrisMtaDevAutomaticOsiDelay** object specifies the delay, in 100 ms increments, before sending the OSI. The eDVA cancels OSI generation if any of the following events occur before the timer expires: ■ line goes on-hook ■ hook flash on the line ■ a new connection is created on the line ■ CMS receives OSI request The valid range for the MIB object is **`0`** (no delay) to **`100`** (10 seconds).

For non-PacketCable configuration settings, contact your ARRIS Technical Support representative.

For more information, see the *PacketCable Network-Based Call Signaling Protocol Specification*, PKT-SP-NCS1.5-I04-120412.

Example

If your configuration requires DSx-QoS, set the feature switch to include the 0x4000 and 0x10000 flags, using PacketACE or a provisioning server. If no other flags are required, the setting would be as follows:

```
SnmpMib = ppCfgMtaCallpFeatureSwitch.0 hexstr: 00.01.40.00 = 00.00.00.00
(default)
+ 00.01.40.00 (additional features)
```

CallP Feature Switches Affecting the SDP

The following feature switches affect the SDP, returned in response to a Create (CRCX) or Modify (MDCX) Connection command.

Bit	Description
0x00000008	Reduce the capability descriptor in the SDP to T38 only (default = 0, no reduction).
0x00000080	Omit mptime parameter in returned SDP (default = 0, mptime included).
0x00001000	NUERA RFC 2833 messaging without request using payload 127 (default = 0, telephone-event is negotiated normally).
0x00080000	LUCENT RFC 2833 messaging without request using payload 94 (default = 0, telephone-event is negotiated normally).
0x00100000	Allow AES encryption for RTP/RTCP (default = 1, AES encryption is negotiated normally).
0x01000000	Send DTMF digits via RFC 2833 with an operator-defined payload without request (default = 0, telephone-event is negotiated normally).

The following CRCX message is used to generate all the SDP examples, unless otherwise specified:

```
CRCX 19901 aaln/1@mta218.dev36 MGCP 1.0 NCS 1.0
C: 1234
M: recvonly
L: mp:20, a:PCMU, fxr/fx:t38-loose, xrm/mcr:on
```

The default feature switch settings are (**`0x0`** for NCS loads, and **`0x0020`** for SIP loads) generate the following SDP:

```
v=0
o=- 381749076 381749076 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
```

```
m=audio 65496 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=mptime:20
a=rtpmap:0 PCMU/8000/1
a=X-pc-secret:base64:ZNos/qs530eSDJ4FvdL2GJBR62lS5UKyQ7n9og4
IaaDbA9Blpg6lM2PfOaHEGg== U5Q5N/eWnimq9Q/yjWwY2hACRIY6a9qqEQ
Us8tm54lEmEE6LXkCB51+3sqxlQg==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 101
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=rtpmap:101 telephone-event/8000/1
a=cpar: a=fmtp:101 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

When the “Reduce Capability Descriptor” switch (**`0x00000008`**) is enabled, and all other switches are set to their default values, the SDP becomes:

```
v=0
o=- 381749076 381749076 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 65496 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=mptime:20
a=rtpmap:0 PCMU/8000/1
a=X-pc-secret:base64:ZNos/qs530eSDJ4FvdL2GJBR62lS5UKyQ7n9og4
IaaDbA9Blpg6lM2PfOaHEGg== U5Q5N/eWnimq9Q/yjWwY2hACRIY6a9qqEQ
Us8tm54lEmEE6LXkCB51+3sqxlQg==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
a=sqn: 0
a=cdsc: 1 image udptl t38
```

In this example, the Capability Descriptor is reduced to only the information required to relay support for UDPTL T.38 to the far end. This allows the far end to support T.38 strict mode as defined in the PacketCable 1.5 NCS specification.

When the “Omit mptime” switch (**`0x00000080`**) is enabled, and all other switches are set to their default values, the SDP becomes:

```
v=0
o=- 382093395 382093395 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 58810 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=rtpmap:0 PCMU/8000/1
a=X-pc-secret:base64:XI51bgXR5MNUdaKXisS0tjYCc90x3f7jA+ojyam
W/O/M2BlCaejlrRLOdApR6w== 6LbN8ULCFGMjcR2T3l1uZuBcfWM2vfGn09
YTmR6OhHfQwC4eE+WWSX7AarnFPA==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
```

```
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 101
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=rtpmap:101 telephone-event/8000/1
a=cpar: a=fmtp:101 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

In this example, neither the `a=ptime` nor the `a=mptime` parameters are included in the SDP since the packetization rate is the default (20 ms). If the Call Agent had specified a different packetization rate, then the `a=ptime` parameter is included as follows:

```
v=0
o=- 382093395 382093395 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 58810 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=ptime:10
a=rtpmap:0 PCMU/8000/1
a=X-pc-secret:base64:XI51bgXR5MNUdaKXisS0tjYCc90x3f7jA+ojyam
W/O/M2BlCaejlrRLOdApR6w== 6LbN8ULCFGMjcR2T3l1uZuBcfWM2vfGn09
YTmR6OhHfQwC4eE+WWSX7AarnFPA==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 101
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=rtpmap:101 telephone-event/8000/1
a=cpar: a=fmtp:101 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

When the “Allow AES Encryption” switch (**`0x00100000`**) is disabled, and all other switches are set to their default values, the SDP becomes;

```
v=0
o=- 381749076 381749076 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 65496 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=mptime:20
a=rtpmap:0 PCMU/8000/1
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 101
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=rtpmap:101 telephone-event/8000/1
a=cpar: a=fmtp:101 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

The last three CPFS bits (**`0x00001000`**, **`0x00080000`**, and **`0x01000000`**) are related to RFC 2833. They are used in specific configurations and are designed to skip CODEC negotiation. For example, when the "Nuera RFC2833" feature is enabled, and all other switches are set to their default values, the SDP becomes:

```
v=0
o=- 382250430 382250430 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 63672 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=mptime:20
a=rtpmap:0 PCMU/8000/1
a=X-pc-secret:base64:Qb8GFLNXP4Z3yiyxFx1Ws9vLph9qG6bTIXezUlz
rwia7iiNvPPkVYdZhZ77NEQ== zZjgwXRR2j5F04lDXefPTV06PT8g31Hn5V
Ea6NJvFPFsPiraDeDI35EI8KO+4A==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 127
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=rtpmap:127 telephone-event/8000/1
a=cpar: a=fmtp:127 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

Note that this feature does not add SDP attributes, but modifies the Capability Descriptor slightly. Payload type 127 is used for RFC 2833 support.

When the "Lucent RFC2833" switch is enabled, and all other switches are set to their default values, the SDP becomes:

```
v=0
o=- 382318343 382318343 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 49688 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=mptime:20
a=rtpmap:0 PCMU/8000/1
a=X-pc-secret:base64:Kue6n+ZSGpXrB2iAJIAUQNst6AtAS7Ad7zC3oGP
ry9XKdURiy4Y6iLaDEhk5lg== GgaMtuqF7/egjksBpQ8SZeWnXCAlr1EeNH
AHV7EY0fv03Y0MYAYa1zz/lvO5dg==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 94
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=rtpmap:94 telephone-event/8000/1
a=cpar: a=fmtp:94 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

This feature does not add SDP attributes, but modifies the Capability Descriptor. In this case, payload type 94 is used for RFC 2833 support.

Note: The three feature switches that affect RFC 2833 negotiation are mutually exclusive. At most, only one of the bits may be set in the CallP Feature Switch. Enabling multiple RFC 2833 features may result in unexpected behavior.

When the “RFC2833 Digits” switch (**0x01000000**) is enabled, and all other switches are set to their default values, the SDP becomes:

```
v=0
o=- 382360201 382360201 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 53560 RTP/AVP 0 101
b=AS:81
a=rtcp-xr:voip-metrics
a=mptime:20 -
a=rtpmap:0 PCMU/8000/1
a=rtpmap:101 telephone-event/8000/1
a=fmtp:101 0-15
a=X-pc-secret:base64:rwZISK4HN5wlZzehiOBSEJXsRQbexmiwm1Ou4pE
nXFr4lSTXQYdAsKFT5lhkkw== tWyPwAvBg6EbSs4+FoY7rWOn0l8pcQPxGm
iwlNGPfo3Suehu0CncQ2egC4JQ6w==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 101
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=fmtp:101 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

Note that this feature affects the SDP, including the Capability Descriptor. the payload type specified by **ppCfgRfc2833DigitPayloadType** is added to the m= line (the default value is “101”) and two new attributes indicate that the endpoint is prepared to receive RFC 2833 digits.

Secondary CallP Feature Switch

The secondary CallP feature switch, **ppCfgMtaCallpfeatureSwitch2**, provides more options for interoperability.

The following table lists the secondary CallP Feature Switch bits supported in TS11.1. The default value is **0x0**.

Bit Value	Description
0x00000001	(NCS only) Set this bit to treat OSI as busy.
0x00000002	Set this bit to ignore handling TDD tones.

0x00000004	(NCS only) Set this bit to remove the optional "number of channels" encoding from the CODEC list in the SDP. For example, with this bit cleared, a typical SDP CODEC entry looks like:
	`:8 PCMA/8000/1`
	With this bit set, the entry looks like:
	`:8 PCMA/8000`

Provisioning General eDVA and Line Parameters

Touchstone firmware provides a great deal of flexibility in configuring the eDVA and individual lines. This procedure covers general and miscellaneous parameters; other procedures in this chapter provide detailed information about more complex features.

Action

Perform the following tasks as needed.

Setting Persistent Line Status

Follow these steps to set persistent line status.

1. To set line status using SNMP, set the **arrisMtaDevPersistentLineStatus** object to either **`ignore`**(0) or **`forceDisable`**(1).

Note: You must set this object using an SNMP manager. If this object is included in a configuration file, the eDVA ignores the setting.

Controlling ToS Byte Marking

By default, Touchstone firmware sets the ToS byte in RTCP packets to a value of 0. Follow these steps to change the value of the RTCP ToS byte.

1. Using an SNMP network manager, change the **arrisMtaDevRtcpTosValue** object to the desired value. Valid range: `0` to `63`.

   The ToS marking changes immediately for all RTCP packets.

Note 1: The value of this object is shifted two bits before being loaded into the ToS byte.

Note 2: A common setting for ToS is `46` (expedited forwarding).

2. To make the change permanent, add the **arrisMtaDevRtcpTosValue** MIB object (with the desired value) to the eDVA provisioning file.

Controlling TurboDOX Functionality

TurboDOX is a TI-proprietary protocol, supported in hardware. Enabling TurboDOX increases performance of TCP-based protocols such as FTP and HTTP.

TurboDOX is enabled by default. You may need to disable TurboDOX during customer traffic testing and lab evaluations. ARRIS recommends that TurboDOX remained enabled in field deployments.

To provision TurboDOX functionality, add the **arrisCmDoc30SetupTurboDoxEnable** object to the CM configuration file. Set this object to `true`(1) to enable TurboDOX (the default), or `false`(2) to disable it.

Controlling IPsec Functionality

IPsec (Internet Protocol Security) is a collection of Internet standards used to encrypt and authenticate IP packets, to provide message integrity and privacy. IPsec provides security at the network layer (all TCP and UDP packets, and layers above).

Note: Touchstone E-UEs use only the IPsec ESP transport mode.

IPsec is enabled by default. Follow these steps to provision IPsec functionality.

1. For each CMS that the eDVA can communicate with, set the **pktcMtaDevCmsIpsecCtrl** object in the eDVA configuration file. The object is indexed by the CMS FQDN. Set the object to `true`(1) to enable IPsec between the eDVA and a particular CMS, and `false`(2) to disable it.
2. Reboot the eDVA to allow the new provisioning to take effect.

Configuring the Ringing Waveform

Telephony Modems support both trapezoidal and sinusoidal ringing waveforms when used with North American country templates. You may need to change the waveform if certain non-EIA compliant subscriber equipment does not recognize the trapezoidal waveform (in short, the phone does not ring).

The following defaults apply:

- North American templates: sinusoidal ringing
- All other country code templates: trapezoidal ringing

Note: Non-North American country code templates using other than 20 Hz ringing do not support sinusoidal ringing.

Configure the ringing waveform as follows.

CAUTION
Potentially service-affecting
Sinusoidal ringing is only supported for North American templates, and other templates using 20 Hz ringing. Use of this feature with other templates may result in loss of service.

1. Set the **arrisMtaDevEndPntRingingWaveform** object using one of the following methods:
    a. In the eDVA configuration file, add the **arrisMtaDevEndPntRingingWaveform** object and set it to `sinusoidal` or `normal` as desired. Then reset the eDVA to allow the change to take effect.
    b. Use an SNMP manager to access the eDVA and set the **arrisMtaDevEndPntRingingWaveform** object to `sinusoidal` or `normal` as desired.

    Note: Setting this object in the configuration file does not write the setting to NVRAM; therefore, if you remove the object from the configuration file and reset the eDVA, it reverts to using the default waveform. Setting the object through SNMP does write the setting to NVRAM.

Configuring Loop Current

Touchstone Telephony Modems support a "boost," or high loop current mode to compensate for faulty CPE or wiring. Use this setting, for example, to alleviate issues related to off-hook not being detected or fax machines failing to operate properly due to increased current draw by the equipment.

Loop current varies depending on boost mode and country template:

- Normal mode: 23 mA
- Boost mode (North American templates): 40 mA
- Boost mode (other templates): 33 mA

eDVAs using North American country templates (that is, the value of the **ppCfgMtaCountryTemplate** object is one of **northAmerica57**(1), **northAmerica33**(17), **northAmerica09**(18), or **northAmerica66**(32)) default to high (boost) loop current. Other country templates default to normal loop current.

Note: If you set the loop current through SNMP, the eDVA does not retain the setting over reboots. If you set the loop current in the configuration file, the setting is written to NVRAM and retained over reboots.

1. To specify normal loop current (the default for non-North American loads), set the **ppCfgPortLoopCurrent**.*line* object to **1**, where *line* is the line number to set (beginning with **.1**).
2. To specify high loop current (the default for North American loads), set the **ppCfgPortLoopCurrent**.*line* object to **2**.

Note: High loop current reduces battery hold-up times.

Configuring Caller ID Options

Follow these steps to configure optional Caller ID behavior.

1. Set the **arrisMtaDevDefaultReasonNoCIDName** object to specify what data the eDVA sends to the CPE device when the CID signal request includes no Caller ID name. Set the MIB object either directly through a network manager, or in the eDVA configuration file. Possible values are:

Value	Sends	Description
unavailable(0)	\`O’	Caller ID displays typically show “Out of area” or “Unavailable.” This is the default for all country templates except Switzerland.
private(1)	\`P’	Caller ID displays typically show “Private.”
sendnothing(2)	’00’	Sends NULL data to the CPE device.
sdmf(3)	number	Sends the number in NA SDMF format.
excludeName(4)	(blank)	Omits name parameters or any reason for the missing name.

2. To set the delay time, in milliseconds, between receiving the ACK from the CPE and transmitting the FSK, set the **arrisMtaDevOffHookFskDelay** object to the desired time.

   Valid range: **0** to **500** milliseconds. Default: **261** for North America loads, **100** (T12 timer value) for Euro-PacketCable loads.

Setting Loop Voltage Management

Touchstone firmware supports Loop Voltage Management as defined in PacketCable specification PKT-SP-MIB-EXSIG-I03-070412. Loop Voltage Management provides four management policies used to control loop voltage behavior during outages or cable cuts.

The Advanced "Product Details" web page shows the current loop voltage management settings under the "Optional Features" heading. If Policy 3 is set, the web page displays the reset timer setting.

Note: Touchstone firmware supports both PacketCable and ARRIS-proprietary MIB objects for provisioning Loop Voltage Management. See "see "*Mapping ARRIS Loop Voltage Objects to PacketCable Objects* (page 61)" for details.

Loop Voltage Management Policies

Use the **pktcEnNcsEndPntLVMgmtPolicy** to set the Loop Voltage Management policy. The following policies are supported:

- **`voltageAtAllTimes`**(1)
- **`voltageUnlessRFQAMabsent`**(2)
- **`voltageBasedOnServiceOrTimers`**(3)
- **`voltageBasedOnService`**(4) (default)

Policy 1: Constant Loop Voltage

When this option is selected, the eDVA maintains loop voltage at all times with the following two exceptions:

- During firmware initialization of the line card. Touchstone eDVAs remove loop voltage for up to 1 second during firmware initialization, although typically this time is shorter.
- When the unit has no power.

Policy 2: QAM Carrier Detect

When this option is selected, the eDVA maintains loop voltage when it can lock onto a QAM carrier, including digital video QAM carriers. The assumption is that if the Telephony Modem can recognize a carrier, the connection is intact (has not been cut by a burglar).

When the Telephony Modem loses its carrier, after the T4 timeout expires (20–30 seconds), the modem scans cached and preset frequencies, then scans the entire spectrum. If the cable is truly cut, the scan takes 1 to 2 minutes to complete. If the modem lost its carrier, but can detect other RF energy (such as analog carriers), the scan can take up to 30 minutes to complete. If the modem does not detect any QAM carriers after scanning the STD and LRC frequencies, it removes loop voltage then continues with slow scanning.

Once the Telephony Modem removes loop voltage, it does not re-apply loop voltage until it re-registers with eDVA provisioning.

Policy 3: eDVA In-Service/Manual Reset

When this option is selected, the Telephony Modem maintains loop voltage when in-service or during manually-initiated resets and T4 timeouts. The Telephony Modem is considered in-service when eDVA TFTP is complete. Both subscriber resets (pushing the Reset button) and headend-initiated resets (SNMP, firmware upgrade) are considered manually-initiated.

Two timers govern the behavior of the eDVA during resets or outages:

pktcEnNcsEndPntLVMgmtResetTimer
The reset timer determines how long the eDVA maintains loop voltage during a reset. The default value is **3** for .TW loads, and **5** in all other loads.

Note: If the reset timer is set to a period longer than the scanning time between resets (typically 7 minutes), the eDVA never drops loop voltage.

pktcEnNcsEndPntLVMgmtMaintTimer
The plant maintenance timer (PMT) determines how long the eDVA maintains loop voltage when the E-UE loses its downstream signal. This timer may be used to maintain loop voltage during extended plant maintenance intervals.

Note: The value returns to the default setting, following a Telephony Modem reset.

When either timer expires without the eDVA coming in-service, the eDVA drops loop voltage.

Policy 4: eDVA In-Service

This is the default policy. Using this option, the eDVA goes through the following steps:

1. When applying initial AC power, Telephony Modems do not apply loop voltage.
2. When the eDVA completes TFTP and the lines are provisioned, it applies loop voltage only if the eDVA completes provisioning and the **pktcMtaDevProvisioningState** MIB object has one of the following values:
   - `pass`(1)
   - `passWithWarnings`(4)
   - `passWithIncompleteParsing`(5)

   If provisioning succeeds, the eDVA applies loop voltage to provisioned lines. If a provisioned line goes off-hook before the eDVA has contacted the call server, the modem immediately attempts to contact the call server and allows the call to continue. Loop voltage is applied even if the call server cannot be reached.
3. After the eDVA device is in service and there is an interruption to the RF, loop voltage remains present on the lines until a T4 timeout occurs (generally 20 to 30 seconds).

Loop Voltage Management MIB Objects

The **pktcEnNcsEndPntLVMgmtTable** table contains the loop voltage policy and timers. This table is indexed by **ifIndex**; an index value of **`1`** applies the policy settings to all eDVA lines.

The objects in the **pktcEnNcsEndPntLVMgmtTable** are:

pktcEnNcsEndPntLVMgmtPolicy

Defines the policy; one of:

- **`voltageAtAllTimes`**(1)
- **`voltageUnlessRFQAMabsent`**(2)
- **`voltageBasedOnServiceOrTimers`**(3)
- **`voltageBasedOnService`**(4)

See see "*Loop Voltage Management Policies* (page 59) for descriptions of each policy.

pktcEnNcsEndPntLVMgmtResetTimer

The time, in minutes, allowed for an eDVA to successfully provision after a reset. This timer applies only when **pktcEnNcsEndPntLVMgmtPolicy** is set to a value of **`voltageBasedOnServiceOrTimers`**(3). In all other cases, reading this object returns a value of zero.

The eDVA starts the timer upon a hard reboot, a soft reset or a T4 timeout. The timer value persists the last configured value (i.e., not the countdown value) of this MIB Object across hard reboots and soft resets.

Valid range: **`0`** to **`1440`** (minutes). Default: **`5`**.

pktcEnNcsEndPntLVMgmtMaintTimer

The time, in minutes, that the eDVA maintains loop voltage regardless of the eDVA's connection or provisioning status. This timer applies only when **pktcEnNcsEndPntLVMgmtPolicy** is set to a value of **`voltageBasedOnServiceOrTimers`**(3). In all other cases, reading this object returns a value of zero. The current timer value persists across soft resets, but resets to zero for a hard reset or power-cycle.

The eDVA starts the timer when it is set to a value greater than zero. The eDVA maintains loop voltage until the timer expires.

Valid range: **`0`** to **`1440`** (minutes). Default: **`0`**.

Mapping ARRIS Loop Voltage Objects to PacketCable Objects

Some earlier versions of Touchstone firmware provided an ARRIS-proprietary version of Loop Voltage Management that is very similar to the PacketCable-standard version. For backward compatibility, TS11.1 supports both ARRIS and PacketCable MIB objects.

The following table defines how ARRIS objects map to PacketCable objects.

ARRIS Object	PacketCable Object	Notes
arrisMtaDevLoopVoltageKey	(none)	Not needed
arrisMtaDevLoopVoltagePolicy `always-voltage-present`(1) `rf-carrier-voltage-present`(2) `in-service-voltage-present`(3) `default-operation`(4)	**pktcEnNcsEndPntLVMgmtPolicy** `voltageAtAllTimes`(1) `voltageUnlessRFQAMAbsent`(2) `voltageBasedOnServiceOrTimers`(3) `voltageBasedOnService`(4)	Roughly equivalent
arrisMtaDevLoopVoltageResetTimeout ▪ Valid range: 8–1800	**pktcEnNcsEndPntLVMgmtResetTimer** ▪ Valid range: 0–1400	Adjust ranges as shown
arrisMtaDevLoopVoltageMaintTimeout ▪ Valid range: integer	**pktcEnNcsEndPntLVMgmtMaintTimer** ▪ Valid range: 0–1440	Adjust ranges as shown

Action

Follow these steps to configure loop voltage management.

1. Set the **pktcEnNcsEndPntLVMgmtPolicy** object to the appropriate policy value:
   - `voltageAtAllTimes`(1)
   - `voltageUnlessRFQAMabsent`(2)
   - `voltageBasedOnServiceOrTimers`(3)
   - `voltageBasedOnService`(4) (default)

Note: For this and other LVM objects, use an index of `.1` to apply the same policy to all lines. If you want to specify different policies for each line, use the **ifIndex** of each line instead.

2. For eDVAs using policy 3, modify the reset timer (if necessary) by setting the **pktcEnNcsEndPntLVMgmtResetTimer** object. Valid range: `0` to `1440` minutes. Default: `5` minutes.

   The line card drops loop voltage after a CM reset, if the eDVA has not been successfully provisioned before the Reset Timeout timer expires.

Note: This object is only used with policy option 3 and is ignored if a policy setting of other than 3 is used.

3. For eDVAs using policy 3, modify the plant maintenance timer (if necessary) by setting the **pktcEnNcsEndPntLVMgmtMaintTimer** object. Valid range: `0` to `1440` minutes (24 hours).
4. Reset the eDVA to enable the new loop voltage policy on the eDVA.

Provisioning HD Voice

Use this procedure to provision G.722 HD (wideband) voice capabilities.

Support Overview

- Touchstone firmware supports HD voice with IPv6 eDVA addressing on Telephony Modems and Telephony Gateways.

Supported Country Templates

Touchstone firmware supports HD audio only with North American country templates:

- `northAmerica57`(1)
- `northAmerica33`(17)
- `northAmerica09`(18)
- `northAmerica66`(32)

Supported HD Audio CODECs

Touchstone firmware supports the following G.722 CODECs:

- G722-48
- G722-56
- G722-64 (G722)

The CODEC name "G722" is equivalent to G722-64.

Note 1: TS11.1 does not support the G722.2 Adaptive Multi Rate (AMR-WB) CODEC.

Note 2: G.722 has no associated Packet Loss Concealment or VAD mechanisms.

Touchstone firmware can advertise either static or dynamic payload types in the SDP, depending on the setting of the **arrisMtaDevHDAudioDefaultPayloadType** object.

TDD Tone Processing Support

TDD tone processing is enabled by default. When the eDVA detects the TDD tone on a connection, it switches to G.711 (PCMU) for the duration of the call.

To disable TDD tone processing, set bit `0x00000002` of the **ppCfgMtaCallpFeatureSwitch2** object.

Considerations

Keep the following in mind before enabling HD voice:

- Changing a modem from standard to HD audio, or from HD to standard audio, requires a reboot.

- Touchstone hardware supports up to four simultaneous G.722 calls. This applies to both 2-line and 4-line models.
- G.722 takes effect only when both ends of a call advertise G.722 support, and when HD audio-capable CPE devices are installed at both ends.
- Line-level MIB objects may be post-provisioned without rebooting the eDVA. Changes take effect with the next established call.

To establish an HD audio call, the following criteria must be met:

- The following objects must be enabled:
  - **arrisMtaDevWBSLIC** (reboot required after changing this value)
  - **arrisMtaDevHDAudioEnable**
  - **arrisMtaDevHDAudioEndPntEnable** (on the lines requiring HD audio)
- The appropriate CODEC array object, corresponding to the signaling type, must contain the string "G722" as shown in the following table:

Signaling Type	CODEC Array Object
NCS	**arrisMtaDevProvisionedCodecArray**
SIP (ARRIS)	**sipCfgProvisionedCodecArray**
SIP (PC20)	**pktcEUERSTBCallPrefCodecList**

The far-end must include G722 in its SDP.

Action

Perform the following tasks as needed.

Provisioning HD Voice using SNMP

Follow these steps to provision HD Voice support in an SNMP manager or configuration file.

1. Set the **arrisMtaDevWBSLIC** object to `enable`(1) to enable hardware support.

   This setting is persistent.

Note: If modified after provisioning is complete, changes to this object do not take effect until after the modem reboots.

2. Set the **arrisMtaDevHDAudioEnable** object to `enable`(1) to allow negotiation of the G.722 CODEC.

   This setting is not persistent.

3. For each line as necessary, set the **arrisMtaDevEndPntHDAudioEnable** object to `enable`(1). This object is indexed by the line number, starting with `1`.

   Setting this object is persistent only if done using an SNMP manager after provisioning is complete.

4. For SIP loads, add “G722” to the CODEC array object corresponding to the signaling type in use:

Signaling Type	CODEC Array Object
NCS	**arrisMtaDevProvisionedCodecArray**
SIP (ARRIS)	**sipCfgProvisionedCodecArray**
SIP (PC20)	**pktcEUERSTBCallPrefCodecList**

   Set the index for each option as required to enable the desired lines.

5. (optional) To specify advertising G.722 using a dynamic payload type, set the **arrisMtaDevHDAudioDefaultPayloadType** object to `dynamic`(1). The default is `static`(0).

6. (optional) To disable TDD tone processing, set bit `0x00000002` of the **ppCfgMtaCallpFeatureSwitch2** object.

Note: Setting this bit disables TDD tone processing regardless of the CODEC in use.

Displaying HD Voice Status

HD voice status can be displayed from the web-based interface, the CLI, or SNMP. Follow these steps to display the status from any of these interfaces.

1. To display HD voice status using the web-based interface, open the Status page. The bottom section, labeled **HD Audio**, shows the status for each line. See Status Screen for more information about the Status page.
2. To display HD voice status using an SNMP browser or manager, read the **arrisMtaDevEndPntHDAudioStatus** object for each line (use the line number as the index, starting with 1). The value is `enabled`(1) for each line that has HD voice enabled.

Echo Cancellation and Analog Fax/Modem Support

Touchstone eDVAs support:

- echo cancellation per ITU G.168/G, with a 28 dB echo return loss
- Group I-III compliant facsimile devices
- analog voice band modems up to v.92

Echo cancellation provides a 32 ms echo cancellation tail.

Echo cancellation operates in one of three modes:

- enabled
- enabled with non-linear processor (NLP) disabled
- disabled

The echo cancellation feature is always enabled for normal voice calls. However, for analog fax/modem calls, the ARRIS eDVA sets the echo cancellation mode depending on detected tones:

Tones Detected	Mode
Calling (CNG), V.21 preamble, or “slow” speed CED (14.4k or lower)	NLP disabled
“high” speed CED (28.8k and higher)	completely disabled

If the eDVA detects fax/modem tones, and a CODEC other than G.711 is active, the eDVA automatically switches to the G.711 CODEC if G.711 was negotiated as a backup CODEC when the call was set up. Upon completion of a fax call, the eDVA automatically re-enables echo cancellation, but does not switch back to the original CODEC unless instructed to switch by the Call Agent. After a modem call completes, the eDVA re-enables echo cancellation when instructed by the Call Agent.

Adaptive Jitter Buffers

Touchstone firmware supports jitter buffers for fax/modem calls. These buffers automatically adapt their size to accommodate the required jitter. See *Configuring Jitter Buffers* (page 77) for configuring jitter buffer sizes.

Configuring the Echo Cancellation Tail Length

The default echo cancellation tail length is 32 ms. Follow these steps to configure the desired echo cancellation tail length.

1. In the eDVA configuration file, add the **arrisMtaDevEchoCancellerTailLength** object and set it to **`eightMs`** or **`thirtyTwoMs`** (default) as desired.
2. Reset the eDVA.

   The eDVA sets the echo cancellation tail as configured, then marks the MIB read-only.

Provisioning RFC 2833 Support

RFC 2833 (also called DTMF Relay) specifies a method for carrying DTMF and other telephony signals and events in RTP packets, instead of sending audio tones over the network. This functionality is especially important when using highly-compressed CODECs such as G.729, which may distort DTMF tones.

The eDVA signals RFC 2833 support by specifying “telephone-event” in its list of available CODECs during negotiation. By default, the CMS instructs the eDVA to enable or disable RFC 2833 and selects the payload type to use. Touchstone firmware provides feature switches to override the CMS and enable RFC 2833 support with a specific payload type.

Controlling RFC 2833 Functionality

Touchstone firmware can enable RFC 2833 functionality regardless of whether the CMS instructs the eDVA to use it. Two MIB objects control the functionality:

ppCfgMtaCallpFeatureSwitch

The following CallP Feature Switch bits control RFC 2833 functionality:

Bit	Description
0x00001000	Nuera RFC 2833 messaging without request using payload 127 (NCS only) Set this bit to instruct the eDVA to generate RFC 2833 events with a payload type of 127 without specifically being instructed to do so (for compatibility with the Nuera RDT).
0x00080000	LUCENT RFC 2833 messaging without request using payload 94 (NCS only) Set this bit to instruct the eDVA to generate RFC 2833 events with a payload type of 94 during call setup, without specifically being instructed to do so.
0x01000000	Send DTMF digits via RFC 2833 with operator-specified payload without request (NCS only) Set this bit to instruct the eDVA to generate RFC 2833 DTMF events using a payload type defined in the **ppCfgRfc2833DigitPayloadType** MIB object without being instructed to do so by the CMS. The default payload type value is 101.

ppCfgRfc2833DigitPayloadType

When bit 0x01000000 is enabled in the CallP Feature Switch, the eDVA sends RFC 2833 events with the payload type specified by this object.

Valid range: **97** to **127**. Default: **101**.

Configuring T.38 Fax Relay Support

Touchstone firmware supports T.38 fax relay, version 0. T.38 fax relay provides higher reliability of fax transmissions using redundancy to tolerate packet loss. Touchstone firmware supports call agent-controlled T.38 as defined in PKT-SP-NCS1.5-I03-070412, Appendix A.

T.38 support requires that SDP capability reporting be enabled (the default setting).

SDP Parameter List for T.38 Strict

When T.38 Strict mode is enabled, the eDVA sends an SDP list as shown below. The bolded portion indicates the capability descriptor.

```
v=0
o=- 48186 48188 IN IP4 10.1.36.218
s=-
c=IN IP4 10.1.36.218
t=0 0
m=audio 61304 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=ptime:20
a=sqn:0
a=cdsc:1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 101
a=cpar:a=rtpmap:96 G729E/8000/1
```

```
a=cpar:a=rtpmap:97 G726-16/8000/1
a=cpar:a=rtpmap:98 G726-24/8000/1
a=cpar:a=rtpmap:2 G726-32/8000/1
a=cpar:a=rtpmap:99 G726-40/8000/1
a=cdsc:11 image udptl t38
a=cpar:a=T38FaxVersion:0
a=cpar:a=T38FaxRateManagement:transferredTCF
a=cpar:a=T38FaxMaxDatagram:160
a=cpar:a=T38FaxUdpEC:t38UDPRedundancy
a=cpar:a=T38MaxBitRate:14400
m=image 0 udptl t38
```

If the capability descriptor causes interoperability issues, set bit 0x00000008 of the CallP Feature Switch. See *CallP Feature Switch* (page 46) for more information about the feature switch.

CallP Feature Switches Affecting the SDP

The following feature switches affect the SDP, returned in response to a Create (CRCX) or Modify (MDCX) Connection command.

Bit	Description
0x00000008	Reduce the capability descriptor in the SDP to T38 only (default = 0, no reduction).
0x00000080	Omit mptime parameter in returned SDP (default = 0, mptime included).
0x00001000	NUERA RFC 2833 messaging without request using payload 127 (default = 0, telephone-event is negotiated normally).
0x00080000	LUCENT RFC 2833 messaging without request using payload 94 (default = 0, telephone-event is negotiated normally).
0x00100000	Allow AES encryption for RTP/RTCP (default = 1, AES encryption is negotiated normally).
0x01000000	Send DTMF digits via RFC 2833 with an operator-defined payload without request (default = 0, telephone-event is negotiated normally).

The following CRCX message is used to generate all the SDP examples, unless otherwise specified:

```
CRCX 19901 aaln/1@mta218.dev36 MGCP 1.0 NCS 1.0
C: 1234
M: recvonly
L: mp:20, a:PCMU, fxr/fx:t38-loose, xrm/mcr:on
```

The default feature switch settings are (**`0x0`** for NCS loads, and **`0x0020`** for SIP loads) generate the following SDP:

```
v=0
o=- 381749076 381749076 IN IP4 10.1.36.219
s=-
```

```
c=IN IP4 10.1.36.219
t=0 0
m=audio 65496 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=mptime:20
a=rtpmap:0 PCMU/8000/1
a=X-pc-secret:base64:ZNos/qs530eSDJ4FvdL2GJBR62lS5UKyQ7n9og4
IaaDbA9Blpg6lM2PfOaHEGg== U5Q5N/eWnimq9Q/yjWwY2hACRIY6a9qqEQ
Us8tm54lEmEE6LXkCB51+3sqxlQg==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 101
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=rtpmap:101 telephone-event/8000/1
a=cpar: a=fmtp:101 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

When the “Reduce Capability Descriptor” switch (**0x00000008**) is enabled, and all other switches are set to their default values, the SDP becomes:

```
v=0
o=- 381749076 381749076 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 65496 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=mptime:20
a=rtpmap:0 PCMU/8000/1
a=X-pc-secret:base64:ZNos/qs530eSDJ4FvdL2GJBR62lS5UKyQ7n9og4
IaaDbA9Blpg6lM2PfOaHEGg== U5Q5N/eWnimq9Q/yjWwY2hACRIY6a9qqEQ
Us8tm54lEmEE6LXkCB51+3sqxlQg==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
a=sqn: 0
a=cdsc: 1 image udptl t38
```

In this example, the Capability Descriptor is reduced to only the information required to relay support for UDPTL T.38 to the far end. This allows the far end to support T.38 strict mode as defined in the PacketCable 1.5 NCS specification.

When the “Omit mptime” switch (**0x00000080**) is enabled, and all other switches are set to their default values, the SDP becomes:

```
v=0
o=- 382093395 382093395 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 58810 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=rtpmap:0 PCMU/8000/1
a=X-pc-secret:base64:XI51bgXR5MNUdaKXisS0tjYCc90x3f7jA+ojyam
W/O/M2BlCaejlrRLOdApR6w== 6LbN8ULCFGMjcR2T3l1uZuBcfWM2vfGn09
YTmR6OhHfQwC4eE+WWSX7AarnFPA==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
```

```
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 101
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=rtpmap:101 telephone-event/8000/1
a=cpar: a=fmtp:101 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

In this example, neither the `a=ptime` nor the `a=mptime` parameters are included in the SDP since the packetization rate is the default (20 ms). If the Call Agent had specified a different packetization rate, then the `a=ptime` parameter is included as follows:

```
v=0
o=- 382093395 382093395 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 58810 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=ptime:10
a=rtpmap:0 PCMU/8000/1
a=X-pc-secret:base64:XI51bgXR5MNUdaKXisS0tjYCc90x3f7jA+ojyam
W/O/M2BlCaejlrRLOdApR6w== 6LbN8ULCFGMjcR2T3l1uZuBcfWM2vfGn09
YTmR6OhHfQwC4eE+WWSX7AarnFPA==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 101
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=rtpmap:101 telephone-event/8000/1
a=cpar: a=fmtp:101 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

When the “Allow AES Encryption” switch (**`0x00100000`**) is disabled, and all other switches are set to their default values, the SDP becomes;

```
v=0
o=- 381749076 381749076 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 65496 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=mptime:20
a=rtpmap:0 PCMU/8000/1
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 101
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=rtpmap:101 telephone-event/8000/1
a=cpar: a=fmtp:101 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
```

```
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

The last three CPFS bits (**0x00001000**, **0x00080000**, and **0x01000000**) are related to RFC 2833. They are used in specific configurations and are designed to skip CODEC negotiation. For example, when the “Nuera RFC2833” feature is enabled, and all other switches are set to their default values, the SDP becomes:

```
v=0
o=- 382250430 382250430 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 63672 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=mptime:20
a=rtpmap:0 PCMU/8000/1
a=X-pc-secret:base64:Qb8GFLNXP4Z3yiyxFx1Ws9vLph9qG6bTIXezUlz
rwia7iiNvPPkVYdZhZ77NEQ== zZjgwXRR2j5F04lDXefPTV06PT8g31Hn5V
Ea6NJvFPFsPiraDeDI35EI8KO+4A==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 127
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=rtpmap:127 telephone-event/8000/1
a=cpar: a=fmtp:127 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

Note that this feature does not add SDP attributes, but modifies the Capability Descriptor slightly. Payload type 127 is used for RFC 2833 support.

When the “Lucent RFC2833” switch is enabled, and all other switches are set to their default values, the SDP becomes:

```
v=0
o=- 382318343 382318343 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 49688 RTP/AVP 0
b=AS:81
a=rtcp-xr:voip-metrics
a=mptime:20
a=rtpmap:0 PCMU/8000/1
a=X-pc-secret:base64:Kue6n+ZSGpXrB2iAJIAUQNst6AtAS7Ad7zC3oGP
ry9XKdURiy4Y6iLaDEhk5lg== GgaMtuqF7/egjksBpQ8SZeWnXCAlr1EeNH
AHV7EY0fv03Y0MYAYa1zz/lvO5dg==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 94
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=rtpmap:94 telephone-event/8000/1
a=cpar: a=fmtp:94 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
```

```
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

This feature does not add SDP attributes, but modifies the Capability Descriptor. In this case, payload type 94 is used for RFC 2833 support.

Note: The three feature switches that affect RFC 2833 negotiation are mutually exclusive. At most, only one of the bits may be set in the CallP Feature Switch. Enabling multiple RFC 2833 features may result in unexpected behavior.

When the "RFC2833 Digits" switch (**`0x01000000`**) is enabled, and all other switches are set to their default values, the SDP becomes:

```
v=0
o=- 382360201 382360201 IN IP4 10.1.36.219
s=-
c=IN IP4 10.1.36.219
t=0 0
m=audio 53560 RTP/AVP 0 101
b=AS:81
a=rtcp-xr:voip-metrics
a=mptime:20 -
a=rtpmap:0 PCMU/8000/1
a=rtpmap:101 telephone-event/8000/1
a=fmtp:101 0-15
a=X-pc-secret:base64:rwZISK4HN5wlZzehiOBSEJXsRQbexmiwm1Ou4pE
nXFr4lSTXQYdAsKFT5lhkkw== tWyPwAvBg6EbSs4+FoY7rWOn0l8pcQPxGm
iwlNGPfo3Suehu0CncQ2egC4JQ6w==
a=X-pc-csuites-rtp:62/51 64/51 60/51 60/50
a=X-pc-csuites-rtcp:81/70 81/71 82/70 82/71 80/70
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 8 15 18 96 97 98 2 99 101
a=cpar: a=fmtp:18 annexb=no
a=cpar: a=fmtp:101 0-15,144,149,159
a=cdsc: 11 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxRateManagement:transferredTCF
a=cpar: a=T38FaxMaxDatagram:161
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
```

Note that this feature affects the SDP, including the Capability Descriptor. the payload type specified by **ppCfgRfc2833DigitPayloadType** is added to the m= line (the default value is "101") and two new attributes indicate that the endpoint is prepared to receive RFC 2833 digits.

PacketCable 1.5 Extended Signaling

Touchstone firmware supports the PacketCable 1.5 NCS Extended Signaling MIB fax detection objects for T.38 signaling. The following objects are supported:

pktcEnNcsMinimumDtmfPlayout
: The minimum time a digit is played when the eDVA receives an RFC 2833 digit event.

pktcEnNcsEndPntQuarantineState
: The endpoint state, based on the NCS quarantine states (for example, notification or lockstep).

pktcEnNcsEndPntHookState

The hook state (on-hook or off-hook) of the endpoint.

pktcEnNcsEndPntFaxDetection

(NCS only) Configures whether the eDVA uses T.38 when detecting a CNG (calling) tone. The default is disabled, which prevents using T.38.

SDP Parameter List Considerations

The complete list of Call Processing features (including T.38 Fax Relay) requires a Session Description Protocol (SDP) parameter list longer than 512 bytes. Some CMSs or P-CSCFs do not support a parameter list longer than 512 bytes. To reduce the SDP parameter list size, set the “Suppress SDP Capability Attribute Parameters” CallP Feature Switch (bit **`0x00000100`**) to **`1`**.

Setting this bit disables T.38 strict mode functionality; T.38 loose mode is still available.

For more SDP-related options, see the CallP Feature Switch details in *CallP Feature Switch* (page 46).

T.38 Provisioning Overview

The following MIB objects control T.38 behavior.

sipCfgPortT38Mode

(SIP only) Sets the T.38 operating mode. It allows the following values:

Value	Description
`t38Off`(1)	(default) Disables T.38.
`t38Loose`(2)	Enables T.38 Loose mode. In loose mode, the eDVA can use T.38 for fax transmission whether or not the remote endpoint has indicated T.38 support.
`t38Strict`(3)	Enables T.38 Strict mode. In strict mode, the eDVA can use T.38 for fax transmission only if the far end indicated support for T.38 during session negotiation.

arrisMtaDevEndPntFaxOnlyLineTimeout

(NCS only) Configures fax-only mode for a line. Specifies the time, in seconds, to wait for fax or modem tones after receiving the SDP from the remote endpoint. If the time expires before detecting fax or modem tones, the eDVA drops the call. Valid range: **`0`** to **`600`**.

sipCfgPortFaxOnlyTimeout

(SIP only) Configures fax-only mode for a line. Specifies the time, in seconds, to wait for fax or modem tones after receiving the SDP from the remote endpoint. If the time expires before detecting fax or modem tones, the eDVA drops the call. Valid range: **`0`** to **`600`**.

sipCfgPortMaxT38HSRedLevel

(SIP only) Sets the maximum high-speed redundancy level used for T.38 fax relay, in both send and receive directions. The value of this object is the number of older data packets included in each T.38 datagram when transferring fax data. The actual redundancy level used is determined by negotiation with the remote endpoint. Valid range: **0** to **2**. Default: **1**.

arrisMtaDevT38Timeout

(NCS only) The time, in seconds, that the eDVA waits for the Call Agent to modify the connection to T.38. The far end receives silence until the Call Agent modifies the connection.

Valid range: **1** to **30** seconds. Default: **10**.

Action

Perform the following tasks as needed.

Controlling T.38 and Fax-Only Modes

Follow these steps to configure T.38 fax detection and fax-only mode.

1. Set the **pktcEnNcsEndPntFaxDetection** object (use the line number as an index, starting with 1) to configure whether the eDVA detects CNG (calling) tones and starts T.38 mode.

   The default value, **`false`**(2), disables detection of CNG tones on the endpoint. Disabling CNG detection prevents the eDVA from using T.38 mode, which may be desired when the fax machines are capable of Super G3 (which uses a higher transmission rate).

Note: If you make this change using an SNMP browser, the new setting takes effect on the next connection.

2. If needed, set the **pktcEnNcsMinimumDtmfPlayout** object to specify the minimum time, in milliseconds, that the eDVA plays out a digit tone when receiving an RFC 2833 digit event on the specified endpoint. The actual play-out time is the maximum of this setting and the time specified in the RFC 2833 packet.

   Valid range: **`40`** to **`100`**, or **`0`** (the default), which always uses the time specified in the RFC 2833 packet.
3. Set fax-only mode by adding the **arrisMtaDevEndPntFaxOnlyLineTimeout** MIB object to the eDVA configuration file. This MIB object is specific to a line, so it must be specified with the line number; for example, **arrisMtaDevEndPntFaxOnlyLineTimeout**.1 for line 1. The value specifies the timeout, in seconds, after which the eDVA drops the call if it does not detect fax or modem tones.

   Valid range (either object): **`0`** (disabled) to **`600`** seconds. The default is **`0`**.

Configuring T.38 MaxDatagram Size

Follow these steps to configure the MaxDatagram size for T.38.

1. In the eDVA configuration file, set the **ppCfgPortT38MaxDatagram** object to the desired datagram size.

   Valid range: **`160`** to **`65535`** octets. Default: **`160`**.
2. Reset the eDVA to have the new datagram size take effect.

Super G3 FAX Support

Touchstone firmware supports SuperG3 FAX transmission, including V.8 data exchange detection.

When the eDVA detects the V.8 signal, fax transmission proceeds depending on the setting of the **arrisMtaDevSuperG3FaxRelay** MIB object:

- When set to **`disable`**(0) (the default), the eDVA uses the V.8 detection as a trigger to set up the endpoint for FAX transmission (adjust the jitter buffer, turn off echo cancellation) and allow the FAX machines to handle the negotiation and transmit the fax using G.711.
- When set to **`enable`**(1), the eDVA uses the V.8 detection as a trigger to start T.38.

  To switch the FAX call from SuperG3 to T.38, the Telephony Modem mutes audio to prevent the FAX machine from negotiating to SuperG3 before switching to T.38. Depending on the T.38 negotiation, this switch either forces the FAX machines to downshift to G3 speeds (<= 14.4) which is the current maximum rate supported for T.38 version 0, or initiate the T.38 over RTP.

  If the connection is negotiated for T.38 version 0, the audio remains muted until the CMS transitions the connection to T.38. During the muted period, the terminating fax machine receives silence. The maximum amount of this silence period can be controlled by the **arrisMtaDevT38Timeout** MIB object. This object defaults to 30 seconds, which is consistent with existing behavior. Alter this value only if FAX transmissions are consistently failing due T.38 not re-trying.

Configuring Jitter Buffers

Touchstone firmware provides three sets of MIB objects to adjust the eDVA jitter buffer:

- Standard—used for typical voice calls. Associated MIB objects are:
  - The **arrisMtaDevVPJitterBufferMode** object controls the jitter buffer behavior (adaptive or fixed modes).
  - The **arrisMtaDevVPNomJitterBuffer** object sets the nominal jitter buffer size (in packets).
  - The **arrisMtaDevVPMaxJitterBuffer** object sets the maximum jitter buffer size (in packets).
- Voice Band Data (VBD)—used for calls involving data transfer, including fax, modem, and POS terminals. The following MIB objects allow the eDVA to automatically override the normal jitter buffer settings when it detects a Voice Band Data call:
  - The **arrisMtaDevVbdOverwriteLineBitmap** object allows per-line control of jitter buffer override.
  - The **arrisMtaDevVbdOverwriteMinJitterBuffer** object specifies the minimum Voice Band Data call jitter buffer setting.
  - The **arrisMtaDevVbdOverwriteNomJitterBuffer** object specifies the nominal Voice Band Data call jitter buffer setting.

   - The **arrisMtaDevVbdOverwriteMaxJitterBuffer** object specifies the maximum Voice Band Data call jitter buffer setting.

  If you change these parameters with an SNMP manager, the new settings take effect starting with the next phone call.

- Custom—provides more precise control over jitter buffer settings. Custom settings may be useful under certain network conditions or applications.
   - The **arrisMtaDevCustomJitterBufferEnabled** object enables customer jitter buffer settings.
   - The **arrisMtaDevCustomMinJitterBuffer** object controls the minimum custom jitter buffer duration.
   - The **arrisMtaDevCustomNomJitterBuffer** object controls the nominal custom jitter buffer duration.
   - The **arrisMtaDevCustomMaxJitterBuffer** object controls the maximum custom jitter buffer duration.

If you change these MIB objects with an SNMP manager, the new settings take effect starting with the next phone call.

Action

Perform the following tasks as needed.

Setting Standard Jitter Buffer Parameters

Follow these steps to set standard jitter buffer parameters.

1. Set the **arrisMtaDevVPJitterBufferMode** object to either **1** (adaptive, the default) or **2** (fixed).
2. Configure the nominal voice call jitter buffer size by setting the **arrisMtaDevVPNomJitterBuffer** object. The value represents a multiple of the packetization rate. Valid range: **1** to **4**. Default: **1**.
3. Configure the maximum voice call jitter buffer size by setting the **arrisMtaDevVPMaxJitterBuffer** object. The value represents a multiple of the packetization rate. Valid range: **1** to **4**. Default: **3**.

Note: The nominal jitter buffer setting must be less than the maximum jitter buffer setting.

Setting Voice Band Data Jitter Buffer Parameters

Follow these steps to enable and configure jitter buffer and override settings for Voice Band Data (fax, modem, POS terminal) calls.

1. Set the **arrisMtaDevVbdOverwriteLineBitmap** to control jitter buffer override on each line. The least significant bit controls line 1, so a value of **3** enables override on lines 1 and 2.
2. Configure the minimum, nominal, and maximum jitter buffer settings for Voice Band Data calls by setting the following objects. The valid range for all three objects is **10** to **135**; the following table shows the default value for each object.

Object	Default Value	Description
arrisMtaDevVbdOverwriteMinJitterBuffer	25	Minimum fax/modem call jitter buffer setting.
arrisMtaDevVbdOverwriteNomJitterBuffer	25	Nominal fax/modem call jitter buffer setting.
arrisMtaDevVbdOverwriteMaxJitterBuffer	135	Maximum fax/modem call jitter buffer setting.

Note: The minimum jitter buffer setting must be less than the nominal setting, which in turn must be less than the maximum jitter buffer setting.

Configuring Custom Jitter Buffer Settings

Follow these steps to configure custom jitter buffer settings.

Note: Custom jitter buffer settings use units of milliseconds, rather than packet multiples used by the standard and VBD methods.

1. Enable custom jitter buffer settings by setting the **arrisMtaDevCustomJitterBufferEnabled** object to **on**(1). The default is **off**(0).

   When custom jitter buffer settings are enabled, the default settings are:

   - Minimum: 5 milliseconds
   - Nominal: 10 milliseconds
   - Maximum: 60 milliseconds

2. Configure the minimum, nominal, and maximum custom jitter buffer settings by setting the following objects. The valid range for all three objects is **5** to **135** (milliseconds).

   arrisMtaDevCustomMinJitterBuffer
   Minimum custom jitter buffer duration.

   arrisMtaDevCustomNomJitterBuffer
   Nominal custom jitter buffer duration.

arrisMtaDevCustomMaxJitterBuffer

Maximum custom jitter buffer duration.

Provisioning European Support

The TS11.1 .EURO load provides support for full Euro-PacketCable provisioning. This requires that new features be configured and existing MIBs be modified.

Action

Perform the following tasks as needed.

Configuring Power Ring Frequency

Follow these steps to configure the power ring frequency.

1. Select the **pktcSigPowerRingFrequency** object and input the appropriate frequency for your network. The following frequencies are valid:
   - 20 Hz (1)
   - 25 Hz (2) (default for Belgium, Germany, Netherlands, Netherlands09, Poland, Spain, Norway, and Switzerland)
   - 33.33 Hz (3)
   - 50 Hz (4) (default for Austria, France, and Hungary)
   - 15 Hz (5)
   - 16 Hz (6)
   - 22 Hz (7)
   - 23 Hz (8)
   - 45 Hz (9)

Configuring On-Hook Caller ID

Follow these steps to configure the on-hook Caller ID functionality.

1. Set the protocol by modifying the **pktcSigDevCallerIdSigProtocol** object. Supported protocols are `fsk`(1) (default) and `dtmf`(2).
2. Select the **pktcSigDevCIDMode** object and select from one of the following modes:

- `duringRingingETS`—The FSK containing the Caller ID information is sent between the first and second ringing pattern. The delay between the end of the first ringing pattern and the start of the transmission for of the FSK containing the Caller ID information is configurable using the **pktcSigDevCIDFskAfterRing** object.
- `dtAsETS` (default for Austria, Belgium, France, Germany, Netherlands, Netherlands09, Norway, Spain, and Switzerland)—The FSK is sent after the Dual Tone Alert Signal but before the first ring pattern.
- `rpAsETS` (default for Hungary and Poland)—The FSK is sent after a Ring Pulse but before the first ring pattern. The delay between the end of the Ring Pulse Alert Signal and the start of transmission of the FSK containing the Caller ID information is configured with the **pktcSigDevCIDFskAfterRPAS** object.
- `lrAsETS–The` Line Reversal occurs first, then the Dual Tone Alert Signal, and finally the FSK is sent but before the first ring pattern. The delay between the end of the Line Reversal and the start of the Dual Tone Alert Signal is configurable via the **pktcSigDevCIDDTASAfterLR** object.

Note 1: The delay between the Dual Tone Alert Signal and the start of transmission of the FSK containing the Caller ID information for both the dtAsETS and lrAsETS Caller ID methods is configurable using the **pktcSigDevCIDFskAfterDTAS** object.

Note 2: The delay between the end of the complete transmission of the FSK containing the Caller ID information and the start of the first ring pattern for dtAsETS, rpAsETS, and lrAsETS caller ID methods is configurable via the **pktcSigDevCIDRingAfterFSK** object.

Configuring Visual Message Waiting Indicator

Follow these steps to configure visual message waiting indicator.

1. Select the **pktcSigDevVmwiMode** object and select from one of the following modes:
   - `dtAsETS` (default for Austria, Belgium, France, Germany, Netherlands, Netherlands09, Norway, Spain, and Switzerland)—The FSK is sent after the Dual Tone Alert Signal but before the first ring pattern.
   - `rpAsETS` (default for Hungary)—The FSK is sent after the Ring Pulse. Configure the delay between the end of the Ring Pulse Alert Signal and the start of the transmission of the FSK containing the VMWI information using the **pktcSigDevVmwiFskAfterRPAS** object.
   - `lrAsETS` (default for Poland)—The Line Reversal occurs first, then the Dual Tone Alert Signal, and finally the FSK is sent. Configure the delay between the end of the Line Reversal and the start of the Dual Tone Alert Signal for VMWI information using the **pktcSigDevVmwiDTASAfterLR** object.

Note: Configure the delay between the end of the Dual Tone Alert Signal and the start of the transmission of the FSK information containing the VMWI information for the dtAsETS and lrAsETS alerting signal methods using the **pktcSigDevVmwiFskAfterDTAS** object.

Configuring Tone Operations

Touchstone firmware allows configuration of call progress tones for North American and European templates. See *Configuring Call Progress Tones* (page 83) for instructions.

Configuring Hook Flash Timing

Touchstone firmware allows configuration of hook flash timing for North American and European templates. See *Configuring Hook Flash Timing* (page 90) for instructions.

Configuring Call Progress Tones

Use this procedure to configure call progress tones. Touchstone .EURO loads support provisioning of call progress tones through the **pktcSigDevToneTable**. North American loads support this table only when the country code setting (set by **ppCfgMtaCountryTemplate**) is one of the North American country codes.

MIB Tables

Touchstone firmware uses two MIB tables to define call progress tones. See *Default Tone Settings* (page 313) for a list of default tone definitions for each country code type.

pktcSigDevToneTable

Defines the tone type, repeat count, and whether the last tone should be held steady after completing the cadence. Each entry in the table contains the following objects:

- **pktcSigDevToneType**: the index for the table (see below).
- **pktcSigDevToneWholeToneRepeatCount**: the number of times to repeat the entire sequence.
- **pktcSigDevToneSteady**: set to `true`(1) to keep the last tone in the sequence on until reaching the timeout.

pktcSigDevMultiFreqToneTable

Defines the actual frequencies for each tone defined in the tone table. Each entry in this table contains the following objects:

- **pktcSigDevToneNumber**: A secondary index, indicating the sequence number of the defined tone. Up to eight tones may be defined for a tone type.
- **pktcSigDevToneFirstFrequency**, **pktcSigDevToneSecondFrequency**, **pktcSigDevToneThirdFrequency**, **pktcSigDevToneFourthFrequency**: Up to four frequencies per defined tone. To disable a frequency, set it to `0`.
- **pktcSigDevToneFreqMode**: Determines how the frequencies define the tone:

    - **`firstModulatedBySecond`**(1): The first frequency is modulated by the second frequency, according to the percentage specified by **pktcSigDevToneFreqAmpModePrtg**. The third and fourth frequencies are ignored.
    - **`summation`**(2): All specified frequencies are added together without adding modulation.
- **pktcSigDevToneFreqAmpModePrtg**: The percentage of amplitude modulation to apply when using the **`firstModulatedBySecond`** setting.
- **pktcSigDevToneDbLevel**: The decibel level for each tone. The default is −40 dBm.
- **pktcSigDevToneOnDuration**: The time, in milliseconds, to play the defined tone.
- **pktcSigDevToneOffDuration**: The time, in milliseconds, of silence before the next tone.
- **pktcSigDevToneFreqRepeatCount**: The number of times to play the defined tone.

The **pktcSigDevToneType** object acts as the index for both tables. The index is one of the following values:

- **`busy`**(1)
- **`confirmation`**(2)
- **`dial`**(3)
- **`messageWaiting`**(4)
- **`offHookWarning`**(5)
- **`ringBack`**(6)
- **`reOrder`**(7)
- **`stutterdial`**(8)
- **`callWaiting1`**(9)
- **`callWaiting2`**(10)
- **`callWaiting3`**(11)
- **`callWaiting4`**(12)

The **pktcSigDevToneSteady** object, when set to **`true`**(1), keeps the last tone on.

The **pktcSigDevToneWholeToneRepeatCount** object defines how many times to repeat the on/off sequence.

The following tones are not supported in the MIB tables or through the supported line package:

- **`alertingSignal`**(13)
- **`specialDial`**(14)
- **`specialInfo`**(15)
- **`release`**(16)
- **`congestion`**(17)
- **`userDefined1`**(18)
- **`userDefined2`**(19)
- **`userDefined3`**(20)
- **`userDefined4`**(21)

Action

Follow these steps to configure call progress tones.

1. Modify the **pktcSigDevToneTable** table to define the repeat count and whether the last tone in the sequence is steady.
2. Modify the **pktcSigDevMultiFreqToneTable** to define the frequencies and duration of each tone in the sequence.
3. To modify a Call Waiting tone, follow the first two steps and then:
   a. Set the **pktcNcsEndPntConfigCallWaitingDelay** object to define the amount of delay between repeats of the Call Waiting tones.

   Note: The **pktcSigDevToneWholeToneRepeatCount** object is ignored for the Call Waiting tones.

   b. Set the **pktcNcsEndPntConfigCallWaitingMaxRep** object to define the repeat count for the Call Waiting tones.

   Note: Do not use the **pktcSigDevMultiFreqToneTable** to configure the delay between repeated Call Waiting tones.

Gain Compensated Tone Generation

Touchstone firmware provides a patent-pending feature to automatically adjust FSK and CAS tone generation to compensate for MSO-selected loss value settings.

Touchstone firmware automatically adjusts FSK and tone levels to compensate for the loss plan values so that generated FSK and CAS tones are always within applicable specifications. You can make changes to the default levels as needed.

On-Hook vs. Off-Hook Gain

Touchstone firmware allows separate gain control for on-hook and off-hook levels. By default, the standard gain control MIB objects control levels for both conditions, but setting the **arrisMtaDevLevelControlOffHookEnable** object to `enable`(1) allows off-hook gain to be configured separately using the **arrisMtaDevLevelControlOffHookCAS** and **arrisMtaDevLevelControlOffHookFSK** objects.

arrisMtaDev...				
LevelControl			GainControl	
OffHookEnable	OffHookFSK	OffHookCAS	FSK	CAS
Disabled	Not used	Not used	Controls both on and off hook gains	Controls both on and off hook gains
Enabled	Controls off-hook gain	Controls off-hook gain	Controls only on-hook gain	Controls only on-hook gain

Action

Perform the following tasks as needed.

Configuring Gain Control using SNMP

Follow these steps as necessary to configure gain control settings.

1. Adjust eDVA-generated on-hook and default off-hook FSK tones (CID and VMWI) by setting the **arrisMtaDevGainControlFSK** object. Valid range: **–10** to **2** (dBm). Default: **0**.
2. Adjust the transmit digital gain adjustment for eDVA-generated on-hook and default off-hook CAS tone by setting the **arrisMtaDevGainControlCAS** object. Valid range: **–2** to **2** (dBm). Default: **0**.
3. To set off-hook FSK and CAS tone levels that are different from the on-hook levels:
   a. Set the off-hook FSK tone level by setting the **arrisMtaDevLevelControlOffHookFSK** MIB object.

      Valid range: **–32** to **–10** (dBm). Default: **–15**.
   b. Set the off-hook CAS tone level by setting the **arrisMtaDevLevelControlOffHookCAS** MIB object.

      Valid range: **–32** to **–10** (dBm). Default: **–15**.
   c. Enable the off-hook gain settings by setting the **arrisMtaDevLevelControlOffHookEnable** MIB object to **enable**(1). The default is **disable**(0), which uses the established **arrisMtaDevGainControlFSK** and **arrisMtaDevGainControlCAS** objects for both on-hook and off-hook levels.
4. Adjust the transmit digital gain adjustment for eDVA-generated Call Progress tones (dial tone, busy tone, ringback, etc.) to the CPE by setting the **arrisMtaDevGainControlLocalTone** object. Valid range: **–2** to **2** (dBm). Default: **0**.
5. Adjust the transmit digital gain adjustment for eDVA-generated Call Progress tones (ringback) to the network by setting the **arrisMtaDevGainControlNetworkTone** object. Valid range: **–2** to **2** (dBm). Default: **0**.
6. Adjust the transmit digital gain adjustment for eDVA-generated DTMF tones to the CPE by setting the **arrisMtaDevGainControlLocalDTMF** object. Valid range: **–15** to **9** (dBm). Default: **0**.

7. Adjust the transmit digital gain adjustment for eDVA-generated DTMF tones to the network by setting the **arrisMtaDevGainControlNetworkDTMF** object. Valid range: **–9** to **9** (dBm). Default: **0**.

CAUTION
Service affecting
Changing the delta Rx/Tx Gain from the default value based on the country template used may affect overall voice transmission quality, local tone levels, digit detection, and modem/fax tone detection. PESQ scores may also be affected when additional loss is introduced.

8. Adjust the transmit digital gain adjustment for voice by setting the **arrisMtaDevGainControlTxVoice** object. Valid range: **–2** to **2** (dBm). Default: **0**.

Note: This setting does not affect local tone or FSK levels.

9. Adjust the receive digital gain adjustment for voice by setting the **arrisMtaDevGainControlRxVoice** object. Valid range: **–16** to **16** (dBm), or **–128** (the default) to use the eDVA-wide setting.

Note: Even though the deltaRx/Tx Gain MIB objects are defined to be line-based, setting the object for any valid line sets the delta gain for all lines. Also, if the same delta Rx/Tx Gain object appears multiple times in the CM configuration file using different indexes, the eDVA uses the last instance to set the gain.

CAUTION
Potentially service-affecting
Setting endpoint gain too high or too low may disable the voice path. Lower settings beyond recommended levels may impact voice quality or fax or modem transmission.

10. Adjust the transmit digital gain adjustment for individual lines by setting the **arrisMtaDevEndPntGainControlTxVoice** object. Valid range: **–16** to **16** (dBm), or **–128** (the default) to use the eDVA-wide setting.
11. Adjust the receive digital gain adjustment for individual lines by setting the **arrisMtaDevEndPntGainControlRxVoice** object. Valid range: **–16** to **16** (dBm), or **–128** (the default) to use the eDVA-wide setting.

Provisioning Preset Downstream Frequencies

Use this procedure to provision one or more preset frequencies. This feature allows Touchstone E-UEs to quickly lock onto a known downstream during initial registration, reducing the initial time required for ranging and registering after installation. You can also clear the list of frequencies.

You provision preset downstream frequencies through SNMP using MIB objects.

Preset Frequency MIB Objects

The following MIB objects control preset frequencies. You can make changes to these objects using an SNMP manager or through the configuration file.

arrisCmDevPresetFrequency
Entries in a table of up to 20 preset frequencies.

arrisCmDevClearPresetFrequencies
Set to `true`(1) to clear the preset frequency table.

arrisCmDevClearCachedFrequencies
Set to `true`(1) to clear the cached frequencies.

Dial Pulse Support

Dial pulse support may be required to support subscriber equipment such as older rotary phones or alarm systems. Touchstone firmware provides two methods of dial pulse support:

- Direct relay—Touchstone eDVAs relay dial pulses to the CMS or P-CSCF.
- In-band tone relay (patent pending)—Touchstone E-UEs detect dial pulses and relay the information to the network as DTMF tones.
- Softswitch (CMS) dial pulse—Touchstone MTAs relay dial pulses to the CMS.

TS11.1 supports 20 pps dialing.

The support method used depends on the network configuration, and is selected by setting the ARRIS-proprietary **arrisMtaDevEndPntDialingMethod** MIB object as follows:

Value	Method	Description
1	Tone	(default) Enables DTMF detection only.
2	Pulse	Enables pulse dialing detection.
3	Tone & Pulse	Enables both DTMF and pulse dialing detection.
4	Pulse with DTMF Relay	Pulse dialing detection with DTMF in-band relay (gateway dial pulse, patent pending).
5	Tone & Pulse with DTMF Relay	Both DTMF and pulse dialing detection with DTMF in-band relay enabled (gateway dial pulse).

Note: Setting the **arrisMtaDevEndPntDialingMethod** object using an SNMP browser stores the value in non-volatile memory.

Inband DTMF Transmission

Touchstone firmware provides a feature to allow inband transmission of DTMF tones, even if a line is configured for pulse-only dialing. Once a call is established, Touchstone eDVAs pass all DTMF tones received from a CPE through the upstream voice path. eDVAs configured for pulse-only dialing operation drop any DTMF tones received before the call is fully established.

This feature allows a subscriber, provided with pulse-only dialing capabilities, to send DTMF tones to the far end. This might be used for calling card number entry, bank card data entry, automated service responses, and similar services.

Inband DTMF transmission is automatically enabled after a call has been established. No special configuration or changes to the dialing method are needed to enable this feature.

Note: If a line is provisioned as pulse-dialing only, the eDVA does not process received DTMF digits beyond treating them as audio to pass to the upstream voice path. If the CMS requests that the eDVA notify the CMS of any collected digits, the received DTMF tones are not reported.

Action

Follow these steps to configure dial pulse support on Touchstone eDVAs.

Note: The **arrisMtaDevEndPntDialingMethod** object setting is stored in non-volatile memory when set through SNMP after the eDVA has completed registration.

1. To enable Gateway (IPDT) dial pulse support, set the **arrisMtaDevEndPntDialingMethod** object to **5** (pulse and DTMF detection).
2. To enable softswitch dial pulse support, set the **arrisMtaDevEndPntDialingMethod** object to **3** (pulse and DTMF detection).
3. To disable dial pulse detection, set the **arrisMtaDevEndPntDialingMethod** object to **1**. This may be necessary in certain situations where internal house wiring problems cause occasional “phantom” dial pulse digits.

Gateway Dial Pulse Example

The following configuration file provides an example of how to configure Gateway dial pulse support.

```
TelephonyConfigFileBeginEnd = 1
SnmpMib = pktcMtaDevEnabled.0 true
SnmpMib = arrisMtaDevEndPntDialingMethod.1 toneAndPulseWithDTMFRelay
SnmpMib = arrisMtaDevEndPntDialingMethod.2 toneAndPulseWithDTMFRelay
SnmpMib = pktcSigDefCallSigTos.0 0
SnmpMib = pktcSigDefMediaStreamTos.0 0
SnmpMib = pktcSigTosFormatSelector.0 ipv4TOSOctet
SnmpMib = pktcMtaDevRealmOrgName.DEV50 "Really Amazing Telephone Company"
SnmpMib = pktcNcsEndPntConfigCallAgentId.9 "ca@sn05.dev2"
SnmpMib = pktcNcsEndPntConfigCallAgentId.10 "ca@sn05.dev2"
SnmpMib = pktcNcsEndPntConfigCallAgentUdpPort.9 2727
SnmpMib = pktcNcsEndPntConfigCallAgentUdpPort.10 2727
SnmpMib = pktcMtaDevCmsKerbRealmName.SN05.DEV2 "SWLAB.ATL.ARRIS"
SnmpMib = pktcMtaDevCmsIpsecCtrl.SN05.DEV2 true
SnmpMib = pktcNcsEndPntConfigMWD.9 10
SnmpMib = pktcNcsEndPntConfigMWD.10 10
```

```
SnmpMib = pktcMtaDevCmsUnsolicitedKeyNomTimeout.SN05.DEV2 20000
SnmpMib = pktcMtaDevRealmOrgName.SWLAB.ATL.ARRIS "Really Amazing Telephone
Company"
TelephonyConfigFileBeginEnd = 255
```

Configuring Hook Flash Timing

Follow these steps to configure hook flash timing.

Note: These settings must be made in the eDVA configuration file.

Default Timing Settings

The country code template determines the default minimum and maximum flash timings, overriding the defaults specified by the PacketCable MIB. See Country Code Templates for a list of default hook flash timings for each supported country code.

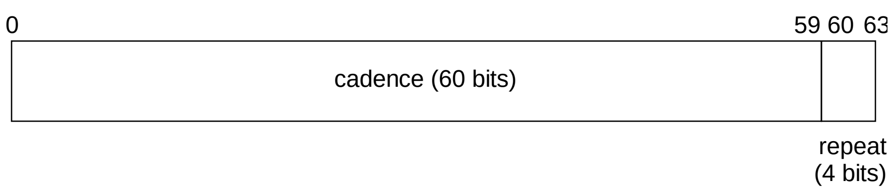
Action

Follow these steps to set hook flash timings.

1. In the eDVA configuration file, set the **pktcNcsEndPntConfigMinHookFlash** MIB object to the minimum time a line needs to be on hook for a valid hook flash. Use the **ifIndex** to specify the line; for example, **pktcNcsEndPntConfigMinHookFlash**.9 specifies line 1. Valid range: **`20`** to **`1550`** milliseconds.
2. In the eDVA configuration file, set the **pktcNcsEndPntConfigMaxHookFlash** MIB object to the maximum time the line needs to be on hook for a valid hook flash. Valid range: **`20`** to **`1550`** milliseconds.

Provisioning Ring Cadences

Ring cadence and ring splash may be provisioned through the PKTC-IETF-MTA-MIB objects **pktcSigDevRgCadence** (standard ring), **pktcSigDevR0Cadence** through **pktcSigDevR7Cadence** (distinctive ringing), and **pktcSigDevRsCadence** (ring splash). These objects consist of the following fields:



The fields are:

cadence

60 bits representing the ring cadence. Each bit represents 100 ms; **`1`** represents tone and **`0`** represents silence. The pattern can be up to 6 seconds long; to specify a shorter

pattern, begin the pattern with zeroes (for example, 15 zero bits at the beginning of the pattern would shorten the cadence to 4.5 seconds).

`repeat`
Specify either **`0000`** for a repeatable cadence, or **`1000`** for a non-repeatable cadence.

For example, a value of **00.00.0F.FF.FF.00.00.00** specifies a repeating ring cadence of 2 seconds on, 2 seconds off (the first two seconds of silence is discarded).

Note: The **pktcSigDevRsCadence** (ring splash) object must always be non-repeatable. The eDVA rejects any attempt to make the ring splash repeatable.

Post-Provisioning

Touchstone firmware provides the ability to post-provision lines and other device definitions, allowing you to add or remove service without restarting the eDVA.

NCS Post-Provisioning

Touchstone NCS loads are fully compliant with PacketCable eDVA Post-Provisioning functionality, described in section 7.6 of the *PacketCable MTA Device Provisioning Specification,* PKT-SP-PROV1.5-I04-090624. You can post-provision:

- CMS entries
- KDC realms
- Endpoints (lines)

Action

Follow these steps to post-provision a line on a Touchstone NCS load.

1. To add or remove a CMS, modify the **pktcMtaDevCmsTable** as required. You can add or remove entries in the table.
2. To add or remove a KDC realm, modify the **pktcMtaDevRealmTable** as required. You can add or remove entries in the table.
3. To add or remove an endpoint, modify the **pktcNcsEndPntConfigTable** as required. You can add or remove entries in the table.

Provisioning ARRIS SIP Loads

This chapter applies only to ARRIS SIP loads.

Overview of SIP Features

This section describes some important features of the SIP loads.

Barge-In

The `Join:` header (RFC 3911) may used to barge-in to an existing call. A brief tone is played to the existing call as the calls are conferenced together.

Loopback

The SIP eDVA can terminate loopback calls. When a loopback call is received, the phone does not ring. It automatically answers the call and loops back media or packets to the originator. The eDVA has a 2 call per line resource limitation. The user may make a call or receive a call while a loopback call is in progress. If another call is placed, either by an incoming call or a new outbound call, the loopback call is disconnected.

Packet Loopback is analogous to NETWLOOP, and controlled by the **sipCfgPacketLoopbackNumber** object. Media Loopback is analogous to NETWTEST, and controlled by the **sipCfgMediaLoopbackNumber** object.

The *Loopback Draft* (*http://www.ietf.org/internet-drafts/draft-ietf-mmusic-media-loopback-08.txt*) specifies the CODEC negotiation involved in setting up a loopback call.

Touchstone firmware also supports a proprietary loopback method. If the eDVA receives a call from a number matching those provisioned in specific MIB objects, it creates a loopback call.

Extended Offhook Processing

Touchstone firmware partially implements extended offhook processing as defined in PKT-SP-RSTF-I08-110127, section 7.1.4.4. Touchstone firmware implements extended offhook processing as shown below.

Origination Mode

Origination mode typically involves the subscriber picking up the phone, then either not dialing a number or pausing too long between digits. PacketCable defines an Origination Mode Dial Time timer and a Long Interdigit Timer (typically 16 seconds each); either timer expiring invokes the “permanent sequence.”

Use the **pktcEUERSTNfBCallOrigDTTimer** object to set the Dial Time timer. The default time is 16 seconds.

Use the **pktcEUERSTNfBCallOrigModLongIntDig** object to set the Long Interdigit timer. The default time is 16 seconds.

Termination Mode

When the eDVA receives the BYE signal, it starts a timer. Prior to TS7.6 MSUP2, the timer is fixed at 20 seconds. In TS7.6 MSUP2 and newer loads, the **sipCfgTermOffHookProcessingDelay** object controls the timer. The valid range for the timer is `0` to `60` seconds; the default is `20` seconds.

If the timer expires before the subscriber hangs up, the eDVA invokes the “permanent sequence.”

Permanent Sequence

A series of MIB objects, defined in the CL-PKTC-EUE-RST-MIB, specify the permanent sequence played in response to an extended offhook. The default behavior is:

D11PLUS loads:

1. Reorder tone for 30 seconds
2. OSI for 1 second
3. Silence for 4 seconds
4. Howler tone for 60 seconds

Other loads:

1. OSI for 1 second
2. Silence for 10 seconds
3. Howler tone for 60 seconds

If the subscriber has not hung up the phone by the time the sequence ends, the line enters a lockout state until the subscriber hangs up.

The following table shows the MIB objects used to provision the permanent sequence and the default values for each object.

Object	.TW Loads	Other Loads
pktcEUERSTNfBCallPermSeqTone1	"file:///PacketCableRST/ro"	"file:///PacketCableRST/osi"
pktcEUERSTNfBCallPermSeqTimer1	30	1

Object	.TW Loads	Other Loads
pktcEUERSTNfBCallPermSeqTone2	"file:///PacketCableRST/osi"	"file:///PacketCableRST/nt"
pktcEUERSTNfBCallPermSeqTimer2	1	10
pktcEUERSTNfBCallPermSeqTone3	"file:///PacketCableRST/nt"	"file:///PacketCableRST/ot"
pktcEUERSTNfBCallPermSeqTimer3	4	60
pktcEUERSTNfBCallPermSeqTone4	"file:///PacketCableRST/ot"	""
pktcEUERSTNfBCallPermSeqTimer4	60	0

Emergency Calls

Any number matching the pattern associated with the digit map action `EMERGENCY-CALL` is treated as an emergency call. Emergency calls have special treatment. The subscriber is, by default, not allowed to terminate an emergency call (this can be overridden by setting the **sipCfgSipFeatureSwitch** bit **`0x00000100`**). If the user goes on hook, and the default behavior is in effect, the eDVA does not send a BYE message. Instead, the call is put on “Network Hold” as follows:

1. When the subscriber goes on-hook during an emergency call, the Touchstone eDVA sends an INVITE (`priority:emergency SDP:a=inactive`). This causes the PSAP operator to hear a tone that indicates that the user went on-hook.
2. The PSAP operator can send an INVITE (`priority:emergency SDP:a=sendrecv`) that causes the eDVA to ring. If the subscriber goes off-hook before receiving such an INVITE, then it should send the invite to re-establish two-way communications.
3. The Network Hold Timer specifies the maximum time that an emergency call is preserved in the Network Hold state. The timer is started every time that the user goes on-hook during an emergency call, and is cleared if the user goes off hook. If the Network Hold Timer expires, then the eDVA sends a BYE message to finally terminate the call. The default value for the Network Hold Timer is 45 minutes.

Emergency calls have the following restrictions:

- Outgoing emergency calls offer only the G.711 CODEC in the SDP.
- Caller ID blocking is overridden on outgoing emergency calls.
- In the INVITE, the eDVA does not include the route received in the `Service-Route:` header that was received in the registration response.
- Incoming emergency calls cannot be transferred.

Incoming calls are treated as emergency calls under either of the following conditions:

- The `P-Asserted-Identity:` header value matches the value of the **sipCfgEmergencyServiceURN** object (the default is `URN:service:sos`). If you change this value, always begin the string with the **URN:** prefix to comply with RFC 5031.
- The Priority header value is “emergency.”

To disconnect any active call when originating or receiving an emergency call, set bit **`0x00001000`** in the **sipCfgSipFeatureSwitch**.

Distinctive Ringing

Several distinctive ringing types, corresponding to R0 through R7, are defined by the country template. If the `Alert-Info:` header is received, Touchstone firmware compares the header to the strings specified in the MIB objects **sipCfgAlertInfoR0** through **sipCfgAlertInfoR7**. If the header matches one of these, the eDVA plays the corresponding R0–R7 tone. TS11.1 also supports tones WT1 through WT4 for use with Call Waiting, and maps R1 through R4 to WT1 through WT4 for a Call Waiting alert.

The following configuration shows the default string settings.

```
{SnmpMib sipCfgAlertInfoR0.0 "< http://127.0.0.1/Bellcore-dr0 >"}
{SnmpMib sipCfgAlertInfoR1.0 "< http://127.0.0.1/Bellcore-dr1 >"}
{SnmpMib sipCfgAlertInfoR2.0 "< http://127.0.0.1/Bellcore-dr2 >"}
{SnmpMib sipCfgAlertInfoR3.0 "< http://127.0.0.1/Bellcore-dr3 >"}
{SnmpMib sipCfgAlertInfoR4.0 "< http://127.0.0.1/Bellcore-dr4 >"}
{SnmpMib sipCfgAlertInfoR5.0 "< http://127.0.0.1/Bellcore-dr5 >"}
{SnmpMib sipCfgAlertInfoR6.0 "< http://127.0.0.1/Bellcore-dr6 >"}
{SnmpMib sipCfgAlertInfoR7.0 "< http://127.0.0.1/Bellcore-dr7 >"}
```

Touchstone firmware supports both the Bellcore-defined tones shown in the listing above, and the equivalent PacketCable 2.0-defined tones `file:///PacketCableRST/r`*n*, where *n* is 0 through 7.

SIP Provisioning Considerations

This section describes SIP provisioning considerations.

Information Required for SIP

The SIP load requires three pieces of information to function properly:

Outbound P-CSCF

The destination device for all outbound messages. The setting is used as the domain in the Request-URI for all outgoing INVITE messages. The E-UE supports a “global” proxy that applies to all lines, and a per-line proxy that applies to a single line. See “*Configuring Per-Line Proxy and Registrar* (page 112)” for details.

Registrar

Registration messages are sent to the outbound proxy’s IP address, but the Request-URI address is provisioned as the registrar address. This setting may or may not be the same as the outbound proxy setting. The E-UE supports a “global” registrar that applies to all lines, and a per-line registrar that applies to a single line. See “*Configuring Per-Line Proxy and Registrar* (page 112)” for details.

Domain Settings

The domain is set as part of the DHCP process of the Telephony Modem using DHCP Option 15. This domain is used in all **to** and **from** URIs that the Telephony Modem generates.

The following are examples of REGISTER and INVITE messages with the following provisioning when bit **`0x04000000`** of the SIP Feature Switch is not set (bit value = **`0`**):

```
Domain             arris-i.org
sipCfgProxyAdr           ser.arris-i.org;5060
sipCfgRegistrarAdr registrar.arris-i.org;5060
User-id (phone #)        7705552001
Called Number            7705552002
REGISTER sip:registrar.arris-i.org:5060 SIP/2.0
From: "SIP1 Line1"<sip:7705552001@arris-i.org>; tag=94b73228-a013d16-13c4-
20-21829ddd-20
To: "SIP1 Line1"<sip:7705552001@arris-i.org>
Call-ID: 94b6e3b0-a013d16-13c4-20-22ce4f5-20
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 10.1.61.22:5060;branch=z9hG4bK-20-7ed6-6787cc6b
Allow: INVITE,ACK,BYE,CANCEL,NOTIFY
Max-Forwards: 70
Contact: "SIP1 Line1"<sip:7705552001@10.1.61.22:5060>
Content-Length: 0
INVITE sip:7705552002@ser.arris-i.org:5060 SIP/2.0
From: "SIP1 Line1"<sip:7705552001@arris-i.org>;
tag=94b73808-a013d16-13c4-62-4e4a1883-62
To: <sip:7705552002@arris-i.org>
Call-ID: 94b6f7d8-a013d16-13c4-62-5370cb92-62@arris-i.org
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.1.61.22:5060;branch=z9hG4bK-62-180b5-631cc050
Allow: INVITE,ACK,BYE,CANCEL,NOTIFY
Max-Forwards: 70
Contact: <sip:7705552001@10.1.61.22:5060>
Content-Type: application/SDP
Content-Length: 168
<SDP REMOVED>
```

Here are examples of REGISTER and INVITE with IP address-based provisioning with bit **`0x04000000`** set (bit value = **`1`**):

```
Domain arris-i.org
sipCfgProxyAdr 10.1.63.10;5060
sipCfgRegistrarAdr 10.1.63.11;5060
sipCfgSipFeatureSwitch 0x04000000
User-id (phone #) 7705552001
Called Number 7705552002
REGISTER sip:10.1.63.11:5060 SIP/2.0
From: "SIP1 Line1"<sip:7705552001@10.1.63.10:5060>; tag=94b73228-a013d16-
13c4-20-21829ddd-20
To: "SIP1 Line1"<sip:7705552001@10.1.63.10:5060>
Call-ID: 94b6e3b0-a013d16-13c4-20-22ce4f5-20
CSeq: 1 REGISTER
Via: SIP/2.0/UDP 10.1.61.22:5060;branch=z9hG4bK-20-7ed6-6787cc6b
Allow: INVITE,ACK,BYE,CANCEL,NOTIFY
Max-Forwards: 70
Contact: "SIP1 Line1"<sip:7705552001@10.1.61.22:5060>
Content-Length: 0
INVITE sip:7705552002@10.1.63.10:5060 SIP/2.0
From: "SIP1 Line1"<sip:7705552001@10.1.63.10:5060>; tag=94b73808-a013d16-
13c4-62-4e4a1883-62
To: <sip:7705552002@10.1.63.10:5060>
Call-ID: 94b6f7d8-a013d16-13c4-62-5370cb92-62@10.1.63.10
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.1.61.22:5060;branch=z9hG4bK-62-180b5-631cc050
Allow: INVITE,ACK,BYE,CANCEL,NOTIFY
Max-Forwards: 70
Contact: <sip:7705552001@10.1.61.22:5060>
```

```
Content-Type: application/SDP Content-Length: 168
<SDP REMOVED>
```

SIP Registration Behavior

All SIP Touchstone firmware loads use the same algorithm for normal registration and re-registration:

- Registration begins after a randomized wait time between 1 and (MaxWaitDelay) seconds. The default for MaxWaitDelay is 10 minutes; the **pktcNcsEndPntConfigMWD** object controls the delay time (in seconds). Note that line 1 uses index 9, line 2 uses index 10, and so on.
- By default, the registrar sends the registration in the 200 OK response to a REGISTER request. The **sipCfgRegExpires** object can specify a suggestion to the registrar for a desired registration expiry value. If the specified value is non-zero, the eDVA uses the smaller of the expiry value returned by the registrar in the 200 OK or the value specified in **sipCfgRegExpires**. If **sipCfgRegExpires** is unconfigured or specified to zero, no expires value is specified in the REGISTER request.
- After registration, the eDVA attempts to re-register after a random time between 50% and 75% of the expiration time.
- If registration is unsuccessful (for example, the eDVA receives no response or a 401 response), the eDVA retries registration using a backoff algorithm at intervals specified by RFC 3261.

The backoff algorithm is exponential. The initial backoff starts at the value specified by **sipCfgRegTimerMin** (default: 60 seconds). The interval between retry attempts doubles until it is greater than the value specified by **sipCfgRegTimerMax** (default: 1800 seconds). The interval between retries is not to exceed the value specified by **sipCfgRegTimerMax**. The eDVA uses the following formula to calculate the delay:

time = min(TimerMax, (TimerMin × 2^(#failures-1)))

The default values are 60 and 1800 seconds, and can be changed by specifying new values in the eDVA configuration file.

Each REGISTER attempt is not a single message. The eDVA will retry REGISTER messages based on the retransmission algorithm specified in RFC 3261. By default, the eDVA retries 7 times over a 32-second interval. The retransmission algorithm is also exponential. The basis interval is specified by T1 (**sipCfgT1**). The total time to retransmit is specified by TimerF (**sipCfgTimerF**).

SIP Feature Switch

The SIP feature switch enables or disables extended features in the SIP load. See “SIP Feature Switch” for a detailed description of each switch.

Provisioning Details

This section describes various SIP features that can be configured in the provisioning file.

Digit Map

SIP digit maps are identical to NCS digit maps:

- An **x** indicates any digit.
- The **|** (pipe) character separates different entries.

After each digit is pressed, the digit collector looks at each entry in the digit map to determine if any entry is complete. If an entry is complete, dialing is complete. There are some exceptions when star codes are specified. For a star code (VSC) to be handled, it must be in the digit map and possibly in another MIB object.

The below example is 10 digit dialing with 2 digit star codes.

```
{SnmpMib sipCfgDigitMap.0 "xxxxxxxxxx|*xx"}
```

Proxy Address

The proxy address specifies the destination where the eDVA sends all SIP requests. You must also specify the proxy type, IP or DNS.

```
{SnmpMib sipCfgProxyAdr.0 "ser.arris-i.org;5060"}
{SnmpMib sipCfgProxyType.0 dns}
```

Registrar Address

The registrar address specifies the text of the top line of the REGISTER request.

```
{SnmpMib sipCfgRegistrarAdr.0 "registrar.arris-i.org;5060"}
{SnmpMib sipCfgRegistrarType.0 dns}
```

Packetization Rate

The packetization rate can be set to 10 or 20 milliseconds.

```
{SnmpMib sipCfgPacketizationRate.0 20}
```

Provisioned CODEC Array

A semicolon-delimited list of CODEC types. This only affects outbound calls. The eDVA attempts to negotiate the CODECs in the list. CODECs are listed in order of preference and used only on initial outbound INVITE messages. Incoming requests answer with the CODECs supported in the initial offer's order.

```
{SnmpMib sipCfgProvisionedCodecArray.0 "PCMU;PCMA;G729;telephone-event"}
```

Repeat Dialing

The following MIB objects control repeat dialing.

sipCfgRepeatDialingInterval

Specifies the time between attempts to connect to other party.

sipCfgRepeatDialingTimeout

A timer value that specifies how long the eDVA should keep trying to connect to the other party.

sipCfgRepeatDialingSessionProgressTimer

Specifies the length of time after a 183 is received before considering the repeat dialing attempt successful. Often a 183 is received prior to a negative response. This timer prevents false positives.

These are the default values.

```
{SnmpMib sipCfgRepeatDialingInterval.0 30}
{SnmpMib sipCfgRepeatDialingTimeout.0 1800}
{SnmpMib sipCfgRepeatDialingSessionProgressTimer.0 2}
```

Call Forwarding Forbidden Numbers

There are certain phone numbers (eg. 911) that Call Forwarding should not accept as forwarding numbers. The **sipCfgCallForwardForbiddenNumbers** MIB object specifies which numbers are not allowed as forwarding numbers. This object can be set only in the configuration file. Multiple numbers may be specified separated by **|** as follows:

```
{SnmpMib sipCfgCallForwardForbiddenNumbers.0 "911|900|1900|0"}
```

Note: The emergency number set in **sipCfgEmergencyNumber** is also not allowed as a forwarding number.

Call Waiting setting Persistent Across Reboot

If the **sipCfgCallWaitingStarCodeSurvivesReset** MIB object is set to **true**, the call waiting state is set in non-volatile memory to ensure the state persists if the eDVA reboots.

```
{SnmpMib sipCfgCallWaitingStarCodeSurvivesReset.0 true}
```

To clear this setting, to re-locate the MTA to another customer that may have a different preference or for some other reason, use the **sipCfgResetCallWaitingStarCode** MIB object to clear the NVM setting. The MIB object controls the setting of this value on a per-line basis, using a bit string to indicate which lines should be reset.

```
# reset lines 1 and 2
{SnmpMib sipCfgResetCallWaitingStarCode.0 0x00000003}
```

Default G.711

When using G.729 as the primary voice CODEC, the eDVA must switch to a G.711 CODEC to support faxes transmission. Touchstone firmware switches to PCMU in this situation by default. Some customers require PCMA as the pass-through CODEC.

Use the **sipCfgDefaultG711** MIB object to specify a different pass-through CODEC, as follows:

```
{SnmpMIb sipCfgDefaultG711.0 pcma}
```

Domain Override

The **sipCfgDomainOverride** MIB object specifies the string to use in the domain in all outbound SIP signaling messages.

```
{SnmpMib sipCfgDomainOverride.0 "arris-i.org"}
```

Emergency Calls

Emergency calls have special treatment. This special treatment is determined based on the outbound dialed string. Provision the emergency number using the **sipCfgEmergencyNumber** object, as follows:

```
{SnmpMib sipCfgEmergencyNumber.0 911}
```

The end user is never allowed to terminate an emergency call. If the user goes on hook, the eDVA does not send a BYE message. Instead, the call is put on "Network Hold" as follows:

1. When the end user goes on-hook during an emergency call, the Touchstone eDVA sends an INVITE (`priority:emergency SDP:a=inactive`). This causes the PSAP operator to hear a tone that indicates that the user went on-hook.
2. The PSAP operator can send an INVITE (`priority:emergency SDP:a=sendrecv`) that causes the eDVA to ring. If the user goes offhook before receiving such an INVITE, then it should send the invite to re-establish two-way communications.
3. The Network Hold Timer specifies the maximum time that an emergency call is preserved in the Network Hold state. The timer is started every time that the user goes on-hook during an emergency call, and is cleared if the user goes off hook. If the Network Hold Timer expires, then the eDVA sends a BYE message to finally terminate the call. The default value for the Network Hold Timer is 45 minutes.

   See PKT-SP-RSTF-I08-110127 section 8.5.5.8 for more details on Network Hold for Emergency Calls, including call flows for various Network Hold scenarios.

Pulse Dialing

The **arrisMtaDevEndPntDialingMethod** MIB object enables pulse dialing. The object is indexed by line number.

```
{SnmpMib arrisMtaDevEndPntDialingMethod.1 toneAndPulse}
```

Line Specific Features

The **sipCfgPortFeatureSettings** MIB object enables various line-specific features. The object is a bit-mask; each bit controls a specific feature as shown in the following table.

Bit	Feature
0x80	Default
0x40	Caller ID disabled
0x20	Anonymous call rejection
0x10	Call Waiting disabled
0x08	Disable Three-Way Calling
0x04	Disable Caller ID display
0x02	Enable Call Transfer

Example:

```
{SnmpMib sipCfgPortFeatureSettings.1 02}
```

Distinctive Ringing

Several distinctive ringing types, corresponding to R0 through R7, are defined by the country template. If the Alert-Info: header is received, Touchstone firmware compares the header to the strings specified in the MIB objects **sipCfgAlertInfoR0** through **sipCfgAlertInfoR7**. If the header matches one of these, the eDVA plays the corresponding R0–R7 tone.

The following configuration shows the default string settings.

```
{SnmpMib sipCfgAlertInfoR0.0 "< http://127.0.0.1/Bellcore-dr0 >"}
{SnmpMib sipCfgAlertInfoR1.0 "< http://127.0.0.1/Bellcore-dr1 >"}
{SnmpMib sipCfgAlertInfoR2.0 "< http://127.0.0.1/Bellcore-dr2 >"}
{SnmpMib sipCfgAlertInfoR3.0 "< http://127.0.0.1/Bellcore-dr3 >"}
{SnmpMib sipCfgAlertInfoR4.0 "< http://127.0.0.1/Bellcore-dr4 >"}
{SnmpMib sipCfgAlertInfoR5.0 "< http://127.0.0.1/Bellcore-dr5 >"}
{SnmpMib sipCfgAlertInfoR6.0 "< http://127.0.0.1/Bellcore-dr6 >"}
{SnmpMib sipCfgAlertInfoR7.0 "< http://127.0.0.1/Bellcore-dr7 >"}
```

Dialing Features

The **sipCfgDialFeatTable** specifies dialing features that are handled by the eDVA. The following example assigns Anonymous Call Reject to the ***40** dialing code.

```
# Set up Anonymous Call Rejection as star code *40
{SnmpMib sipCfgDialFeatName.1 anonCallReject}
{SnmpMib sipCfgDialFeatCode.1 "*40" }
{SnmpMib sipCfgDialFeatTone.1 stutterTone}
# Works on first 4 lines
{SnmpMib sipCfgDialFeatActive.1 0.0.0.F}
# optional -- currently only used for
# repeat dialing (where its value should be 02)
{SnmpMib sipCfgDialFeatMode.1 01}
```

See “Supported Dialing Features” for a list of eDVA-supported dialing features.

Hybrid features, which are passed up to the proxy in an INVITE message, are controlled by the **sipCfgDialProxyTable**. The following example assigns ***50** as a hybrid feature.

```
# Active star code *50 as a hybrid feature
{SnmpMib sipCfgDialProxyCode.1 "*50" }
{SnmpMib sipCfgDialProxyTone.1 stutterTone}
# Works on first 4 lines
{SnmpMib sipCfgDialProxyActive.1 0.0.0.F}
# optional -- default is to use INVITE,
# may be used to send REFER instead.
{SnmpMib sipCfgDialProxyMessageType.1 01}
```

Proxy features are sent directly to the proxy in an INVITE message. No configuration is necessary to handle these features. The proxy itself defines what feature is associated with a particular dialing code.

Call Transfer

Uses the target dialog (RFC 4538) if supported by far-end; otherwise, sends REFER on the existing Call Leg.

In most cases, call transfer is initiated by the pivot phone going on hook. This triggers a transfer if Call Transfer is enabled, and the call is in one of the **threeWayCalling**, **callingHolding**, or conference states. When advanced flash digit handling is enabled, call transfer can be initiated by the **4** digit.

The transfer is performed by the pivot phone. The pivot phone sends a REFER message to the original call. This REFER message includes a ReferTo: header which notifies the party being transferred who to contact. Embedded in the ReferTo: header is a Replaces: header (RFC 3891). The party receiving the REFER message then sends an INVITE to the party referenced in the ReferTo: header. The Replaces: header is then copied to its own header in this INVITE. The party receiving the INVITE with the Replaces: header uses this header to determine which call to disconnect and replace with the transferred party.

Feature Capabilities

Ten MIB objects allow control of certain features on a per-line basis. The objects follow a naming convention of **sipCfg**`feat`**Capability**, where `feat` is the specific feature. Each MIB object consists of a hexadecimal bit string (32 bits wide) where each bit is a flag corresponding to a different line. These objects can be only set in the configuration file.

When a feature capability of a particular line is set to zero (off), that feature is disabled for that line regardless of the value of any other MIB object, including **sipCfgFeatureSettings**. By default, all feature capabilities are enabled for every line; all ten MIB objects default to the hex value **FFFFFFFF** (which is a string of 32 ones). When a feature is enabled, the user may still enable or disable the feature using dial codes. However, if the feature capability is disabled in the configuration file, the dial code to enable that feature does not work.

The following table lists the controllable features and the corresponding MIB object controlling its capability:

Feature	MIB Object
Caller ID Display	**sipCfgCallerIdDisplayCapability**
Caller ID Send	**sipCfgCallerIdSendCapability**
Anonymous Call Rejection	**sipCfgAnonCallRejectionCapability**
Call Waiting	**sipCfgCallWaitingCapability**
Three Way Calling	**sipCfgThreeWayCallCapability**
Call Transfer	**sipCfgCallTransferCapability**
Call Forwarding	**sipCfgCallForwardCapability**
Call Return	**sipCfgCallReturnCapability**
Call Redial	**sipCfgCallRedialCapability**
Call Holding	**sipCfgCallHoldCapability**

Examples:

To disable three-way calling for lines 1 and 3, add the following line to the configuration file:

```
{SnmpMib sipCfgThreeWayCallCapability.0 FFFFFFFA}
```

Note: For an 8- or 12-line Telephony Modem, this is identical to setting the value to 00000FFA because only the 12 least significant bits are used.

To disable three-way calling for all lines on a Telephony Modem, modify the setting as follows:

```
{SnmpMib sipCfgThreeWayCallCapability.0 00000000}
```

To enable three-way calls for all lines, delete any settings for **sipCfgThreeWayCallCapability** from the configuration file and reset the modem.

For any of these changes to take effect, the modem needs to be reset to download the updated configuration file.

Timers

- Timer T1 — **sipCfgT1** (PacketCable 2.0 MIB also specified); specifies the initial interval between retransmission attempts.
- Timer T2 — **sipCfgT2** (TS7.6 MSUP2 and newer, PacketCable 2.0 MIB also specified); specifies the maximum retransmit interval for non-INVITE requests and INVITE responses. See RFC 3261 for details.
- Timer T4 — PacketCable 2.0 MIB; see RFC 3261 for details.
- Timer B — **sipCfgTimerB** specifies the number of retransmission attempts or the time to keep retransmitting INVITE messages. The **sipCfgMaxRetrans** object (deprecated) specifies the number of retries.
- Timer F — **sipCfgTimerF** specifies the time to keep retransmitting non-INVITE messages.
- Timers D and H — **sipCfgInviteLinger**

- Timer K — **sipCfgGenLinger**

Minimal Example

The following configuration file fragment provides a minimal example of SIP configuration.

```
set mcns_config_params {
{TelephonyConfigFileBeginEnd 1}
# eDVA Enabled
{SnmpMib pktcMtaDevEnabled.0 1}
# Proxy Address
{SnmpMib sipCfgProxyAdr.0 "ser.arris-i.org;5060"}
{SnmpMib sipCfgProxyType.0 1}
# Registrar Address -- used in REGISTER request URI
{SnmpMib sipCfgRegistrarAdr.0 "registrar.arris-i.org;5060"}
{SnmpMib sipCfgRegistrarType.0 1}
#### Line 1 Configuration
# Phone Number
{SnmpMib sipCfgPortUserName.1 7705558001}
# Caller-ID Display
{SnmpMib sipCfgPortDisplayName.1 "David Line1"}
# Proxy Authentication Username
{SnmpMib sipCfgPortLogin.1 traff}
# Proxy Authentication Password
{SnmpMib sipCfgPortPassword.1 password}
# Line 1 Enabled
SnmpMib ifAdminStatus.9 1 }
# Try registration time within 10 seconds of getting config file
{SnmpMib pktcNcsEndPntConfigMWD.9 10}
#### Line 2 Configuration
# Phone Number
{SnmpMib sipCfgPortUserName.2 7705558002}
# Caller-ID Display
{SnmpMib sipCfgPortDisplayName.2 "David Line2"}
# Proxy Authentication Username
{SnmpMib sipCfgPortLogin.2 traff}
# Proxy Authentication Password
{SnmpMib sipCfgPortPassword.2 password}
# Line 2 Enabled
{SnmpMib ifAdminStatus.10 1 }
# Try registration time within 10 seconds of getting config file
{SnmpMib pktcNcsEndPntConfigMWD.10 10}
{TelephonyConfigFileBeginEnd 255}
}
```

Provisioning SIP Support

TS11.1 provides SIP support for eDVAs. This feature requires modifications to the CM configuration file and a new eDVA configuration file. This feature requires a specialized firmware load.

Per-Line Proxy/Registrar Objects

The following table describes MIB objects for per-line proxy/registrar support. These objects must be set in the configuration file to be effective. For details about the SIP registration process, see “*SIP Registration Behavior* (page 97).”

MIB Object		Description (Per-line)
Global	Per-line	
sipCfgRegistrarAdr	**sipCfgPortRegistrarAdr**	Registrar Server
	sipCfgPortRegistrarPort	Registrar Server port. The per-line setting is only valid if paired with a setting for **sipCfgPortRegistrarAdr**.
sipCfgRegistrarTyp e	**sipCfgPortRegistrarType**	The type of address specified by the **sipCfgPortRegistrarAdr** object. It is only valid if paired with a setting for **sipCfgPortRegistrarAdr**. When both objects are defined in the configuration file, this value overrides the setting in the **sipCfgRegistrarType** for this line. Valid values for this setting are `ipv4`(0) and `dns`(1).
sipCfgProxyAdr	**sipCfgPortProxyAdr**	Proxy Server address
	sipCfgPortProxyPort	Proxy Server port. It is only valid if paired with a setting for **sipCfgPortProxyAdr**.
sipCfgProxyType	**sipCfgPortProxyType**	The type of address specified by the **sipCfgPortProxyAdr** object. It is only valid if paired with a setting for **sipCfgPortProxyAdr**. When both objects are defined in the configuration file, this value overrides the setting in **sipCfgProxyType** for this line. Valid values for this setting are `ipv4`(0) and `dns`(1).

Use the line number as the index for each object; for example, **sipCfgPortProxyAdr**.2 specifies the proxy IP address for line 2.

Note: Per-line proxy/registrar is a *device*-only change; it cannot be changed using post-provisioning.

T.38 Provisioning Overview

The MIB object **sipCfgPortT38Mode** controls T.38 behavior. It allows the following values:

Value	Description
`t38Off`(1)	(default) Disables T.38.
`t38Loose`(2)	Enables T.38 Loose mode. In loose mode, the eDVA can use T.38 for fax transmission whether or not the remote endpoint has indicated T.38 support.
`t38Strict`(3)	Enables T.38 Strict mode. In strict mode, the eDVA can use T.38 for fax transmission only if the far end indicated support for T.38 during session negotiation.

The MIB object **arrisMtaDevEndPntFaxOnlyLineTimeout** configures fax-only mode for a line. This value of this object specifies the time, in seconds, to wait for fax or modem tones after receiving the SDP from the remote endpoint. If the time expires before detecting fax or modem tones, the eDVA drops the call. Valid range: **`0`** to **`600`**.

The MIB object **sipCfgPortMaxT38HSRedLevel** sets the maximum high-speed redundancy level used for T.38 fax relay, in both send and receive directions. The value of this object is the number of older data packets included in each T.38 datagram when transferring fax data. The actual redundancy level used is determined by negotiation with the remote endpoint. Valid range: **`0`** to **`2`**. Default: **`1`**.

Global Call Feature Control

The **sipCfgPortFeatureSettings** object allows you to control operation of the following call features:

- outbound Caller ID
- anonymous call rejection
- call waiting
- three-way calling

The **sipCfgPortFeatureSettings** object is structured as a collection of bit flags, as shown in the following table. The default value is **`0`**.

Bit	Description
0x40	callerIdPermanentDisable Set this bit to set the default outbound Caller ID method to "restrictive." The default setting presents Caller ID.
0x20	anonCallRejectionEnable Set this bit to enable anonymous call rejection. The subscriber can use a "star" code to disable anonymous call rejection if desired. The default setting permits anonymous calls.
0x10	callWaitingPermanentDisable Set this bit to disable Call Waiting. The subscriber can use a "star" code to enable Call Waiting if desired. The default setting is to enable Call Waiting.
0x08	threeWayCallingDisable Set this bit to disable hook flash processing during an active call. The default setting is to allow hook flash processing.
0x04	callIdReceiptDisable Set this bit to disable local CallerID display.
0x02	callTransferEnable Set this bit to enable Call Transfer. The bit can only be set in the configuration file.

Two objects control the persistence of Call Waiting settings:

sipCfgCallWaitingStarCodeSurvivesReset

Set this object to `true`(2) to enable storage of the Call Waiting Permanent Disable state in non-volatile memory. The default is `false`(1).

Note: If a subscriber disables Call Waiting (using a star code) with this object enabled, and the Telephony Modem is subsequently reissued to another subscriber, the new subscriber may assume that Call Waiting is disabled.

sipCfgResetCallWaitingStarCode

Set this object to `0xFFFFFFFF`, using an SNMP browser, to clear the Call Waiting Permanent Disable state from non-volatile memory for all lines.

Per-line Call Feature Control

Ten MIB objects provide per-line control over common calling features, allowing subscribers to order each feature separately. Each object is a map of 32 bits; the least significant bit represents line 1. Setting a bit to `0` disables the corresponding feature for that line.

These MIB objects must be set in the eDVA configuration file, and are not persistent across reboots. The default value for all these objects is `0xFFFFFFFF` (feature enabled on all lines).

Feature	MIB Object
Caller ID Display	**sipCfgCallerIdDisplayCapability**
Caller ID Send	**sipCfgCallerIdSendCapability**
Anonymous Call Rejection	**sipCfgAnonCallRejectionCapability**
Call Waiting	**sipCfgCallWaitingCapability**
Three-way Calling	**sipCfgThreeWayCallCapability**
Call Transfer	**sipCfgCallTransferCapability**
Call Forwarding	**sipCfgCallForwardCapability**
Call Return	**sipCfgCallReturnCapability**
Call Redial	**sipCfgCallRedialCapability**
Call Hold	**sipCfgCallHoldCapability**

Each Telephony Modem ignores bits beyond its line capacity. For example, on a TM702 Telephony Modem, **`0x00000003`** is equivalent to **`0xFFFFFFFF`**.

Action

Perform the following tasks as necessary:

CM Configuration File Changes

Follow these steps to modify the cable modem configuration file for SIP support.

1. Modify the **PcIpClassification** MIB. The Classification for Upstream and Downstream packets should be set up with a different port number to indicate the correct SIP source port(s). **PcIpSourcePortStart** and **PcIpSourcePortEnd** should be set to **5060** in both **UpstreamPacketClassification** and **DownstreamPacketClassification**.
2. (optional) Set the **arrisMtaCfgRTPDynPortStart** and **arrisMtaCfgRTPDynPortEnd** MIBs to the desired port range used for sending SIP RTP voice packets. The valid range for the start and end ports is **`1024`** to **`65535`**. The default range is **`49152`** to **`65535`**.

eDVA Configuration File Changes

Follow these steps to enable SIP support in the eDVA configuration file.

1. Modify the **ppCfgMtaCallpFeatureSwitch** object. To use QoS for a SIP eDVA load, the value of the feature switch must be set to `0x4020` or decimal `16416`. If these bits are not set in the feature switch, the eDVA transmits RTP packets using Best Effort. Other QoS-specific settings may be needed depending on the CMTS (and its firmware load) used.
2. Enable the eDVA device by modifying the **pktcMtaDevEnabled** object. The object is used to control the eDVA device; it is not line specific. The value must be set to `true` to provide dial tone on individual lines.
3. To enable individual lines, set the **ifAdminStatus** object for the line to **1**. The object **ifAdminStatus**.9 corresponds to line 1, **ifAdminStatus**.10 corresponds to line 2, and so on.
4. Add a **ppcfgMtaCountryTemplate** MIB setting to change tones, line card configuration, and certain Euro-PacketCable defaults (such as CallerID and VMWI) to match local requirements.
5. Enter the SIP user name by modifying the **sipCfgPortUserName** object for each line. The user name should be the phone number associated with the line. This will be used in the Caller ID string at a later date.

   For line specific provisioning, **sipCfgPortUserName**.1 and .2 (and so on) represent the line numbers. The user name must be less than 255 characters. If a string contains any special characters, the entire string must be enclosed within double quotes.
6. Enter the display name by modifying the **sipCfgPortDisplayName** object. This is the display name in the SIP messages that will be used for caller name delivery at a later date. The name must be less than 255 characters. If a string contains any special characters the entire string must be enclosed within double quotes.
7. Enter the login name by modifying the **sipCfgPortLogin** object. The login name is to be specified for each device to meet the requirements for HTTP digest authentication. To provide login name privacy, this setting when read displays as "XXXXXXX" (255 Max).
8. Enter the password by modifying the **sipCfgPortPassword** object. The password is the paired requirement for the HTTP authentication on SIP messages. In order to provide password privacy, this setting when read displays as "XXXXXXX" (255 Max).
9. Enter the digit map specification by modifying the **sipCfgDigitMap** object. The digit map support on ARRIS SIP eDVA is specified in the configuration file using TLV-43 and sub-TLV69. It applies to all the lines on the Telephony Modem.

   The format of the string is the same as a digit map used in MGCP (see "*Digit Map* (page 98)" for details). The maximum length for the digit map is 2048 characters.

Note 1: This digit map applies only to initial dialing. See "*Configuring Repeat Dialing* (page 118)" to set up busy indication dialing.

Note 2: TS7.1 does not support TLV-43; therefore, you cannot use TLV-43 and sub-TLV69 as in previous Touchstone firmware versions. If the digit map is less than 255 octets, you can use TLV-11.

10. To define a "busy" digit map (that is, in effect when busy tone is playing), specify the map using the **sipCfgBusyDigitMap** object. The digit map format is the same as for the default digit map, and can be set only in the configuration file.
11. If desired, modify the digit map timers:
    - T par (partial dial time-out)—modify the **pktcNcsEndPntConfigPartialDialTO** object. The default is `16` seconds.
    - T crit (critical dial timeout)—modify the **pktcNcsEndPntConfigCriticalDialTO** object. The default is `4` seconds.

Note: SIP loads support the NCS digit map timers.

12. Set the SIP call feature switch by modifying the **sipCfgSipFeatureSwitch** object. Each bit corresponds to a supported SIP call feature. See "SIP Feature Switch" for valid settings. The default value is `0`.
13. Enter the SIP provisioned CODEC by modifying the **sipCfgProvisionedCodecArray** object. This is the list of CODECs offered in the OFFER SDP. The order of this list is also the order of preference used in the OFFER SDP.

    The CODEC list is a string separated by semi-colons (';'). The default string is "PCMU;PCMA."
14. Enter the packetization rate by modifying the **sipCfgPacketizationRate** object. The supported packetization rate is 20 milliseconds.
15. (optional) Set the **arrisMtaCfgRTPDynPortStart** and **arrisMtaCfgRTPDynPortEnd** objects to the desired port range used for sending SIP RTP voice packets. The valid range for the start and end ports is `1024` to `65535`. The default range is `49152` to `65535`.
16. (optional) Modify the **sipCfgMaxUDPSize** object to set the maximum UDP packet size. SIP packets larger than this threshold are sent using TCP.

    Default: `0`
17. Globally enable or disable calling features, if desired, by adding the **sipCfgPortFeatureSettings** object. See "Call Feature Control" for a list of calling features this this object controls.
18. Enable or disable calling features on a per-line basis, if desired, by adding the **sipCfg** *feature***Capability** object for the specific features. See "Per-line Call Feature Control" for a list of calling features that can be enabled or disabled on each line.

Setting up Timers

Touchstone firmware provides MIB objects for controlling the registration and retransmission (T1) timers. These objects must be set in the eDVA configuration file.

Note: The default values for these objects correspond to RFC 3261 guidelines.

1. To configure the registration wait time, set the **sipCfgRegTimerMin** and **sipCfgRegTimerMax** objects to the desired minimum and maximum wait times.

   The defaults for these objects are **`0`** and **`1800`** seconds, respectively. At startup, an eDVA waits for a random amount of time (bounded by these objects) before beginning registration.

2. To change the T1 timer value, in milliseconds, set the **sipCfgT1** object. T1 is the base interval for the exponential back-off algorithm, used for retransmitting INVITE messages.

   Default: **`500`** (ms).

3. To change the number of transmission attempts for outgoing INVITE messages, set the **sipCfgMaxRetrans** object. The default is **`7`**.

   The defaults for this object and **sipCfgT1** provide the following retransmission timing:

   a. Initial INVITE (0 s)
   b. First retransmission (0.5 s) (500 ms delay)
   c. Second retransmission (1.5 s) (1 s delay)
   d. Third retransmission (3.5 s) (2 s delay)
   e. Fourth retransmission (7.5 s) (4 s delay)
   f. Fifth retransmission (15.5 s) (8 s delay)
   g. Sixth (final) retransmission (31.5 s) (16 s delay)

4. To specify a preferred session expiry timeout value, set the **sipCfgSessionExpires** object to the desired value (in seconds). When this object is set, the Telephony Modem configures the SIP INVITE message depending on the value of the object:

Value	INVITE Behavior
0	No Session-Expires header included.
1–89	The Session-Expires header includes a value of 1800.
90+	The Session-Expires header includes the value specified in the object.

   The default is **`1800`** seconds.

   Set this object to **`0`** to allow the proxy to control the session expiry timer.

5. To specify a suggested registration expiry timer value, set the **sipCfgRegExpires** object to the desired value. The Telephony Modem includes the specified expiry value in the REGISTER request contact header.

   The actual expiry time is the lesser of the suggested value and the expiry value returned in the 200 OK response.

   The default value is **0**, which omits the expiry value from the REGISTER request.

Configuring Per-Line Proxy and Registrar

Follow these steps to provision per-line proxy and registrar. Each MIB object specified must be set in the eDVA configuration file. These objects override the default proxy and registrar settings, so any line that does not specify per-line MIB objects uses the default proxy and registrar.

1. Override the default proxy address, port, and address type by setting the **sipCfgPortProxyAdr**, **sipCfgPortProxyPort**, and **sipCfgPortProxyType** objects in the eDVA configuration file; for example:

   ```
   { SnmpMIB sipCfgPortProxyAdr.line "host" }
   { SnmpMIB sipCfgPortProxyPort.line port }
   { SnmpMIB sipCfgPortProxyType.line type }
   ```

   Where:

   - *line* is the line number to use the override (**1** to the number of lines supported by the eDVA);
   - *host* and *port* are the IP address or FDQN, and port number of the SIP proxy;
   - *type* is the IP address type (**0** for IPv4 and **1** for DNS).

2. Override the default registrar address, port, and address type by setting the **sipCfgPortRegistrarAdr**, **sipCfgPortRegistrarPort**, and **sipCfgPortRegistrarType** objects in the eDVA configuration file; for example:

   ```
   { SnmpMIB sipCfgPortRegistrarAdr.line "ipaddr" }
   { SnmpMIB sipCfgPortRegistrarPort.line port }
   { SnmpMIB sipCfgPortRegistrarType.line type }
   ```

   Where:

   - *line* is the line number to use the override;
   - *ipaddr* and *port* is the IP address and port number of the SIP registrar;
   - *type* is the IP address type (**0** for IPv4 and **1** for DNS).

3. Restart the eDVA to make the per-line proxy/registrar settings take effect.

Specifying a SIP Domain Name

Follow these steps to specify a SIP domain name other than the domain specified in the proxy or provisioned FQDN.

1. Add the **sipCfgDomainOverride** object to the configuration file. This object must contain the desired domain name.

Note: When this object is set, the eDVA ignores the “Domain Override" SIP Feature Switch setting.

2. Restart the eDVA to download and apply the updated configuration.

Provisioning SIP Features

Use this procedure to configure end-user features with SIP loads.

Requirements and Limitations

To provision SIP features, you must first modify the CM and eDVA configuration files as described in "*Provisioning SIP Support* (page 104)."

You can provision up to 50 dialing features and up to 50 proxy dialing features in the eDVA configuration file.

Call Feature Control

The **sipCfgPortFeatureSettings** object allows you to control operation of the following call features:

- outbound Caller ID
- anonymous call rejection
- call waiting
- three-way calling

The **sipCfgPortFeatureSettings** object is structured as a collection of bit flags, as shown in the following table. The default value is **`0`.**

Bit	Description
0x40	callerIdPermanentDisable Set this bit to set the default outbound Caller ID method to “restrictive.” The default setting presents Caller ID.
0x20	anonCallRejectionEnable Set this bit to enable anonymous call rejection. The subscriber can use a “star” code to disable anonymous call rejection if desired. The default setting permits anonymous calls.
0x10	callWaitingPermanentDisable Set this bit to disable Call Waiting. The subscriber can use a “star” code to enable Call Waiting if desired. The default setting is to enable Call Waiting.
0x08	threeWayCallingDisable Set this bit to disable hook flash processing during an active call. The default setting is to allow hook flash processing.
0x04	callIdReceiptDisable Set this bit to disable local CallerID display.
0x02	callTransferEnable Set this bit to enable Call Transfer. The bit can only be set in the configuration file.

Proxy Dialing Features

Some dialing features require the eDVA to handle the tones, but the proxy handles the actual messaging. These are known as *hybrid features*. Five MIB objects control the setup and requirements for hybrid dialing features. The index number of each object groups the objects by dialing feature. These objects replace the **sipCfgDialProxyMap** object supported in loads prior to TS5.2.

sipCfgDialProxyNumber
The dialing feature number. See the table below for dialing feature values.

sipCfgDialProxyCode
A string containing the dialing code that activates the feature (for example, “*88”).

sipCfgDialProxyTone
The response tone; either `stutterTone`(1) or `dialTone`(2).

sipCfgDialProxyActive
A 32-bit string that identifies which lines enable the dialing feature. The least significant bit corresponds to line 1; for example, the value `3` activates the dialing feature on lines 1 and 2.

sipCfgDialProxyMessageType
For a proxy-based dialing feature, determines the type of message sent to the proxy: `invite`(1) or `refer`(2). Most dialing features should use `invite`(1).

sipCfgDialProxyMethod

(optional) Determines how the dialing code is passed to the proxy: **`default`**(0) prepends the dialing code to the dial string; **`pc20`**(1) sends the dialing code to the proxy as the host of the SIP URI and the dial string in a user parameter in the SIP URI.

Supported Dialing Features

Supported dialing features are:

Value	Feature
`anonCallReject`(1)	Anonymous Call Reject (ANNCJ)
`anonCallRejectDisable`(2)	Anonymous Call Reject Disable (ANCJD)
`callForwardBusy`(30)	Call Forward Busy (CALBE)
`callForwardBusyDisable`(31)	Call Forward Busy Disable (CALBD)
`callForwardUncond`(32)	Call Forward Fixed/Variable (Unconditional) (CALFV)
`callForwardUncondDisable`(33)	Call Forward Fixed/Variable Disable (CFFDS)
`callForwardNoAnswer`(34)	Call Forward No Answer (CALFN)
callForwardNoAnswerDisable(35)	Call Forward No Answer Disable (CFNAD)
`warmline`(36)	Warmline
`warmlineDisable`(37)	Disable Warmline
`callReturn`(50)	Call Return (CALRT)
`callReDial`(60)	Call Redial
`callHold`(61)	Call Hold
`repeatDialingEnable`(62)	Enable Repeat Dialing
`repeatDialingCancel`(63)	Cancel Repeat Dialing
`callWaitTempDisable`(70)	Call Waiting Temp Disable (CALWD)
`callWaitPermDisableToggle`(71)	Call Waiting Permanent Disable Toggle (CLWPD)
`callWaitPermanentDisable`(72)	Call Waiting Permanent Disable
`callWaitPermanentEnable`(73)	Call Waiting Permanent Enable
`callerIDPermBlockToggle`(90)	Caller ID Permanent Block Toggle (CIDPB)
`callerIDTempEnable`(91)	Caller ID Temp Enable (CIDTE)
`callerIDTempBlock`(92)	Caller ID Temp Block (CIDTB)

Action

Perform the following tasks as necessary:

Setting up Dialing Features

Add the MIB objects described below to the eDVA configuration file to set up dialing features. The **sipCfgDialFeatMap** MIB object, provided in earlier loads, is supported for backwards compatibility but no longer documented. The MIB objects for each feature are distinguished by the index; for example, **sipCfgDialFeatName**.1 and **sipCfgDialFeatName**.2 are two different features.

Note: Certain dialing features, including Hotline, Warmline, and Repeat Dialing, require further configuration. See the appropriate task for any extended configuration required.

1. Add the **sipCfgDialFeatName** object to enable particular dialing features. See "Supported Dialing Features" for a list of supported features.
2. Add the **sipCfgDialFeatCode** object to define a dialing code for a particular feature. You can specify up to three codes, separated by a comma, for each feature. For example, use ***70,1170** to allow either ***70** or **1170** to activate a feature.
3. Add the **sipCfgDialFeatTone** object to specify the confirmation tone used when activating a feature. The choices are **`stutterTone`** (default) and **`dialTone`**.
4. Add the **sipCfgDialFeatActive** object to assign a feature to one or more lines on the eDVA. The value for this object is a bit mask; each bit represents one line. The least significant bit corresponds to line 1.

   Examples: A value of **`0.0.0.3`** enables the dialing feature for line 1 and line 2. A value of **`0.0.0.2`** enables the feature only for line 2.
5. Set the **sipCfgDialFeatMode** object to enable the feature for the appropriate dialing phases. This object is a set of bits:

Bit Value	Description
0x02	Busy
0x01	Initial dialing

   These bits can be combined; a value of **`3`** allows the dialing code to be used both during initial dial tone and during a busy signal. The default is **`1`**.
6. For hybrid features—those features where the eDVA handles the tones, but the proxy handles the actual messaging—add the objects described in "Proxy Dialing Features."

Configuring Warmline or Hotline

The SIP load includes support for specifying a hotline or warmline number in the configuration file. A hotline automatically dials the specified number as soon as the specified line goes off-hook; a warmline automatically dials the specified number after providing dial tone for a specified amount of time. Add the objects described below to the eDVA configuration file to set up a hotline or warmline for a specific line.

1. In the configuration file, set the **sipCfgPortWarmOrHotlineNumber** object to the phone number to dial. This object is specific to a line, so it must be specified with the line number; for example, **sipCfgPortWarmOrHotlineNumber**.2 for line 2.
2. In the configuration file, set the **sipCfgPortWarmLineTimeout** object to the timeout value (in seconds) for a warmline, or to `0` for a hotline. This object is specific to a line, so it must be specified with the line number; for example, **sipCfgPortWarmLlineTimeout**.1 for line 1.
3. To allow the subscriber to specify a warmline number, add an entry to the **sipCfgDialFeatTable** in the configuration file. The following example specifies ***53** and ***54** as the feature codes to enable and disable warmline dialing, enabling the feature on lines 1 and 2.

```
SnmpMib = sipCfgDialFeatName.15 warmline
SnmpMib = sipCfgDialFeatCode.15 "*53"
SnmpMib = sipCfgDialFeatTone.15 stutterTone
SnmpMib = sipCfgDialFeatActive.15 hexstr: 0.0.0.3
SnmpMib = sipCfgDialFeatName.16 warmlineDisable
SnmpMib = sipCfgDialFeatCode.16 "*54"
SnmpMib = sipCfgDialFeatTone.16 stutterTone
SnmpMib = sipCfgDialFeatActive.16 hexstr: 0.0.0.3
```

   The **sipCfgDialFeatActive** object specifies the lines on which the specified feature is active. It consists of a series of bits, with the least significant bit corresponding to line 1. Other effects are:

   - Setting the warmline feature code uses the value set in the **sipCfgPortWarmLineTimeout** MIB object for the timeout. If the object is not set for a line in the configuration file, the default is 5 seconds.
   - A subscriber-specified warmline number replaces the value for **sipCfgPortWarmOrHotlineNumber** that was specified in the configuration file.
   - The **sipCfgPortWarmOrHotlineNumber** MIB object, whether set in the configuration file or by the subscriber, shows the currently-configured warmline number. If the subscriber disables warmline, This object contains a 0-length string.
   - If you do not configure warmline feature codes, the number specified in **sipCfgPortWarmOrHotlineNumber** is permanent and cannot be changed or disabled by the subscriber.

Configuring Repeat Dialing

Follow these steps to configure the Repeat Dialing feature. Configuring any busy indication dialing feature uses the same steps.

1. Add two entries to the **sipCfgDialFeatTable** in the eDVA configuration file, to set up and cancel Repeat Dialing. The following example enables Repeat Dialing when the subscriber presses ***5** during a busy signal, and cancels Repeat Dialing with **#5**:

```
/* enable Repeat Dialing */
{SnmpMib sipCfgDialFeatName.1 Integer 62}
{SnmpMib sipCfgDialFeatCode.1 String "*5"}
{SnmpMib sipCfgDialFeatTone.1 Integer 1} /* stutter */
{SnmpMib sipCfgDialFeatActive.1 HexString 0x00000003}
{SnmpMib sipCfgDialFeatMode.1 Integer 2} /* busy mode only */
/* cancel Repeat Dialing */
{SnmpMib sipCfgDialFeatName.2 Integer 63}
{SnmpMib sipCfgDialFeatCode.2 String "#5"}
{SnmpMib sipCfgDialFeatTone.2 Integer 1} /* stutter */
{SnmpMib sipCfgDialFeatActive.2 HexString 0x00000003}
/* Repeat Dialing timer settings */
{SnmpMib sipCfgRepeatDialingInterval.0 10}
{SnmpMib sipCfgRepeatDialingTimeout.0 30}
{SnmpMib sipCfgRepeatDialingSessionProgressTimer.0 60}
```

2. Add the **sipCfgBusyDigitMap** MIB object to the eDVA configuration file, specifying the strings that can be matched while the eDVA is playing a busy tone:

```
SnmpMib = sipCfgBusyDigitMap.0 String "*x|#x";
```

3. Add the following MIB objects to the eDVA configuration file to set related timers:

sipCfgRepeatDialingInterval
The time, in seconds, between repeat dialing attempts. Default: **`30`** seconds.

sipCfgRepeatDialingTimeout
The time, in seconds, that Repeat Dialing is active (and unsuccessful) before the eDVA cancels the feature. Default: **`1800`** seconds.

sipCfgRepeatDialingSessionProgressTimer
The time, in seconds, the eDVA waits after receiving a “183 Session Progress” provisional response before alerting the subscriber. This delay is needed because many PSTN calls receive this response before receiving a negative INVITE. Default: **`2`** seconds.

Note: This object must be set in the configuration file. Changes made to this object persist across reboots.

4. Reset the eDVA to enable the feature.

Configuring T.38 and Fax-Only Modes

Follow these steps to configure T.38 and fax-only modes for a line. Each MIB object specified must be set in the eDVA configuration file.

1. Set the T.38 mode by adding the **sipCfgPortT38Mode** MIB object to the eDVA configuration file. This MIB object is specific to a line, so it must be specified with the line number; for example, **sipCfgPortT38Mode** .2 for line 2.

   Valid settings are: **`t38Off`**(1), **`t38Loose`**(2), and **`t38Strict`**(3). The default is **`t38Off`**.

2. Set fax-only mode by adding the **sipCfgPortFaxOnlyTimeout** MIB object to the eDVA configuration file. This MIB object is specific to a line, so it must be specified with the line number; for example, **sipCfgPortFaxOnlyTimeout**.1 for line 1. The value specifies the timeout, in seconds, after which the eDVA drops the call if it does not detect fax or modem tones.

   Valid range: **`0`** (disabled) to **`600`** seconds. The default is **`0`**.

Configuring Distinctive Ring/Alert Tones

Follow these steps to set the expected Alert-Info strings for distinctive ringing and alert (Call Waiting) tones. Each MIB object specified must be set in the eDVA configuration file. For more information, see *Distinctive Ringing* (page 95).

1. In the eDVA configuration file, set the following MIB objects:

MIB Object	Description
sipCfgAlertInfoR0	The value of the Alert-Info header field to instruct the eDVA to play the R0 ring cadence.
sipCfgAlertInfoR1	The value of the Alert-Info header field to instruct the eDVA to play either the R1 ring cadence (call not active) or the WT1 call waiting tone (call active).
sipCfgAlertInfoR2	The value of the Alert-Info header field to instruct the eDVA to play either the R2 ring cadence (call not active) or the WT2 call waiting tone (call active).
sipCfgAlertInfoR3	The value of the Alert-Info header field to instruct the eDVA to play either the R3 ring cadence (call not active) or the WT3 call waiting tone (call active).
sipCfgAlertInfoR4	The value of the Alert-Info header field to instruct the eDVA to play either the R4 ring cadence (call not active) or the WT4 call waiting tone (call active).
sipCfgAlertInfoR5	The value of the Alert-Info header field to instruct the eDVA to play the R5 ring cadence.
sipCfgAlertInfoR6	The value of the Alert-Info header field to instruct the eDVA to play the R6 ring cadence.
sipCfgAlertInfoR7	The value of the Alert-Info header field to instruct the eDVA to play the R7 ring cadence.

   The default value for each MIB object is `<file://Bellcoredr`*x*`>`, where *x* is the ring cadence (0 through 7). You may need to change these if the P-CSCF sends different Alert-Info strings.

1. To configure distinctive ringing to alert the subscriber to a call on hold when the line is on hook, set the **sipCfgDistinctiveRingingForCallHold** object. The allowed values are:
   - **`standard`**(0) (default) — use the standard ring cadence.
   - **`r0`**(1) — use the R0 ring cadence.

- `r1`(2) — use the R1 ring cadence.
- `r2`(3) — use the R2 ring cadence.
- `r3`(4) — use the R3 ring cadence.
- `r4`(5) — use the R4 ring cadence.
- `r5`(6) — use the R5 ring cadence.
- `r6`(7) — use the R6 ring cadence.
- `r7`(8) — use the R7 ring cadence.

Chapter 5

Provisioning PacketCable 2.0 SIP Loads

This chapter applies only to SIP PC20 loads.

PacketCable 2.0 Concepts

This section describes PacketCable 2.0 terminology and concepts.

Terminology

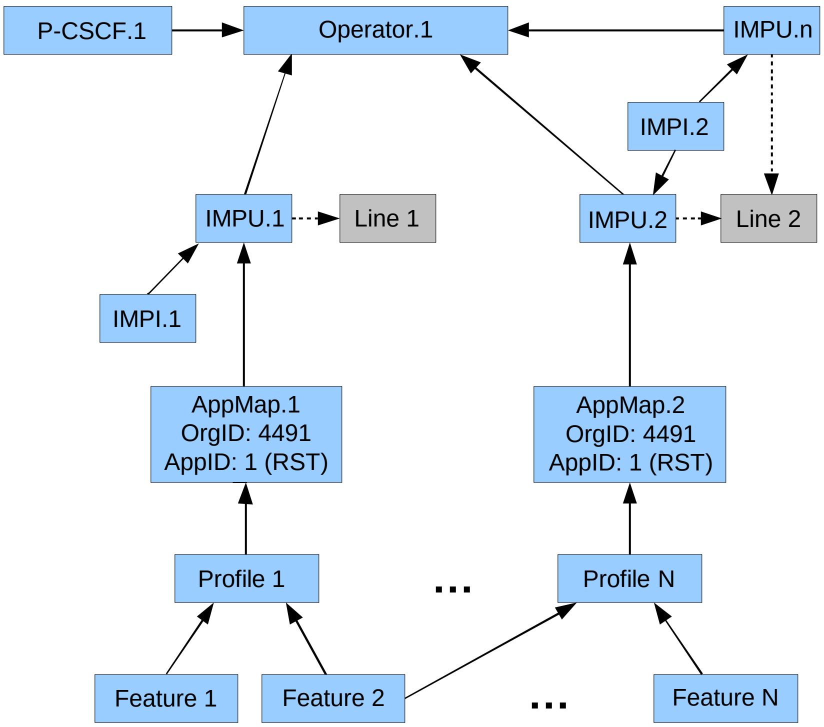
PacketCable 2.0 introduces new terminology and abbreviations for telephony components within a network. The following table shows PacketCable 1.x terms with their equivalent PacketCable 2.0 terms.

PC1.x	PC2.0	Description
eMTA	E-UE, E-DVA	A device consisting of a combined DOCSIS CM and a PacketCable eDVA (MTA).
MTA	eUE, eDVA	The logical PacketCable device.
Proxy	P-CSCF	Proxy-Call Session Control Function. In PacketCable 2.0, the P-CSCF communicates with the registrar, so it is no longer necessary to specify a registrar in eDVA configuration.

This document uses the terms *E-UE* for the entire Telephony Modem, *eDVA* for the telephony portion, and *CM* for the cable modem portion.

Configuration Concepts

Provisioning an eDVA conforms to the following model. Note that TS11.1 uses a simplified version of the PacketCable 2.0 model, and the simplified version is shown as follows:



The PacketCable 2.0 Residential SIP Telephony (RST) model is an abstract representation of the capabilities and features available on a phone line. The intent is to provide flexibility for future services. The RST specification introduces the following terms:

Operator

The operator indirectly associates users with a P-CSCF. Multiple operators could be associated with various third-party telephony providers or with remote locations in the MSO's network. TS11.1 supports one operator definition.

User

A user consists of two components:

- IMPU (IP Multimedia Public User) — the public identity of the user. This could be the phone number assigned to the user, or some other unique user name. The IMPU associates a user with a phone line through the **pktcEUEUsrIMPUAdditionalInfo** object. Multiple users may share a line, implementing a Teen Line service or as part of a transition to a new area code or exchange.
- IMPI (IP Multimedia Private Identity) — the private identity of the user. Primarily, the IMPI defines the credentials for the IMPU.

Application

Defines the service(s) to be provided to a user. TS11.1 supports only the Residential SIP

Telephony (RST) application. The application map defines the application and selects a profile.

Profile

Selects a digit map and an associated collection of features. A profile may be applied to one or more users.

Feature

Defines capabilities and dialing features available to a profile. Features include basic call functionality (including timer durations), announcements, Caller ID display, and common dialing features.

Each feature is defined as an entry in one or more tables, allowing profiles to contain different feature configurations. User-based features are indexed normally; network-based features (or network-based components of features) use the Operator index.

The following restrictions apply:

- Touchstone firmware supports only one P-CSCF (but can fail over to a second P-CSCF) and only one operator per E-UE.
- Touchstone firmware supports only the PacketCable Residential SIP Telephony (RST) application.
- Multiple users can be assigned to a line, but a user can be associated only with one line.

Each box in the above figure is associated with a table in the PacketCable 2.0 MIBs. Arrows represent an index to a row in each table, connected with another table. The boxes labeled IMPU and IMPI are contiguous since TS11.1 requires them to be permanently associated.

Supported Features

The following table shows which RST feature MIB tables are supported in TS11.1.

RST Feature	MIB Table Support	
	Network	User
Basic Call	Partial	Partial
Announcement	Partial	No
Activation Status	n/a	Partial
No Answer Timeout	n/a	Partial
CallerID	n/a	Partial
CallerID Display	n/a	Partial
CallerID Block	n/a	No
CallerID Delivery	n/a	No
Call Forward	No	Partial
Call Hold	n/a	No
Call Transfer	n/a	No
Do Not Disturb	n/a	No
MWI	Partial	n/a
Auto Recall	n/a	No
Auto Callback	n/a	No
Busy Line Verify	No	n/a
Emergency Services	Partial	n/a
Selective Call Forward	n/a	No

For details about partially supported feature tables, see see "*Standards Compliance* (page 17).

DHCP Option 60 Support

TS11.1 uses DHCP option 60 in DHCP Discover messages to advertise PacketCable support. The option contains the string “pktc2.0” to indicate PacketCable 2.0 support.

The following table lists the sub-options sent with DHCP option 60 and their meanings:

Sub-option	Value	Description
1	0x02	PacketCable 2.0
2	0x02	Number of endpoints (2)
3	0x00	TGT support (no)
4	0x00	HTTP download file access method (no)
5	0x01	MTA-24 Event Syslog notification
9	0x01	NVRAM Ticket/Session Key Storage supported
10	0x01	Provisioning Event Reporting supported
11	0x0106090f	Supported CODECs:
		01 = other
		06 = PCMU
		09 = PCMA
		0F = telephone-event
13	0x01	Echo Cancellation supported
16	0x09	**ifIndex** of first phone line
18	0x0007	Supported provisioning flows: Secure, Hybrid, Basic
19	0x01	T.38 Version support (v0)
20	0x01	T.38 Error Correction support (redundancy)
21	0x01	RFC 2833 DTMF relay supported
22	0x01	Voice Metrics supported
23	0x02001f	CableLabs MIBs supported: EN-SIG, EN-MTA, EVENT-MIB, SIG-MIB, MTA-MIB
24	0x00	Multiple Grants per Interval (no)
25	0x00	V.152 support (no)
26	0x00	Certificate Bootstrapping (no)

Overview of SIP Features

This section describes some important features of the SIP loads.

Barge-In

The `Join:` header (RFC 3911) may used to barge-in to an existing call. A brief tone is played to the existing call as the calls are conferenced together.

Loopback

The SIP eDVA can terminate loopback calls. When a loopback call is received, the phone does not ring. It automatically answers the call and loops back media or packets to the originator. The eDVA has a 2 call per line resource limitation. The user may make a call or receive a call while a loopback call is in progress. If another call is placed, either by an incoming call or a new outbound call, the loopback call is disconnected.

Packet Loopback is analogous to NETWLOOP, and controlled by the **sipCfgPacketLoopbackNumber** object. Media Loopback is analogous to NETWTEST, and controlled by the **sipCfgMediaLoopbackNumber** object.

The *Loopback Draft* (*http://www.ietf.org/internet-drafts/draft-ietf-mmusic-media-loopback-08.txt*) specifies the CODEC negotiation involved in setting up a loopback call.

Touchstone firmware also supports a proprietary loopback method. If the eDVA receives a call from a number matching those provisioned in specific MIB objects, it creates a loopback call.

Extended Offhook Processing

Touchstone firmware partially implements extended offhook processing as defined in PKT-SP-RSTF-I08-110127, section 7.1.4.4. Touchstone firmware implements extended offhook processing as shown below.

Origination Mode

Origination mode typically involves the subscriber picking up the phone, then either not dialing a number or pausing too long between digits. PacketCable defines an Origination Mode Dial Time timer and a Long Interdigit Timer (typically 16 seconds each); either timer expiring invokes the “permanent sequence.”

Use the **pktcEUERSTNfBCallOrigDTTimer** object to set the Dial Time timer. The default time is 16 seconds.

Use the **pktcEUERSTNfBCallOrigModLongIntDig** object to set the Long Interdigit timer. The default time is 16 seconds.

Termination Mode

When the eDVA receives the BYE signal, it starts a timer. Prior to TS7.6 MSUP2, the timer is fixed at 20 seconds. In TS7.6 MSUP2 and newer loads, the

sipCfgTermOffHookProcessingDelay object controls the timer. The valid range for the timer is **0** to **60** seconds; the default is **20** seconds.

If the timer expires before the subscriber hangs up, the eDVA invokes the "permanent sequence."

Permanent Sequence

A series of MIB objects, defined in the CL-PKTC-EUE-RST-MIB, specify the permanent sequence played in response to an extended offhook. The default behavior is:

D11PLUS loads:

1. Reorder tone for 30 seconds
2. OSI for 1 second
3. Silence for 4 seconds
4. Howler tone for 60 seconds

Other loads:

1. OSI for 1 second
2. Silence for 10 seconds
3. Howler tone for 60 seconds

If the subscriber has not hung up the phone by the time the sequence ends, the line enters a lockout state until the subscriber hangs up.

The following table shows the MIB objects used to provision the permanent sequence and the default values for each object.

Object	.TW Loads	Other Loads
pktcEUERSTNfBCallPermSeqTone1	"file:///PacketCableRST/ro"	"file:///PacketCableRST/osi"
pktcEUERSTNfBCallPermSeqTimer1	30	1
pktcEUERSTNfBCallPermSeqTone2	"file:///PacketCableRST/osi"	"file:///PacketCableRST/nt"
pktcEUERSTNfBCallPermSeqTimer2	1	10
pktcEUERSTNfBCallPermSeqTone3	"file:///PacketCableRST/nt"	"file:///PacketCableRST/ot"
pktcEUERSTNfBCallPermSeqTimer3	4	60
pktcEUERSTNfBCallPermSeqTone4	"file:///PacketCableRST/ot"	""
pktcEUERSTNfBCallPermSeqTimer4	60	0

Emergency Calls

Any number matching the pattern associated with the digit map action `EMERGENCY-CALL` is treated as an emergency call. Emergency calls have special treatment. The subscriber is, by default, not allowed to terminate an emergency call (this can be overridden by setting the **sipCfgSipFeatureSwitch** bit **0x00000100**). If the user goes on hook, and the default behavior is in effect, the eDVA does not send a BYE message. Instead, the call is put on "Network Hold" as follows:

1. When the subscriber goes on-hook during an emergency call, the Touchstone eDVA sends an INVITE (`priority:emergency SDP:a=inactive`). This causes the PSAP operator to hear a tone that indicates that the user went on-hook.
2. The PSAP operator can send an INVITE (`priority:emergency SDP:a=sendrecv`) that causes the eDVA to ring. If the subscriber goes off-hook before receiving such an INVITE, then it should send the invite to re-establish two-way communications.
3. The Network Hold Timer specifies the maximum time that an emergency call is preserved in the Network Hold state. The timer is started every time that the user goes on-hook during an emergency call, and is cleared if the user goes off hook. If the Network Hold Timer expires, then the eDVA sends a BYE message to finally terminate the call. The default value for the Network Hold Timer is 45 minutes.

Emergency calls have the following restrictions:

- Outgoing emergency calls offer only the G.711 CODEC in the SDP.
- Caller ID blocking is overridden on outgoing emergency calls.
- In the INVITE, the eDVA does not include the route received in the `Service-Route:` header that was received in the registration response.
- Incoming emergency calls cannot be transferred.

Incoming calls are treated as emergency calls under either of the following conditions:

- The `P-Asserted-Identity:` header value matches the value of the **sipCfgEmergencyServiceURN** object (the default is `URN:service:sos`). If you change this value, always begin the string with the **URN:** prefix to comply with RFC 5031.
- The Priority header value is "emergency."

To disconnect any active call when originating or receiving an emergency call, set bit **`0x00001000`** in the **sipCfgSipFeatureSwitch**.

Distinctive Ringing

Several distinctive ringing types, corresponding to R0 through R7, are defined by the country template. If the `Alert-Info:` header is received, Touchstone firmware compares the header to the strings specified in the MIB objects **sipCfgAlertInfoR0** through **sipCfgAlertInfoR7**. If the header matches one of these, the eDVA plays the corresponding R0–R7 tone. TS11.1 also supports tones WT1 through WT4 for use with Call Waiting, and maps R1 through R4 to WT1 through WT4 for a Call Waiting alert.

The following configuration shows the default string settings.

```
{SnmpMib sipCfgAlertInfoR0.0 "< http://127.0.0.1/Bellcore-dr0 >"}
{SnmpMib sipCfgAlertInfoR1.0 "< http://127.0.0.1/Bellcore-dr1 >"}
{SnmpMib sipCfgAlertInfoR2.0 "< http://127.0.0.1/Bellcore-dr2 >"}
{SnmpMib sipCfgAlertInfoR3.0 "< http://127.0.0.1/Bellcore-dr3 >"}
{SnmpMib sipCfgAlertInfoR4.0 "< http://127.0.0.1/Bellcore-dr4 >"}
{SnmpMib sipCfgAlertInfoR5.0 "< http://127.0.0.1/Bellcore-dr5 >"}
{SnmpMib sipCfgAlertInfoR6.0 "< http://127.0.0.1/Bellcore-dr6 >"}
{SnmpMib sipCfgAlertInfoR7.0 "< http://127.0.0.1/Bellcore-dr7 >"}
```

Touchstone firmware supports both the Bellcore-defined tones shown in the listing above, and the equivalent PacketCable 2.0-defined tones `file:///PacketCableRST/r`*n*, where *n* is 0 through 7.

Configuring PacketCable 2.0 SIP

Use this procedure to configure PacketCable 2.0 SIP for Touchstone products.

Configuration Overview

The most efficient way to configure PacketCable 2.0 SIP is to use a “top-down/bottom-up” approach, building and interlinking the MIB tables in the following order:

1. Operator table
2. P-CSCF table
3. Feature and Digit Map tables
4. Profile to Feature tables
5. Application Map tables
6. IMPI table
7. IMPU table

See *PacketCable 2.0 Concepts* (page 122) for an overview of how the tables interconnect.

Action

Perform the following tasks in the order shown.

Configuring Operator Information

Operator information includes the Operator and P-CSCF tables. Follow these steps:

1. Configure the Operator table. The index for each object is typically **1**; this index is used in other tables to refer to this Operator table.
   a. Set the **pktcEUEDevOpDomain** object to the domain name; for example, atl.ga.example.com.

b. Set the **pktcEUERSTKeepAliveSetting** object to set the in-service/out-of-service status:
   - Set to **`on`**(1) to use the keep-alive mechanism to determine the status.
   - Set to **`off`**(2) to use the RSTF in-service state requirements.
   - Set to **`conditional`**(3) to set the keep-alive mechanism depending on the requirements in the `200 OK` response to the REGISTER message.

c. Set the **pktcEUEDevOpRowStatus** object to **`createAndGo`**.

2. Configure the P-CSCF table. The index for each object is *.oper*.**1**, where oper is the index of the operator table (usually 1).

   a. Configure the P-CSCF address:

   pktcEUEDevPCSCFAddrType
   Either **`ipv4`** or **`ipv6`** to specify the IP address type, or **`dns`** to specify a fully-qualified domain name.

   pktcEUEDevPCSCFAddr
   The address of the P-CSCF associated with the operator.

   pktcEUEDevPCSCFSipPort
   (optional) The SIP port. Default: **`5060`**.

   b. (optional) If desired, modify the following SIP timers:

   pktcEUEDevPCSCFTimerT1
   SIP Timer T1. Default: **`500`** ms.

   pktcEUEDevPCSCFTimerT2
   SIP Timer T2. Default: **`4000`** ms.

   pktcEUEDevPCSCFTimerT4
   SIP Timer T4. Default: **`5000`** ms.

   pktcEUEDevPCSCFTimerTD
   SIP Timer TD. Default: **`32000`** ms.

3. Set the **pktcEUEDevPCSCFRowStatus** object to **`createAndGo`**.

Configuring Users and Features

Follow these steps to configure user and feature tables:

1. To configure features, proceed to *Provisioning PacketCable 2.0 Features* (page 133).
2. To configure user information, proceed to *Provisioning PacketCable 2.0 Users* (page 145).
3. To configure application profiles, proceed to *Provisioning PacketCable 2.0 Application Profiles* (page 147). Digit map configuration is described in *Configuring PacketCable 2.0 Digit Maps* (page 139).
4. To configure the Application Map, proceed to *Provisioning PacketCable 2.0 Application Maps* (page 148).

Configuring Extended Offhook Processing

Follow these steps to provision extended offhook processing.

1. Provision the origination mode timers (Origination Mode Dial Time and Long Interdigit Timer) in one of the two following ways:
   a. In the digit map, configure the **T** (Dial Time) and **L** (Long Interdigit Timer) timers.
   b. Configure the following two MIB objects:
      - **pktcEUERSTNfBCallOrigDTTimer**
      - **pktcEUERSTNFBCallOrigModLongIntDig**

   The default value for both timers is 16 seconds.

Note: Use either the digit map or the MIB objects to configure these timers; do not use both.

2. To set the sequence of tones played, and the duration of each, set the following MIB objects:

   pktcEUERSTNfBCallPermSeqTone1
   The URI of tone 1. Example: `file:///PacketCableRST/ro`

   pktcEUERSTNfBCallPermSeqTimer1
   The duration, in seconds, for tone 1.

   pktcEUERSTNfBCallPermSeqTone2
   The URI of tone 2.

   pktcEUERSTNfBCallPermSeqTimer2
   The duration, in seconds, for tone 2.

   pktcEUERSTNfBCallPermSeqTone3
   The URI of tone 3.

   pktcEUERSTNfBCallPermSeqTimer3
   The duration, in seconds, for tone 3.

Note: The **pktcEUERSTNfBCallPermSeqTone4** and **pktcEUERSTNfBCallPermSeqTimer4** objects are not supported.

3. (optional) Configure the offhook processing delay time (in seconds) by setting the **sipCfgTermOffHookProcessingDelay** object. Valid range: **`0`** to **`60`** seconds. Default: **`20`** seconds.

Post-Provisioning SIP Lines

Follow these steps to change the provisioning status of one or more lines without rebooting the eDVA.

Note: This task applies only to line-level parameters.

The following MIB objects require a restart of the SIP stack and are not updated by this feature:

- **sipCfgGenLinger**
- **sipCfgTimerF**

1. Create a configuration file containing the line-level parameters to change, and make the file available for download on a reachable TFTP server.
2. Using an SNMP network manager, set the **arrisMtaDevSipConfigFileURL** object to the URL of the configuration file.
3. Using an SNMP network manager, set the **arrisMtaDevSipDwnldConfig** object to **on**(2).

   The eDVA downloads the configuration file and applies the line-level changes. Touchstone firmware does not take lines out of service or interrupt calls in process.
4. To make the configuration changes permanent, modify the standard configuration file to reflect the changes.

Note: If the post-provisioning changes are not copied to the standard configuration file, the eDVA loses those changes when re-initialized.

Configuring T.38 and Fax-Only Modes

Configuring T.38 and fax operation is identical for both PC20 and ARRIS SIP loads. See Configuring SIP T.38 and Fax-Only Modes for details.

Provisioning PacketCable 2.0 Features

Use this procedure to provision individual PacketCable 2.0 features.

Feature Support

In PacketCable 2.0, a "feature" includes anything that can be enabled or configured. If all features are disabled, the eDVA can receive incoming calls and make only emergency outgoing calls. By default, all features are enabled and have reasonable default provisioning.

All features are defined and enabled in the **pktcEUERSTAppProfileToFeatTable**. Some features may be further configured as follows:

digit map

In addition to defining valid phone numbers, the digit map defines Vertical Service Codes used to invoke or control dialing features.

Network table

Network tables are associated with an operator table, and control provisioning associated with all users. Not all features have network tables.

User table

User tables are associated with individual users, and control provisioning for that user. Not all features have user tables.

The **pktcEUERSTAppFeatID** object specifies a feature type. The following are valid features:

- `digitMap`(2) — A digit map that enables the supported features.
- `basicCall`(3) — Basic Call capabilities. The user-side table defines the CODECs available; the network-side table defines timers and the off-hook alert sequence.
- `announcement`(4) — Announcements. The Announcement Map table defines tones to play when the eDVA receives various response codes (such as "486 Busy").
- `statusChange`(5) — UE Status Change. This feature defines the minimum registration expiration interval.
- `noAnsTimeout`(6) — Defines how long the eDVA rings a line before sending a "480 Temporarily Unavailable" response to the originator.
- `callerId`(7) — Sets the preferred presentation status (anonymous or public) on Caller ID for outgoing calls.
- `callerIdDisplay`(8) — Controls the display of Caller ID information for incoming calls.
- `callerIdBlocking`(9) — Controls per-call Caller ID Blocking. This feature is configured entirely through the digit map. No other provisioning is supported.
- `callerIdDelivery`(10) — Controls per-call Caller ID Delivery. This feature is configured entirely through the digit map. No other provisioning is supported.
- `callForwarding`(11) — Controls audible indicators for Call Forwarding (CFV). The network table controls the reminder dialtone when CFV is active; the user table controls Ring Reminder.
- `callWaiting`(12) — The Call Waiting feature is configured entirely through the digit map. Beyond enabling and disabling the feature, no other provisioning is supported.
- `callHold`(13) — The Call Hold feature is configured entirely through the digit map. Beyond enabling and disabling the feature, no other provisioning is supported.
- `callTransfer`(14) — Call Transfer is supported in the digit map. Setting the **pktcEUERSTCXIncomingOnly** object to `true`(1) allows Call Transfer only when the current call (first call leg) is an incoming call (this is the default setting for .TW loads).
- `threeWayCalling`(15) — 3-Way Calling is enabled in the digit map. and in the ARRIS-proprietary **arrisSipMib**.
- `doNotDisturb`(16) — Do Not Disturb is enabled in the digit map. The **pktcEUERSTDnDTable** configures confirmation tones for enabling and disabling the feature.
- `subscrProgPin`(17) — Not supported.
- `msgWaitIndicator`(18) — Message Waiting Indicator (MWI) subscription duration is configured through a network table. If the MWI Application Server (MWI AS) is different from the P-CSCF, the ARRIS-proprietary **arrisSipMib** can be used to specify the address of the MWI AS, and to control the behavior of MWI.
- `autoRecall`(19) — Auto Recall is supported through the digit map. The **pktcEUERSTAutoRclTable** is not supported in TS11.1.
- `autoCallback`(20) — Auto Callback is supported through the digit map. The **pktcEUERSTAutoCbTable** is not supported in TS11.1.
- `busyLineVerify`(21) — Not supported.

- `emergencySvc`(22) — Emergency Service timers are configured through the **pktcEUERSTNfEmSvcTable**. The eDVA fills emergency service-related **arrisSipMib** objects with provisioned or server-provided information.
- `scf`(23) — Selective Call Forwarding is not supported.
- `acr`(24) — Anonymous Call Rejection is configured entirely through the digit map. No other provisioning is supported.
- `solicitorBlocking`(25) — Solicitor Blocking is configured entirely through the digit map. No other provisioning is supported.
- `distinctAlerting`(26) — Distinctive Alerting is configured entirely through the digit map. No other provisioning is supported.
- `speedDialing`(27) — Not supported.
- `cot`(28) — Customer Originated Call Trace (COT) is configured entirely through the digit map. No other provisioning is supported. Use the CallP **cotdump** CLI command to display information captured by the eDVA in response to the subscriber using the COT dialing code.
- `heldMedia`(29) — Held Media. Enables an active bi-directional media stream to be held. No configuration is required or supported for this feature.
- `hotline`(31) — Hotline/Warmline. When the subscriber goes off-hook and does not dial a digit before the **pktcEUERSTHotlineOffhookTimer** timer expires, the eDVA automatically dials a number. By setting the timeout to 0, this feature acts as a hotline, automatically dialing the designated number as soon as the subscriber goes off-hook.

Note: The ARRIS-proprietary MIB objects **sipCfgWarmOrHotlineNumber**, **sipCfgWarmLineTimeout**, and **sipCfgPortWarmOrHotlineEnable** may also be used to provision hotline or warmline. If the **pktcEUERSTHotlineDestAddress** object is not provisioned, the eDVA uses the ARRIS SIP objects.

In addition to the PacketCable 2.0 features described above, Touchstone firmware provides the following ARRIS-specific features.

- Warmline/Hotline — when the subscriber goes off-hook and does not dial a digit before the **sipCfgPortWarmLineTimeout** timer expires, the eDVA automatically dials the number specified by the **sipCfgPortWarmOrHotlineNumber** object. By setting the timeout to 0, this feature acts as a hotline, automatically dialing the designated number as soon as the subscriber goes off-hook.

Note: TS11.1 does not support dialing codes for subscriber control over the Warmline/Hotline feature.

- 3-way Calling — An ARRIS extension to the PacketCable 2.0 digit map provides subscriber control over 3-way calling. Two actions, 3WC-ENABLE and 3WC-DISABLE, allow provisioning dialing codes for use with 3-way calling.

P-CSCF Dialing Features

In TS11.1, any dialing features not supported by the digit map are treated as P-CSCF dialing features. To enable a P-CSCF dialing feature, map the P-CSCF-defined dialing code to the `MAKE-CALL` action, passing the dialing code to the P-CSCF.

Example:

```
"*99" : MAKE-CALL ( "sip:" #0 =domain =dialString )
```

The P-CSCF must instruct the eDVA to play stutter tone or provide other confirmation, if necessary.

Action

Perform the following tasks as needed.

Basic Call Configuration

Follow these steps to configure a basic call feature.

1. Configure the user-side table, **pktcEUERSTBasicCallTable**, as follows:

   pktcEUERSTBCallPrefCodecList

   (optional) A comma-delimited list of the CODECs sent in the SDP. If left unconfigured, the eDVA offers G.711.

   Note: For backward-compatibility with earlier versions of Touchstone firmware, this object also accepts a semicolon-delimited list supported by the now-deprecated **sipCfgProvisionedCodecArray** object.

   pktcEUERSTBCallStatus

   Set to **`createAndGo`**(4).

2. (optional) Configure the network-side table, **pktcEUERSTNfBasicCallTable**, as follows. The index is *.oper.x*, where *oper* is the index of the entry in the Operator table. All the values in this table have reasonable defaults.

pktcEUERSTNfBCallByeDelay
The Bye delay, in seconds. When set to a non-zero value, a called party can hang up the phone then pick up before the timer expires without ending the call.

pktcEUERSTNfBCallPermSeqTone1
The URI specifying the first tone in the permanent sequence. See *Permanent Sequence* (page 93) for details and defaults.

pktcEUERSTNfBCallPermSeqTimer1
The duration of the first tone in the permanent sequence.

pktcEUERSTNfBCallPermSeqTone2
The URI specifying the second tone in the permanent sequence.

pktcEUERSTNfBCallPermSeqTimer2
The duration of the second tone in the permanent sequence.

pktcEUERSTNfBCallPermSeqTone3
The URI specifying the third tone in the permanent sequence.

pktcEUERSTNfBCallPermSeqTimer3
The duration of the third tone in the permanent sequence.

pktcEUERSTNfBCallPermSeqTone4
The URI specifying the last tone in the permanent sequence.

pktcEUERSTNfBCallPermSeqTimer4
The duration of the last tone in the permanent sequence.

pktcEUERSTNFBCallOrigModLongIntDig
The long interdigit timer, in seconds. If you configure the "L" timer in the digit map, do not set this object.

pktcEUERSTNfBCallStatus
Set to `createAndGo`(4). Setting only this object creates an entry with reasonable default values.

Configuring the Status Change Feature

In TS11.1, Status Change is used to periodically re-register with the network. One entry in the **pktcEUERSTUEActStatChgTable** is supported in TS11.1 and applies to the entire eDVA.

Follow these steps to configure the Status Change entry.

1. Set the **pktcEUERSTUEActStatChgRegExp** object to the desired registration expiration time, in seconds.
2. Set the **pktcEUERSTUEActStatChgStatus** object to `createAndGo`(4).

Configuring No Answer Timeout

Follow these steps to configure the No Answer timeout feature. This feature allows the eDVA to disconnect a call if the called party does not answer before the timer expires.

1. Set the **pktcEUERSTNoAnsTODuration** object to the desired No Answer timeout, in seconds.
2. Set the **pktcEUERSTNoAnsTOStatus** object to `createAndGo`(4).

Configuring Caller ID

Configuring Caller ID requires setup of several tables. Follow these steps to configure Caller ID.

1. Set up the user's presentation by creating an entry in the **pktcEUERSTCIDTable** with the following values:

   pktcEUERSTCIDPPS
   Set the presentation status to `public`(2).

   pktcEUERSTCIDStatus
   Set to `createAndGo`(4).

2. Set up the display feature by creating an entry in the **pktcEUERSTCIDDisTable** with the following values:

   pktcEUERSTCIDDisDefCountry
   Enter the country code data, to be stripped from the display information sent to the subscriber's CPE. For example, the US country code is `1`.

   pktcEUERSTCIDDisTimeAdj
   The adjustment, in minutes, from UTC.

   pktcEUERSTCIDDisDSTFlag
   Set to `1` to adjust for Daylight Savings Time, or `0` to ignore DST.

   pktcEUERSTCIDDisDSTInfo
   To use the above two objects for adjusting the time, set this to an empty string. Otherwise, set to a POSIX timezone string as defined in RFC 4833. If you set this object, the eDVA ignores the above two objects.

   pktcEUERSTCIDDisCIDCWActStat
   Set to `true`(1) to disable CID-CW.

   Note: You can disable CID-CW for the entire eDVA by setting bit `0x02000000` of the SIP Feature Switch **sipCfgSipFeatureSwitch**.

   pktcEUERSTCIDDisStatus
   Set to `createAndGo`(4).

Configuring Emergency Services

Follow these steps to configure emergency services. See *Emergency Calls* (page 94) for an overview of emergency call processing.

1. Add a rule to the digit map to associate a dial string with emergency calls. For example:

   `"911" : EMERGENCY-CALL ("sip:" "911" =domain =Emergencytg )`

2. To configure PacketCable 2.0 emergency service features, provision the following objects in the **pktcEUERSTNfEmSvcTable**. The index is *.oper.x*, where *oper* is the index of the entry in the Operator table.

   pktcEUERSTNfEmSvcNwHoldTimer
   (non-TW loads only) The emergency services network hold timer value, in minutes. Default: **45**.

   pktcEUERSTNfEmSvcHowlTimer
   The emergency services howler timer, in seconds. Default: **3**.

   pktcEUERSTNfEmSvcDSCPValMedia
   The DSCP value for network media (RTP) packets associated with emergency calls.

   pktcEUERSTNfEmSvcDSCPValSig
   The DSCP value for network signaling packets associated with emergency calls.

   pktcEUERSTNfEmSvcStatus
   Set to **`createAndGo`**(4) to write the entry and put it in service.

3. To change the URN used to identify incoming emergency calls, set the **sipCfgEmergencyServiceURN** object to the desired URN. The default setting is "`URN:service:sos`."

4. To disconnect active calls when originating or receiving an emergency call, set bit **`0x00001000`** in the **sipCfgSipFeatureSwitch** object.

Note: The eDVA does not notify the subscriber when disconnecting an active call in response to an incoming emergency call.

Configuring Distinctive Ring/Alert Tones

Setting Alert-Info strings for distinctive ringing and alert (Call Waiting) tones is identical for both PC20 and ARRIS SIP loads. See *Configuring Distinctive Ring/Alert Tones* (page 120) for details.

Configuring PacketCable 2.0 Digit Maps

The PacketCable *Residential SIP Telephony Feature Specification*, PKT-SP-RSTF-I08-110127, describes digit maps and provides an example map. The PacketCable 2.0 digit map specification is radically different from that defined in earlier specifications: earlier maps simply defined valid dialing sequences and left their interpretation to the Call Server (NCS) or vendor-defined methods (previous ARRIS SIP firmware versions use the **sipCfgDialFeat*** MIB objects); the new digit maps both define valid dialing sequences and associate actions with them.

General Digit Map Structure

A digit map consists of:

timer definitions

The following timers are defined by PacketCable 2.0 specifications. All are specified in seconds, and can be defined in 0.1 second increments.

- S (Short Interdigit Timer): when a subscriber's dialing string matches a pattern, but subsequent digits could match a longer pattern, pausing long enough for the "S" timer to expire selects the shorter pattern.

  Default: 4 seconds.
- Z (Long Duration Timer): used in patterns to indicate that the following key is held down for a minimum amount of time.

  Default: 2 seconds.

The following timers are supported in TS11.1 digit maps, but are deprecated. Use the appropriate MIB objects to set these timer values.

- T (Start Timer): the length of time allowed, after beginning to play dialtone, for the subscriber to dial the first digit. Typically, if this timer expires without the subscriber dialing anything, the UE plays reorder tone. Use the **pktcEUERSTNfBCallOrigDTTimer** object to specify this timer.

  Default: 16 seconds.
- L (Long Interdigit Timer): the allowable time between digits if the "S" timer is not specified. Use the **pktcEUERSTNFBCallOrigModLongIntDig** object to specify this timer.

  Default: 16 seconds.

symbol definitions

A *symbol* is a string constant, used to define common pattern or action strings and to make rules more human-readable. An example symbol definition is:

```
LocalNumber = [2-9]x{6}
```

map definitions

maps consist of a series of rules defined as:

```
pattern : action
```

Patterns are completely defined in the *Residential SIP Telephony Feature Specification*. The following list describes the more common components of a pattern.

- Numbers 0 through 9, and the * and # symbols, represent standard keypad keys. The characters "x" and "X" represent a single numeric keypress.
- S, T, L, and Z represent the defined timers. Of these, the Z timer must precede a keypress.

- An = sign followed by a name represents a defined symbol. Example: `=LocalNumber`
- Square brackets enclosing a series of keys, such as `[2345]`, match any single keypress in that series. A range of numeric keys may be represented using a hyphen, so `[2-5]` is equivalent to `[2345]`. The character “x” or “X” is equivalent to `[0-9]`. If the first character in the series is a caret (^), then the series matches a keypress not represented — for example, `[^01*#]` is equivalent to `[2-9]`.
- A keypress (or a series enclosed in square brackets) followed by a number or range in curly brackets, such as `x{6}`, matches when the specified number (or some number within the range) of matching keys are pressed. For example, `1x{2-4}#` matches `1xx#`, `1xxx#`, or `1xxxx#`.
- All or part of a pattern may be enclosed in parentheses to specify a *sub-pattern*. A sub-pattern may be used in an action, as described below.
- Two slashes (//) indicate that the rest of the line is a comment.

Actions, like patterns, are completely defined in the *Residential SIP Telephony Feature Specification*. An action consists of a command, followed by a parameter string enclosed in parentheses. Multiple actions can be specified, and are separated by a semicolon (;). TS11.1 supports the following commands in actions:

- MAKE-CALL (*uri*) — URI representing the specified destination (usually a phone, but could include dialing codes handled in the network)
- CID-DELIVER (*string*) — send the specified Caller ID information
- CID-SUPPRESS (*string*) — do not send Caller ID information
- CW-TOGGLE — toggle Call Waiting
- EMERGENCY-CALL (*uri*) — call the specified emergency number
- HOLD-ACTIVATE — hold the active call
- COT-ACTIVATE — perform a Customer-Originated Trace of the last call
- RECALL — play Recall tone
- REORDER — play Reorder tone
- RETURN (*string*) — specifies the actual value returned by the map
- SB-MAINT (*string*) — solicitor blocking
- SD-PROGRAM (*vsc*, *string*) — adds a number to the Speed Dial list
- 3WC-ENABLE — enables 3-way calling (ARRIS extension).
- 3WC-DISABLE — disables 3-way calling (ARRIS extension).
- USEMAP (=*map-name*) — applies the specified map to input received after matching this rule
- CNDB-TOGGLE — toggles the Caller ID privacy setting for the next outgoing call.
- ACR-ACTIVATE — activates rejection of incoming calls with blocked Caller ID information.
- ACR-DEACTIVATE — allows incoming calls with blocked Caller ID information.
- FEATURE-CHECK (*featureID*, [*failureURI*]) — if the feature *featureID* is not enabled, the eDVA plays the specified *failureURI* (or reorder tone if not specified).
- DND-ACTIVATE — activates Do Not Disturb (ARRIS extension).
- DND-DEACTIVATE — deactivates Do Not Disturb (ARRIS extension).

The following is a subset of the more common components of parameters to action commands.

- A string enclosed in quotes, such as `"@telco.example.net"`, is a literal string.

- An = sign followed by a name represents a defined symbol. Example: `=LocalNumber`
- A pound sign (#) followed by a number represents a sub-pattern within a matched pattern. For example, if the defined pattern was `([2-9]xx)(555(xxxx))` and the dial string matched was `6785551212`, then `#2` returns the value "`5551212`." The pattern `#0` represents the entire matched dial string.
- A pound sign, followed by a number and a "v" specifies a sub-pattern returned by a map.

A series of components are concatenated; for example,

```
"sip:" #0 "@example.net" =dialstring
```

constructs a single parameter string.

TS11.1 Compliance with PacketCable 2.0

TS11.1 generally conforms to PacketCable 2.0 specifications for digit maps, with the following exceptions:

- The Start Timer "T" is supported, with a default value of 16 seconds, and can be changed in the digit map. However, its behavior is hard-coded to play reorder tone, equivalent to defining the rule `"T" : REORDER` in the digit map. Configuring this timer in the digit map is deprecated; use the **pktcEUERSTNfBCallOrigDTTimer** object to set this timer.
- The Long Interdigit Timer "L" can be defined but is not supported in rules. However, if the Short Interdigit Timer "S" is not defined, the "L" timer definition is applied to rules where "S" is specified. Configuring this timer in the digit map is deprecated; use the **pktcEUERSTNFBCallOrigModLongIntDig** object to set this timer.
- The keys "A" through "D" are not supported.
- Symbols are supported, but the definition of a symbol must be a constant string. Thus, the following symbol definition is supported:

  ```
  localNumber = "[2-9]x{6}"
  ```

  but this is not:

  ```
  ldNumber = "1" =AreaCode =localNumber
  ```

- The following action commands are not supported:
  - AC-ACTIVATE, AC-DEACTIVATE
  - AR-ACTIVATE, AR-DEACTIVATE
  - CFV-PROGRAM, CFV-DEACTIVATE
  - DA-MAINT
  - DND-PROGRAM
  - SCF-PROGRAM
  - SPP-PROGRAM

Specifying a Digit Map

Follow these steps to specify a digit map.

1. Use a text editor to create a digit map. Note the restrictions above.

2. In the configuration file, set the **pktcEUERSTDMValue**.1 MIB object to the text of the digit map. Note that you will need to use the **LongSnmpMib** TLV to accommodate the size of the map.

Note: In TS11.1, a single digit map applies to all lines in the E-UE.

Example Digit Map

The following is an example digit map.

```
// Timer values
TIMER T=16 // Start timer. The length of time allowed to dial the
 // first digit from the time dial tone is applied
TIMER S=4 // Short interdigit timer. Used when critical timing should
 // be performed, such as when the dialed digits constitute
 // a complete address, but additional digits may
 // constitute a different complete address
TIMER L=16 // Long interdigit timer. The allowable time between digits
 // if the short interdigit timer has not been indicated in
 // the digit map.
TIMER Z=2.0 // Long duration timer. The duration a particular digit is
 // to be held in order to be detected.
 // Symbols
domain = "@tel.example.com"
areaCode = "303"
dialString = ";user=dialstring" // Just to shorten things
homeEmergencyNumber = "911"
localEmergencyNumber = "911" // alternate emergency number
lcltg = ";tgid=Local_trunk-group-id"
ldtg = ";tgid=Long_Distance_trunk-group-id"
intltg = ";tgid=International-trunk-group-id"
Emergencytg = ";tgid=911PSAP-trunk-group-id"
Opertg = ";tgid=Local_Operator-trunk-group-id"
ldcic = ";cic=0333"
intlcic = ";cic=8937"
TollFreecic = ";cic=0110"
// Maps
MAP MainTable = // This is where processing starts
 "T" : REORDER // Reorder Tone or Annc.
 "0S" : MAKE-CALL ("sip:0" =domain =Opertg)
 "0#" : MAKE-CALL ("sip:0" =domain =Opertg)
 "00" : MAKE-CALL ("sip:0" =ldcic =domain =ldtg)
 "(=Emergency)" : EMERGENCY-CALL ("sip:" "911" =domain =Emergencytg )
 // map N11 to routing number or reorder if not assigned
 "211" : MAKE-CALL ("sip:" "+13035551111" =domain =lcltg)
 "311" : MAKE-CALL ("sip:" "+13035552222" =domain =lcltg)
 "411" : MAKE-CALL ("sip:" "411" =domain =Opertg)
 "511" : REORDER // Reorder Tone
 "611" : MAKE-CALL ("sip:" "+13035552224" =domain =lcltg)
 "711" : MAKE-CALL ("sip:" "+18885552225" =TollFreecic =domain =lcltg)
 "811" : MAKE-CALL ("sip:" "+19035552226" =ldcic =domain =ldtg)
 "( =Speedcall)" : MAKE-CALL ( "sip:" #1v)
 "(=PhoneNumber)" : MAKE-CALL (#1v)
 "(=ImmediateVSCs)" : RETURN
 "(=DelayedVSCs)" : RETURN
 // Press # for 2 seconds & get recall dial tone
 "Z#" : RECALL; USEMAP(=MainTable)
 // Any other digit strings
 "(x{1-15})S" : REORDER
```

```
 "(x{1-15})#" : REORDER
MAP PhoneNumber =
 "(=LocalPhoneNumbers)" : RETURN ("+1" #1v =domain =lcltg)
 "0(=LocalPhoneNumbers)" : RETURN ("0" #1v =domain =Opertg)
 "1{0-1}([579]00[2-9]x{6})" : RETURN ("+1" #1 =domain =lcltg)
 // 500, 700, & 900 numbers
 "1{0-1} (=TollFreeNumbers)" : RETURN ("+1" #1v =TollFreecic =domain
=lcltg)
 "101(=DialAround)" : RETURN ( "sip:" #1v ";dai=no-presub" =domain =lcltg)
 "1(=WZ1InternationalNumbers)" : RETURN ("+1" #1v =intlcic =domain =intltg)
 "0(=WZ1InternationalNumbers)" : RETURN ("0" #1v =intlcic =domain =intltg
) "011(=InternationalNumbers)" : RETURN ("+" #1v =intlcic =domain =intltg)
 "01(=InternationalNumbers)" : RETURN ("01" #1v =intlcic =domain =intltg)
 "1{0-1} (=LDPhoneNumbers)" : RETURN ("+1" #1v =ldcic =domain =ldtg)
 "0(=LDPhoneNumbers)" : RETURN ("0" #1v =ldcic =domain =ldtg)
MAP Emergency = // Matches emergency dial
"(=localEmergencyNumber)" : RETURN
 "(=homeEmergencyNumber)" : RETURN
 "[01](=homeEmergencyNumber)" : RETURN
MAP Speedcall = // Matches Speed Call list (either one or two digit)
 // two-digit speed dialing
 "21S" : RETURN ( "sip:" "+19137654321" =ldcic =domain =ldtg)
 "22S" : RETURN ( "sip:" "+13037654321" =domain =lcltg)
 "23S" : RETURN ( "sip:" "+18007654321" =TollFreecic =domain =lcltg)
 "24S" : RETURN ( "sip:" "+529137654321" =intlcic =domain =intltg)
 "21#" : RETURN ( "sip:" "+19137654321" =ldcic =domain =ldtg)
 "22#" : RETURN ( "sip:" "+13037654321" =domain =lcltg)
 "23#" : RETURN ( "sip:" "+18007654321" =TollFreecic =domain =lcltg)
 "24#" : RETURN ( "sip:" "+529137654321" =intlcic =domain =intltg)
MAP LocalPhoneNumbers = // Matches local phone numbers
 "(=areaCode)(=Local7DigitNumbers)" : RETURN( =areaCode #2v )
 "( =Local7DigitNumbers )" : RETURN( =areaCode #1v )
MAP Local7DigitNumbers = // Matches local 7 digit numbers
 "(212x{4})" : RETURN( #1 )
 "(213x{4})" : RETURN( #1 )
 "(222[1-3]x{3})" : RETURN( #1 )
 "(2267x{3})" : RETURN( #1 )
 // could be 100+ entries for local digit map
MAP WZ1InternationalNumbers = // Matches international WZ1 numbers
 "(204x{7})" : RETURN( #1 )
 // 42 more entries for area codes of Canada & Caribbean
MAP TollFreeNumbers = // 800 and friends
 "(800[2-9]x{6})" : RETURN( #1 )
 "(866[2-9]x{6})" : RETURN( #1 )
 "(877[2-9]x{6})" : RETURN( #1 )
 "(888[2-9]x{6})" : RETURN( #1 )
MAP LDPhoneNumbers = // Matches Long Distance phone#
 // Local, 500, 700, 800, 900 & WZ1 International
 // eliminated by prior matches
 "([2-9]x{6})S" : RETURN( =areaCode #1 )
 "([2-9]x{6})#" : RETURN( =areaCode #1 )
 "([2-9]x{9})" : RETURN( #1 )
MAP InternationalNumbers = // Matches international non-WZ1 numbers
 "([2-9]x{1-14})S" : RETURN( #1 )
 "([2-9]x{1-14})#" : RETURN( #1 )
MAP DialAround = // Matches dial around phone#
 "0([2-9]x{2})(=daiPhoneNumbers)" : RETURN( #2v ";cic=0" #1 )
 // (3 digit CIC dialed by user)
 "1(x{4})(=daiPhoneNumbers)" : RETURN( #2v ";cic=1" #1 )
 // (4 digit CIC dialed by user)
MAP daiPhoneNumbers = // valid dai phone numbers
 "[5789]00" : REORDER
 "S" : RETURN
 "#" : RETURN
 "866" : REORDER
```

```
 "877" : REORDER
 "888" : REORDER
 "(=LocalPhoneNumbers)" : RETURN ("+1" #1)
 "0(=LocalPhoneNumbers)" : RETURN ("0" #1)
 "1{0-1} (=WZ1InternationalNumbers)" : RETURN ("+1" #1)
 "0(=WZ1InternationalNumbers)" : RETURN ("0" #1)
 "011(=InternationalNumbers)" : RETURN ("+" #1)
 "01(=InternationalNumbers)" : RETURN ("01" #1)
 "1{0-1}(=LDPhoneNumbers)" : RETURN ("+1" #1)
 "0(=LDPhoneNumbers)" : RETURN ("0" #1)
MAP ImmediateVSCs = // Matches and executes immediate VSCs.
 // Returns nothing.
 "*74" // (SD8)
 : RECALL; USEMAP (=SD8)
 "*75" // (SD30)
 : RECALL; USEMAP (=SD30)
 "*[78]7" // (ACR-ACTIVATE/ACR-DEACTIVATE)
 : MAKE-CALL ( "sip:" #0 =domain =dialString)
 "*9[01]" // (DND-ACTIVATE/DND-DEACTIVATE)
 : MAKE-CALL ( "sip:" #0 =domain =dialString)
 "*96" // (SB-MAINT)
 : SB-MAINT (#0) // solicitor blocking maintenance
 "*72" // (CFV-ACTIVATE)
 // Play Recall Dial Tone
 // Collect digits using subset digitmap
 // Prepend "*72." onto dialed address using SIP URI
 // The following is an example implementation:
 : RECALL; USEMAP(=ForwardingNumber)
 "*73" // (CFV-DEACTIVATE) (reuse *72 for this)
 : MAKE-CALL ( "sip:*72." =domain =dialString)
MAP DelayedVSCs = // Make some state change, then continue processing
dialing
 "*92" : HOLD-ACTIVATE; RECALL; USEMAP(=MainTable)
 "*67" : CID-SUPPRESS
 "*82" : CID-DELIVER
 "*70" : CW-TOGGLE; RECALL; USEMAP(=MainTable) // The CALL-WAITING action
must give the right tone.
MAP SD8 = // Program one-digit speed dial number
 "[2-9]" : SD-PROGRAM ( "sip:*74." #0 =domain =dialString)
 "[^2-9]" : REORDER
 "S" : REORDER
 "#" : REORDER
MAP SD30 = // Program two-digit speed dial number
 "[2-4]x" : SD-PROGRAM ( "sip:*75." #0 =domain =dialString)
 "[^2-4]" : REORDER
 "S" : REORDER
 "#" : REORDER
 "[2-4]S" : REORDER
MAP ForwardingNumber = // Just for programming CFV
 "(=PhoneNumber)" : MAKE-CALL ( "sip:*72." #1 =domain =dialString)
```

Provisioning PacketCable 2.0 Users

Use this procedure to set up users and associate them with credentials and lines.

Action

Follow these steps:

1. If using Certificate Bootstrapping, skip to step 3.
2. Configure an IMPI table entry in the eDVA configuration file. The index is usually **`1`**. Set the objects in this entry as follows:

   pktcEUEUsrIMPIIdType
   The identification type for the private identity. Only **`privateIdentity`**(4) is supported in TS11.1 and is the default.

   pktcEUEUsrIMPIId
   The user's login information used to authenticate the eDVA to the headend.

   pktcEUEUsrIMPICredsType
   The type of credentials used to confirm the private identity. Only **`password`**(3) is supported in TS11.1 and is the default.

   pktcEUEUsrIMPICredentials
   The user's password. Reading this object always returns an empty string.

   pktcEUEUsrIMPIRowStatus
   Set to **`createAndGo`**(4).

Note: In most cases, a single IMPI table entry is associated with all public users.

3. Configure an IMPU table entry in the eDVA configuration file. Set the objects in this entry as follows:

   pktcEUEUsrIMPUIdType
   (optional) The public user identification type. Both **`publicIdentity`**(3) (the default) and **`userName`**(6) are supported in TS11.1.

   pktcEUEUsrIMPUId
   The public user identification. The content of this object depends on the value of **pktcEUEUsrIMPUIdType**:

   - **`publicIdentity`**(3): *phonenumber@domain*
   - **`userName`**(6): *phonenumber* (Touchstone firmware adds the default domain to the phone number to create the full public identity)

   pktcEUEUsrIMPUIMPIIndexRef
   The index into the **pktcEUEUsrIMPITable**, specifying the IMPI entry associated with the public user.

   pktcEUEUsrIMPUDispInfo
   The display name in SIP messages, associated with Caller ID name delivery.

   pktcEUEUsrIMPUOpIndexRefs
   The index into the **pktcEUEDevOpTable**, specifying the operator entry associated with the public user. In TS11.1, this is usually **`1`** since only one operator entry is supported.

   pktcEUEUsrIMPUAdminStat
   (optional) Accept the default of **`active`**(1) to enable the line.

pktcEUEUsrIMPUAdditionalInfo

If only one public user is associated with a phone line, the eDVA automatically maps the first user to line 1 and the second user to line 2. For multiple phone numbers associated with a line, as is the case with a Teen Line service or an NPA or exchange transition, set this object as follows:

IEP# `line`**;OEP#** `line`

The *`line`* is the **ifIndex** for the line (usually **`9`** for line 1 and so on), and must be same for both IEP (Input EndPoint) and OEP (Output EndPoint).

pktcEUEUsrIMPURowStatus

Set to **`createAndGo`**(4).

4. (TS7.6 MSUP2 and newer) Configure the optional Certificate Bootstrapping feature, by modifying the configuration file, as follows:
   a. Verify that the CableLabs Service Provider Root certificate in the E-UE and the Certificate Bootstrapping server are identical.
   b. Set the **pktcEUECBEnable** object to **`true`**(1).
   c. Set the **pktcEUECBData** object to the HTTPS URL of the Certificate Bootstrapping configuration file. This is an XML file, and is downloaded using HTTP over TLS.
   d. Reset the eDVA to use the new provisioning.
   e. (optional) To verify operation, walk the **pktcEUEUsrIMPITable** to see the settings.

Provisioning PacketCable 2.0 Application Profiles

Use this procedure to provision the Application Profile. The profile collects feature sets to associate with a user.

Before you can provision the Application Profile, you must configure the features as described in *Provisioning PacketCable 2.0 Features* (page 133).

Indexing

The **pktcEUERSTAppProfileToFeatTable** uses a dual index for each entry: *`.profile.feature`*.

Action

Follow these steps to provision an Application Profile. The profile must be provisioned in the eDVA configuration file.

1. Set the **pktcEUERSTAppFeatID** to the feature type being specified. See *Provisioning PacketCable 2.0 Features* (page 133) for a list of valid feature types.
2. Set the **pktcEUERSTAppFeatIndexRef** object to the index of an entry in the **pktcEUERSTAppProfileToFeatTable**. This specifies the feature definition to use.

3. (optional) Set the **pktcEUERSTAppAdminStat** object to `active` (the default) to activate this feature, or `inactive` to disable this feature.
4. Set the **pktcEUERSTAppStatus** to `createAndGo`(4).

Provisioning PacketCable 2.0 Application Maps

Use this procedure to associate an application and an application profile with a user.

Prerequisites

Before provisioning an application map, perform the following procedures:

1. *Provisioning PacketCable 2.0 Features* (page 133)
2. *Provisioning PacketCable 2.0 Application Profiles* (page 147)
3. *Provisioning PacketCable 2.0 Users* (page 145)

Indexing

The **pktcEUEUsrAppMapTable** uses a dual index for each entry: *.user.map*, where *user* index is the index of the IMPU entry that this map is associated with; the *map* is the application index (usually `1` since TS11.1 supports only the PacketCable RST application).

Action

Follow these steps to provision an application map. You must add these objects to the eDVA configuration file.

1. Set the **pktcEUEUsrAppMapAppOrgID** object to the enterprise number of the organization defining the application: always `4491` for PacketCable.
2. Set the **pktcEUEUsrAppMapAppIdentifier** object to the application: always `1` for Residential SIP Telephony (RST).
3. Set the **pktcEUEUsrAppMapAppIndexRef** object to the index of the entry in the **pktcEUERSTAppProfileToFeatTable** corresponding to the profile associated with this map.
4. (optional) Set the **pktcEUEUsrAppMapAppAdminStat** object to `active` (default) to enable the application for this user, or `inactive` to disable the application.
5. Set the **pktcEUEUsrAppMapRowStatus** object to `createAndGo`(4).

Configuring SIP Failure Response Tones

Touchstone firmware complies with PacketCable 2.0 specifications for playing tones in response to various SIP failure messages. Use this procedure to override the default behavior.

Priority

The eDVA follows these steps to determine which tone to play when receiving a failure response.

1. If the (**sipCfgSipFeatureSwitch**) object has the `playBusyToneOnReject`(0x00000400) bit set, the eDVA always plays busy tone.
2. If the `playBusyToneOnReject`(0x00000400) bit is not set, and the P-CSCF specifies a tone in the Error-Info header of the failure response, the eDVA plays the specified tone.
3. If the **pktcEUERSTNfAncMapURI**.*code* object corresponding to the response code is set, the eDVA plays the configured tone.
4. The eDVA plays the default response tone as defined in the table below:

Response Code	Tone
486	Busy
487	Silence
600	Busy
Other (except 401 or 407)	Reorder

Action

Perform the following tasks as needed.

Playing Busy Tone for All Errors

To override PacketCable 2.0 behavior and play busy tone for all failure responses, follow these steps.

1. Set the `0x00000400` bit of the **sipCfgSipFeatureSwitch** in the eDVA configuration file.
2. Restart the eDVA to make the change take effect.

Configuring Individual Response Tones

Follow these steps to configure a tone to play for a specific failure response.

1. For each response code, set the **pktcEUERSTNfAncMapRspCode** object, either in an SNMP manager or in the eDVA configuration file. This object must be indexed by both the domain entry (TS11.1 allows only one domain per E-UE) and the response code; for example, **pktcEUERSTNfAncMapRspCode**.1.404, and contain the URI of the PacketCable-defined tone to play. For example:

```
{pktcEUERSTNfAncMapRspCode.1.404 "file:///PacketCableRST/bz"}
```

The following failure codes are supported:

- 404, 406, 408
- 480, 484, 486, 487
- 500, 503, 504
- 600, 603

Note: A URI sent in the Error-Info header of the failure response takes precedence over these settings.

2. If using an SNMP manager, set the **pktcEUERSTNfAncMapStatus** object, indexed by the same domain and response code, to **`createAndGo`**(4). Otherwise, restart the eDVA to make the change take effect.
3. To remove an entry, set the **pktcEUERSTNfAncMapStatus** object, indexed by the same domain and response code, to **`destroy`**(6).

Note: Attempting to remove the entry for response codes 486, 487, or 600 resets the entry to its default value.

Configuring MWI Support

Touchstone firmware provides improved control over both visual and audible MWI (Message Waiting Indication) indicators.

Note: The tasks in this procedure apply only to PacketCable 2.0 SIP deployments. ARRIS-proprietary legacy SIP deployments are not supported.

Action

Perform the following tasks as needed.

Clearing MWI Indicators

It may be necessary to clear MWI indicators in response to a subscriber trouble call. Follow these steps to clear the indicators.

1. Set the **sipCfgPortMWIClear**. line object to **`true`**(1). The line is the line number for which the indicators should be cleared; use **`1`** for line 1, and so on.

2. Repeat step 1 for each line that needs to be cleared.

Provisioning the MWI Subscription

By default, the eDVA subscribes to the P-CSCF MWI package (message-summary). Follow these steps to provision the eDVA to subscribe to a different server's MWI package.

1. Set the **sipCfgMWITargetAddrType** object to **`ipv4`**(1) or **`ipv6`**(2).
2. Set the **sipCfgMWITargetAddr** object to the IP address of the device that the eDVA should subscribe the MWI package to.
3. (optional) Set the **sipCfgMWITargetPort** object to the port number of the MWI service. The default is **`5060`**.
4. To recommend a subscription duration, set the **pktcEUERSTNfMWISubDuration** object to the recommended time before the subscription expires.

   Valid range: **`0`** to **`4000000`** seconds. Default: **`3600`**.

Voice Mail Subscription Watchdog

In this TS11.1 release, PC20 SIP loads support automatic re-subscription to MWI. The new **sipCfgMWISubscriptionCheckInterval** object specifies the interval, in minutes, that the eDVA checks the MWI subscription state. At each interval, if the line is operational and MWI is unsubscribed, it automatically restarts the subscription.

The default value of 0 disables this feature.

Operations

Operations encompass automatic monitoring of environment, detecting and determining faults and alerting admins.

Battery Management

Touchstone firmware provides a sophisticated management and monitoring scheme to maximize battery hold times and extend the useful life of backup batteries.

Initial Battery Charging

When the E-UE is powered up, whether for the first time or after replacing a battery, it begins a battery charging and testing sequence. See the *Touchstone Battery Reference Manual* for details about the charging and testing sequence.

Battery Telemetry

Touchstone firmware provides battery telemetry through the following management interfaces:

- LED display
- Web-based interface
- CLI
- SNMP

Power Failure Operation

When an E-UE with battery backup capabilities loses AC power, it immediately takes the following actions:

1. Shuts off some LEDs to conserve battery power.
2. Shuts down the data services running over all LAN ports after 30 seconds (default) of power loss. You can change the amount of time before the E-UE disables data services, by setting the **arrisMtaDevPwrSupplyDataShutdownTime** object.
3. After the shutdown timer expires, E-UEs disable bonded mode, switching to 1x1 unbonded (see see "*Advanced Power Management* (page 153) for details).

LED Changes

LED behavior changes under specific conditions, as described below.

LED Operation Changes During Battery Charging

The Battery LED only flashes when AC power is not present and the battery is low, or else when the battery needs to be replaced. The Battery LED does not flash during normal charging.

Installer Visual Indication of Foreign Loop Voltage

LEDs flash any time the line card transitions to the Line Card Over-current Protection State. This indicates foreign voltage is present on the loop. This condition occurs most often at installation time.

Battery Status Monitoring

Touchstone firmware provides battery status monitoring through SNMP. Battery MIB objects provide an estimate of remaining battery charge as a percentage of full charge, and in minutes remaining to depletion.

E-UEs have battery charger hardware that reports an accurate estimate of battery capacity within 5 minutes of initialization.

The "Power Supply Telemetry" alarms and logs provide a report of any status changes to the power system, including the battery and the charger.

Highest Charger Temperature Recording

Touchstone firmware provides a feature to record and report the historic high charger temperature:

arrisMtaDevPwrSupplyHighestTemperature
Provides the highest temperature, in degrees C, recorded by the battery charger.

arrisMtaDevPwrSupplyHighestTemperatureTime
Provides the date and time that the E-UE recorded the high temperature.

arrisMtaDevPwrSupplyHighestTemperatureClear
Resets the highest temperature and time data.

Advanced Power Management

Touchstone E-UEs with battery backup support enhanced low-power features to maximize battery hold-up time.

When the E-UE loses AC power, it starts the data shutdown timer with a default value of 30 seconds. If the timer expires before AC power is restored, the device:

- powers down Ethernet and wireless (if equipped) interfaces.
- switches to 1x1 unbonded mode after all lines go on-hook, informing the CMTS using a CM-STATUS message.

Once AC power is restored, the E-UE:

- activates data interfaces (as provisioned)
- attempts to restore its original bonding mode
- clears codeword counters as a side effect

CMTS Considerations

For best results, the CMTS should fully support CM-STATUS partial service messages. If the CMTS does not support CM-STATUS, add the **arrisCmDoc30SetupPowerSaveMode** object to the CM configuration file with a value of **`reinitmac`**(1). This causes the CM to re-register in unbonded (1x1) mode upon loss of AC power, and re-register to bonded mode when AC power is restored.

Note: Re-registration may take up to 90 seconds.

When the CMTS is upgraded, change the value to the default **`partialservice`**(0), or remove the object from the CM configuration file.

About IPv6 Support

IPv6 allows cable operators to expand their IP address ranges for cable modems, allowing reclamation of traditional IPv4 addresses for use with eDVAs. TS11.1 supports the following IPv6-related protocols:

- DHCPv6
- DNSv6
- TFTPv6
- TODv6
- SNMPv6
- TELNETv6
- HTTPv6
- SSHv6

Note: Touchstone firmware does not support Early Authentication and Encryption (EAE) functionality.

Supported Hardware

Touchstone firmware supports single- and dual-stack CPE traffic on all supported Touchstone devices as follows:

- supports single stack IPv4 traffic to and from CPEs.
- supports single stack IPv6 traffic to and from CPEs.
- supports IPv4 and IPv6 traffic simultaneously to and from CPEs.

Touchstone firmware supports IPv6 operation on all supported Data Gateway and Telephony Gateway products.

IPv6 Multicast Support

TS11.1 supports multicast forwarding as described below.

Before Registration

The CM:

- filters multicast traffic not addressed to the IPv6 Link Local Scope All Nodes Address or the Solicited Node Addresses.
- learns the pre-registration DSID from the MDD message.
- forwards pre-registration multicast traffic to its IP stack based on the pre-registration DSID as defined in the "DSID based Filtering and Forwarding by a Cable Modem" section of CM-SP-MULPIv3.0-I12-100115.

During Registration

The CM reports the following IPv6 multicast capabilities in DHCP Option 125:

Sub-Option	Description	Value
32	Multicast DSID support	24
33	Multicast DSID forwarding (GMAC promiscuous)	2
34	Frame Control Type forwarding	1

After Registration

The CM:

- stops forwarding multicast traffic labeled with the Pre-Registration DSID, after receiving the REG-RSP message.
- forwards multicast traffic based on the DSIDs and Group MAC Addresses received in the REG-RSP message, as specified by the "DSID based Filtering and Forwarding by a Cable Modem" section of CM-SP-MULPIv3.0-I12-100115.
- filters multicast traffic based on DSIDs and Group MAC Addresses received in the REG-RSP message. The CMTS always includes the IPv6 Link Local Scope All Nodes Address, and the CM's Solicited Node Addresses, in the REG-RSP message.

- supports Multicast DSID Encodings (TLV-50.4) described in the ``Multicast Encodings'' section of CM-SP-MULPIv3.0-I12-100115, as well as Security Association Encodings (TLV-51) described in the ``Security Association Encoding'' section of CM-SP-MULPIv3.0-I12-100115, received in the REG-RSP message.
- uses the Dynamic Bonding Change mechanism defined in CM-SP-MULPIv3.0-I12-100115, to maintain and learn new DSID values and Security Associations for IPv6 multicast.

The CM also forwards Neighbor Discovery packets sent to:

- The CPEs' Solicited Node multicast IPv6 addresses
- The All Nodes multicast address onto CPE ports

Neighbor Discovery forwarding can be controlled by configuring filters to allow or prohibit forwarding.

IPv6 Management

Managing Touchstone E-UEs using IPv6 addressing in the CM, and IPv4 addressing in the eDVA, require SNMP management software capable of handing both IPv4 & IPv6, or separate tools for CM and eDVAs.

Filtering IPv6 Traffic

TS11.1 supports the following filtering-related features:

- MLD Snooping
- Upstream Drop Classifiers

Each of these features is described below.

MLD Snooping

MLD Snooping allows CPE devices using either MLDv1 or MLDv2 to acquire IPv6 addresses, and to register and receive multicast IPv6 downstream traffic. Unless filters specifically prohibit it, MLD Snooping allows all multicast IPv4 and IPv6 traffic to pass.

Note 1: BPI treats IPv6 multicast traffic like normal traffic (IPv4 multicast traffic is treated as multicast).

Note 2: In TS11.1, MLDv2 Source Filtering is ignored.

When the CM receives a DOCSIS Reinit-MAC command, it clears the MLD database and downstream IPv6 multicast filtering records.

Upstream Drop Classifiers

An Upstream Drop Classifier is a Classifier created by the CM to filter upstream traffic. The CM performs IP protocol filtering using either Upstream Drop Classifiers or IP filters. Traffic may also be filtered from the WAN to the LAN, set by the user.

If a packet matches the specified packet-matching criteria of an Upstream Drop Classifier, it is dropped. Unlike QoS Classifiers, Upstream Drop Classifiers do not refer to a Service Flow.

TCP/UDP Packet Classification Encodings are defined for IPv4 or IPv6 and may be present in a Service Flow Classifier of either type. If those classifiers are present in combination with IPv6 classifier encodings, they apply to the IPv6 classifiers.

The CM reports the number of Upstream Drop Classifiers supported, using the Upstream Drop Classification Support capability (TLV 5.38) in the Registration Request message.

Coexistence

TS11.1 supports SNMPv3 co-existence for both the ARRIS version of coexistence and coexistence using the coexistence MIBs.

The following is an example provisioning file fragment for setting up co-existence.

```
SnmpMib = snmpCommunityStatus.rocablelabs createAndGo
SnmpMib = snmpCommunityName.rocablelabs "ro_cm"
SnmpMib = snmpCommunitySecurityName.rocablelabs "rotesting1"
SnmpMib = snmpCommunityStorageType.rocablelabs volatile
SnmpMib = snmpCommunityStatus.rwcablelabs createAndGo
SnmpMib = snmpCommunityName.rwcablelabs "rw_cm"
SnmpMib = snmpCommunitySecurityName.rwcablelabs "rwtesting1"
SnmpMib = snmpCommunityStorageType.rwcablelabs volatile
SnmpMib = vacmSecurityToGroupStatus.1 rotesting1 createAndGo
SnmpMib = vacmGroupName.1 rotesting1 "rotesting2"
SnmpMib = vacmSecurityToGroupStorageType.1 rotesting1 volatile
SnmpMib = vacmSecurityToGroupStatus.1 rwtesting1 createAndGo
SnmpMib = vacmGroupName.1 rwtesting1 "rwtesting2"
SnmpMib = vacmSecurityToGroupStorageType.1 rwtesting1 volatile
SnmpMib = vacmSecurityToGroupStatus.2 rotesting1 createAndGo
SnmpMib = vacmGroupName.2 rotesting1 "rotesting2"
SnmpMib = vacmSecurityToGroupStorageType.2 rotesting1 volatile
SnmpMib = vacmSecurityToGroupStatus.2 rwtesting1 createAndGo
SnmpMib = vacmGroupName.2 rwtesting1 "rwtesting2"
SnmpMib = vacmSecurityToGroupStorageType.2 rwtesting1 volatile
SnmpMib = vacmAccessStatus.rotesting2 1 1 createAndGo
SnmpMib = vacmAccessContextMatch.rotesting2 1 1 exact
SnmpMib = vacmAccessReadViewName.rotesting2 1 1 "docsisManagerView"
SnmpMib = vacmAccessStorageType.rotesting2 1 1 volatile
SnmpMib = vacmAccessStatus.rwtesting2 1 1 createAndGo
SnmpMib = vacmAccessContextMatch.rwtesting2 1 1 exact
SnmpMib = vacmAccessReadViewName.rwtesting2 1 1 "docsisManagerView"
SnmpMib = vacmAccessWriteViewName.rwtesting2 1 1 "docsisManagerView"
SnmpMib = vacmAccessStorageType.rwtesting2 1 1 volatile
SnmpMib = vacmAccessStatus.rotesting2 2 1 createAndGo
SnmpMib = vacmAccessContextMatch.rotesting2 2 1 exact
SnmpMib = vacmAccessReadViewName.rotesting2 2 1 "docsisManagerView"
SnmpMib = vacmAccessStorageType.rotesting2 2 1 volatile
SnmpMib = vacmAccessStatus.rwtesting2 2 1 createAndGo
SnmpMib = vacmAccessContextMatch.rwtesting2 2 1 exact
SnmpMib = vacmAccessReadViewName.rwtesting2 2 1 "docsisManagerView"
SnmpMib = vacmAccessWriteViewName.rwtesting2 2 1 "docsisManagerView"
SnmpMib = vacmAccessStorageType.rwtesting2 2 1 volatile
SNMPV3Kickstart =
  SNMPV3SecurityName = "docsisManager"
  SNMPV3PublicNumber = hexstr: C1.FC.52.36.97.06.C0.22.99.61.D5.CA.C5.
4F.C6.68.51.71.A8.DC.69.AB.EB.D6.21.AC.AC.1D.FC.A6.0A.3A.8E.77.B5.15.AB.
AC.60.7C.5F.EB.AF.5F.86.B8.3F.2B.A1.DB.3D.ED.51.E2.EB.5D.E0.6A.EB.2D.AE.
```

```
E3.A4.DA.AC.DA.30.42.DC.A2.3C.5B.FE.65.83.B8.B8.9E.48.02.FB.70.A5.E9.97.
0C.95.9F.96.44.B4.BA.B4.2C.71.97.D1.1A.96.99.C9.4F.9C.53.3F.00.24.3E.1A.
12.AB.23.CF.DB.05.6C.97.62.4B.B2.A3.FC.7D.91.F4.90.C7.7C
```

DHCPv6 MIB Objects

The following MIB objects indicate the DHCP mode (v4 or v6) and assigned IP address. The address type for each object is one of the following:

unknown(1):
: The address type is unknown

ipv4(1):
: DHCPv4

ipv6(2):
: DHCPv6

ipv4z(3):
: IPv4 non-global address with a zone index

ipv6z(4):
: IPv6 non-global address with a zone index

dns(16):
: DNS domain name

The objects are:

docsDevServerDhcpAddressType
: The IP address type of the assigned address, or **unknown**(0) if the IP address was statically assigned.

docsDevServerDhcpAddress
: The IP address assigned to the CM, or an empty string if the IP address was statically assigned.

The following MIB objects indicate the IP address and type of the assigned time and TFTP servers:

docsDevServerTimeAddressType
: The IP address type for the time server.

docsDevServerTimeAddress
: The IP address of the time server.

docsDevServerConfigTftpAddressType
: The IP address type of the TFTP server.

docsDevServerConfigTftpAddress
: The IP address of the TFTP server.

The Syslog server address must be configured manually, either through an SNMP manager or the configuration file. In either case, the following two objects must be set in the order shown:

docsDevEvSyslogAddressType
The IP address type; either **1** (IPv4) or **2** (IPv6).

docsDevEvSyslogAddress
The IP address of the Syslog server, or a blank string to disable Syslog transmission.

The following read-only MIB objects are part of **arrisCmDoc30DhcpCmParameters**, and provide DHCP information.

arrisCmDoc30DhcpCmIpAddrType
The type of the currently leased IP address.

arrisCmDoc30DhcpCmIpAddr
The currently leased IP address.

arrisCmDoc30DhcpCmSubNetMask
The current IP subnet mask in use.

arrisCmDoc30DhcpCmGatewayIpAddr
The current IP gateway address in use.

arrisCmDoc30DhcpCmConfigFile
The CM configuration file name.

SNMP Access

The CM expects SNMP access to use the same IP mode (IPv4 or IPv6) that it is provisioned with.

Event Reporting

TS11.1 supports PacketCable 1.0 event reporting functionality and a number of proprietary ARRIS events. The following enterprise numbers may appear in events generated by Touchstone E-UEs:

Number	Source
4115	ARRIS
4491	PacketCable

Collecting Events

Events may be collected by:

- The **docsDevEventTable** (part of the DOCS-CABLE-DEVICE-MIB) in the E-UE keeps CM events until the E-UE is rebooted or powered down.

- The **pktcDevEventTable** (part of the PKTC-EVENT-MIB) in the E-UE keeps eDVA events until the E-UE is rebooted or powered down. The default configuration stores events only in the local event tables.
- A specified Syslog server (specify a Syslog server using the **pktcDevEvSyslogAddress** object).
- SNMP servers (the E-UE sends the events in a **pktcDevEventNotify** message). See *Event Formats* (page 160) below for details.

Event Formats

TS11.1 firmware provides log messages for both the cable modem and eDVA sections of the Touchstone E-UE.

Cable Modem Log Format

Cable modem logs require the DOCS-CABLE-DEVICE-MIB. Cable modem log messages consist of the following information:

- EventIndex—Provides relative ordering of the objects in the event log. The value of this object always increases except when:
  - the log is reset using the **docsDevEvControl** object
  - the device reboots and does not implement non-volatile storage for this log
  - the value reaches 2^{31} (the index is a 32-bit counter that rolls over to zero at this limit)

  The next value for all the above cases is 1.
- EventFirstTime—The time that this entry was created.
- EventLastTime—If multiple events are reported via the same entry, the time that the last event for this entry occurred, otherwise this should have the same value as EventFirstTime.
- EventCounts—The number of consecutive event instances reported by this entry. This starts at 1 with the creation of this row and increments by 1 for each subsequent duplicate event.
- EventLevel—The priority level of this event as defined by the vendor. These are ordered from most serious (emergency) to least serious (debug).
- EventId—For this product, uniquely identifies the type of event that is reported by this entry.
- EventText—Provides a human-readable description of the event, including all relevant context.

eDVA Log Format

eDVA logs function within the context of the PKTC-EVENT-MIB. eDVA log messages consist of the following information:

Event Index

Provides relative ordering of the objects in the event log. This also serves as a indicator of event sequence. The object value always increases except when:

- the log is reset using the **pktcDevEvControl** object
- the device reboots and does not implement non-volatile storage for this log
- the value reaches 2^{31} (the index is a 32-bit counter that rolls over to zero at this limit)

The next entry for all the above cases is 1.

Event Time
: Provides a human-readable description of the time at which the event occurred.

Event Level
: The priority level of this event as defined by the vendor.

Event Enterprise number
: The IANA enterprise number: **4115** for ARRIS events, **4491** for PacketCable events.

Event ID
: ID for a specific event to which the priority and display string are matched. Event IDs are vendor specific.

Event Text
: The text message associated with the event. Corresponds to the **pktcDevEvProgrammableId** or **pktcDevEvFixedId** MIB objects.

Mac Address
: Provides the MAC address of the device generating the event.

FQDN/Endpoint ID
: The FQDN or IP address of the device, with an endpoint identifier. If the event is not specific to an endpoint, the identifier is simply the FQDN or IP address. If the event is specific to an endpoint, the format is `AALN/`*`X`*`:`*`FQDN/IP address`*. If the event is battery related, the format is *`FQDN/IP address`*`:BATT_PORT:1`.

Note: If the **arrisMtaDevEventHideMacFQDNandIPAddress** object is set to **`enable`**(1), the endpoint FQDN and IP address are masked with asterisk (*) characters as described below.

Hiding FQDN and IP Information in the Event Log

The Event Log web page, described in see "*Using the Web-based Troubleshooting Interface* (page 289), displays the FQDN (or IP address) of the eDVA for eDVA-related events in the **Endpoint Name** field. TS11.1 firmware can hide this information from basic users for security purposes.

Note: Users with access to the Advanced pages (using the Password of the Day) can always see FQDN and IP address information.

Follow these steps to enable or suppress FQDN and IP information in the Event Log.

1. To suppress FQDN and IP information, set the **arrisMtaDevEventHideMacFQDNandIPAddress** MIB object to `enabled`(1) in the CM or eDVA configuration file and reboot the eDVA.

   For basic users, the Event Log page replaces FQDN and IP information by asterisks:

PACKETCABLE(MTA) Events

Date Time	Event Level	Enterprise	Event ID	Description	MAC Address	Endpoint Name
1/1/1970 0:00	5	4115	14	Power Supply Telemetry Log - BATTERY TEST SCHEDULED	00.00.CA.CB.24.C7	
1/1/1970 0:00	5	4115	14	Power Supply Telemetry Log - BATTERY MISSING	00.00.CA.CB.24.C7	
9/6/2006 13:30	5	4115	16	MTA TFTP: Successful	00.00.CA.CB.24.C7	************
9/6/2006 13:30	5	4115	14	Power Supply Telemetry Log - BATTERY TEST SCHEDULED	00.00.CA.CB.24.C7	************
9/6/2006 13:30	5	4115	14	Power Supply Telemetry Log - BATTERY MISSING	00.00.CA.CB.24.C7	************
9/6/2006 13:30	5	4115	7	Voice Line Loop Current Change to High, Line Number = 1	00.00.CA.CB.24.C7	AALN/1:************
9/6/2006 13:30	5	4115	7	Voice Line Loop Current Change to High, Line Number = 2	00.00.CA.CB.24.C7	AALN/2:************
9/6/2006 13:30	5	4115	26	MTA PROV: Successful!	00.00.CA.CB.24.C7	************
9/6/2006 13:30	5	4115	3	Voice Line State Change, Line Number = 1, Prev State = OOS_NR_UNPROV, New State = OOS_NR	00.00.CA.CB.24.C7	AALN/1:************
9/6/2006 13:30	5	4115	3	Voice Line State Change, Line Number = 2, Prev State = OOS_NR_UNPROV, New State = OOS_NR	00.00.CA.CB.24.C7	AALN/2:************

Note: Users that have entered the correct Password of the Day can see FQDN and IP information on the Advanced Event Log webpage regardless of the setting of the **arrisMtaDevEventHideMacFQDNandIPAddress** object.

2. To display FQDN and IP information to all users, set the **arrisMtaDevEventHideMacFQDNandIPAddress** object to `disabled`(0) in the CM or eDVA configuration file and reboot the eDVA. This is the default setting.

   The Event Log page displays FQDN and IP information to all users:

PACKETCABLE(MTA) Events

Date Time	Event Level	Enterprise	Event ID	Description	MAC Address	Endpoint Name
1/1/1970 0:00	5	4115	14	Power Supply Telemetry Log - BATTERY TEST SCHEDULED	00.00.CA.CB.24.C7	
1/1/1970 0:00	5	4115	14	Power Supply Telemetry Log - BATTERY MISSING	00.00.CA.CB.24.C7	
8/30/2006 17:15	5	4115	16	MTA TFTP: Successful	00.00.CA.CB.24.C7	mta135.dev44
8/30/2006 17:15	5	4115	14	Power Supply Telemetry Log - BATTERY TEST SCHEDULED	00.00.CA.CB.24.C7	mta135.dev44
8/30/2006 17:15	5	4115	14	Power Supply Telemetry Log - BATTERY MISSING	00.00.CA.CB.24.C7	mta135.dev44
8/30/2006 17:15	5	4115	7	Voice Line Loop Current Change to High, Line Number = 1	00.00.CA.CB.24.C7	AALN/1:mta135.dev44
8/30/2006 17:15	5	4115	7	Voice Line Loop Current Change to High, Line Number = 2	00.00.CA.CB.24.C7	AALN/2:mta135.dev44
8/30/2006 17:15	5	4115	26	MTA PROV: Successful!	00.00.CA.CB.24.C7	mta135.dev44
8/30/2006 17:15	5	4115	3	Voice Line State Change, Line Number = 1, Prev State = OOS_NR_UNPROV, New State = OOS_NR	00.00.CA.CB.24.C7	AALN/1:mta135.dev44
8/30/2006 17:15	5	4115	3	Voice Line State Change, Line Number = 2, Prev State = OOS_NR_UNPROV, New State = OOS_NR	00.00.CA.CB.24.C7	AALN/2:mta135.dev44

References

The following CableLabs® specifications define the PacketCable 1.0 event reporting mechanism:

PKT-SP-MEM1.5-I03-070412

PacketCable Management Event Mechanism

PKT-SP-EVEMIB1.5-I02-050812
PacketCable Management Event MIB Specification

PKT-TR-MEMEVENT-ID-V01-0000929
PacketCable Management Event Identifiers

Event Summary

Touchstone firmware generates events (alarms and log messages) from both the CM and eDVA components. This manual documents only ARRIS-specific events; for DOCSIS CM events, see the *DOCSIS 3.0 Operations Support System Interface Specification*, CM-SP-OSSIv3.0-I15-100115.

Event Handling

Events may be collected at:

- The **docsDevEventTable** (part of the DOCS-CABLE-DEVICE-MIB) in the E-UE keeps CM logs until the E-UE is rebooted or powered down.
- The **pktcDevEventTable** (part of the PKTC-EVENT-MIB) in the E-UE keeps eDVA logs until the E-UE is rebooted or powered down. The default configuration stores logs only in the local event tables.
- A specified Syslog server (specify a Syslog server using the **pktcDevEvSyslogAddress** object).
- SNMP servers (the E-UE sends the log in a **pktcDevEventNotify** message).

eDVA States

The following is a list of valid eDVA states and their meanings. The **ppSurvMtaMaintState** object contains the current state.

In Service, Normal
The eDVA is operating normally.

In Service, Trouble
The eDVA is providing service, but a problem has been detected (typically loss of AC power).

Out of Service
The eDVA is out of service due to a detected problem. Both voice and data services may be affected.

eDVA Line States

The following is a list of valid eDVA line states and their meanings. The value of the **ppSurvPortMaintState** object provides the current line state.

In Service, Normal
The line is operating normally.

In Service, Trouble, Family Equipment Failure
A problem with the eDVA subsystem is preventing the line from functioning properly. All lines in this state may indicate that the eDVA failed to download its firmware image.

In Service, Trouble, Test Failed
The line failed diagnostics. The *Voice Line Diag Failed* (page 167) log provides more details.

In Service, Trouble, Diagnostics
The line is running diagnostics; if no problems are discovered, the line will be returned to service when diagnostics are finished.

In Service, Trouble, Line Card Protection
The line card is in an overcurrent protection state. This state usually indicates either a short between tip and ring, or a foreign voltage being applied to tip and ring.

Out of Service, Normal, Unprovisioned
The line is not provisioned but has no known problems.

Out of Service, Normal
The line card is out of service and provisioned, but has no known problems.

Out of Service, Trouble
The line is out of service due to a detected problem.

Out of Service, Trouble, Diagnostics
The line is out of service, and is running diagnostics.

E-UE Battery States

The following is a list of valid battery states for E-UEs with battery backup. The value of the **arrisMtaDevBatteryOperState** object provides the current battery state.

`normal`(12)
The E-UE is operating on AC power. The battery is charged and in good condition.

`acFail`(11)
The E-UE is operating on battery power.

`batteryLow`(7)
The E-UE is operating on AC power, but during a power outage has drawn down the battery to the capacity indicated by the **arrisMtaDevPwrSupplyLowBatteryThresh** object.

`batteryLow-replaceBattery`(5)
The E-UE is operating on AC power. However, in addition to the Battery Low condition described above, the battery has deteriorated and should be replaced.

`shutdownWarning`(2)
: The E-UE has nearly exhausted its battery power, and will lose power if AC power is not restored within a few minutes.

`batteryMissing`(8)
: The battery has been removed or has failed in such a way to appear to be removed.

`replaceBattery`(10)
: The E-UE is operating on AC power. However, the battery has deteriorated and should be replaced.

`unavailable`(0)
: The E-UE does not support battery telemetry.

`invalid`(1)
: Indicates a possible problem with the E-UE or the battery system.

`batteryReversedShorted`(3)
: The battery has either been installed backwards or the terminals have been shorted.

`batteryLow-acFail`(6)
: The E-UE is operating from battery power, and has entered the "Battery Low" condition described above.

`acFail-replacebattery`(9)
: The E-UE is operating from battery power. In addition, the battery has deteriorated as described in "Battery Replace" above and should be replaced.

`batteryLow-replaceBattery-acFail`(4)
: The E-UE is operating from battery power, and has entered the "Battery Low" condition described above. In addition, the battery has deteriorated as described in "Battery Replace" above and should be replaced.

`testInProgress`(13)
: The E-UE is testing the battery and charger system.

`chargerFailure`(14)
: The E-UE has twice failed (initial attempt and one retry) to download charger firmware. This indicates a hardware problem with the E-UE. The battery charger is disabled and backup battery power is not available.

eDVA Event Summary

The following is a list of ARRIS and PacketCable eDVA events that the TS11.1 firmware can generate. See the section describing the event for further details.

Ent.	ID	Severity	Log Text
4115	1	information(5)	Voice Line Diag Failed
4115	2	information(5)	Voice Line Diag Passed
4115	3	information(5)	Voice Line State Change

Ent.	ID	Severity	Log Text
4115	1	information(5)	Voice Line Diag Failed
4115	4	information(5)	Voice Line Provisioning Complete
4115	6	information(5)	Voice Line Protection State Change
4115	10	information(5)	State Changed
4115	14	information(5)	MTA TFTP: Failed
4115	16	information(5)	MTA TFTP: Successful
4115	26	information(5)	MTA PROV: Successful!
4115	31	information(5)	Loop Voltage Management: Policy Missing
4115	32	information(5)	Loop Voltage Management: Bad Key
4115	33	information(5)	Loop Voltage Management: Policy Out of Range
4115	34	information(5)	Loop Voltage Management: Policy Change
4115	35	information(5)	Loop Voltage Management: Policy 3 Timer Out of Range; Default Value Used
4115	37	information(5)	Call Stats
4115	39	information(5)	Last NCS Message Received
4115	65519	information(5)	Power Supply Telemetry Alarm - Battery Missing
4115	65520	information(5)	Power Supply Telemetry Alarm - Battery Low
4115	65521	information(5)	Power Supply Telemetry Alarm - Replace Battery
4115	65523	information(5)	SIP General Failure
4115	65524	information(5)	SIP Network Failure
4115	65526	information(5)	SIP Authentication Failure
4115	65527	information(5)	SIP Registration Timeout
4115	65528	information(5)	SIP Proxy Loss of Communications
4115	65533	major(2)	Voice Line Failure
4115	46	information(5)	MTA DHCP RENEW: Lease Renewal delay; Voice line offhook
4115	47	information(5)	MTA DHCP REBIND: Lease Renewal delay; Voice line offhook
4115	65529	major(2)	Power Supply Telemetry Alarm
4115	2417164301	information(5)	SSH LOGIN ACCEPTED
4115	2417164302	information(5)	SSH LOGIN REJECTED
4115	2417164303	information(5)	SSH LOGIN REJECTED - MAX ATTEMPTS
4115	2417164296	information(5)	Touchstone SW Upgrade Failed Before Download Attempt
4115	2417164297	information(5)	Touchstone SW Upgrade Failed
4115	2417164298	information(5)	Touchstone SW Upgrade Successful
4115	2417164299	minor(3)	Touchstone SW Upgrade Aborted due to Battery AC-FAIL Condition
4115	2417164304	minor(3)	Touchstone SW Upgrade Aborted due to Call in Progress

Ent.	ID	Severity	Log Text
4115	1	information(5)	Voice Line Diag Failed
4115	2417164305	information(5)	Touchstone SW Upgrade Reboot Delayed due to Call in Progress
4115	2417164308	information(5)	Gateway has reset
4115	2417164309	information(5)	Unit has been restored to factory defaults
4491	65528	minor(3)	Battery Not Low
4491	65529	minor(3)	Battery Low
4491	65530	minor(3)	Battery Present
4491	65531	minor(3)	Battery Missing
4491	65532	minor(3)	Battery Good
4491	65533	minor(3)	Replace Battery
4491	65534	minor(3)	AC Restored
4491	65535	minor(3)	AC Fail

ARRIS Events

The following are ARRIS eDVA-related events. ARRIS events use the enterprise number **4115**.

Voice Line Diag Failed

The eDVA has failed manual diagnostics for the specified line.

Format:

Voice Line Diag Failed, Line Number = *line*, Failure Reason = *reason*

Fields:

The fields are as follows:

- *line*—The line number that failed diagnostics. The first line is line 1.
- *reason*—one of the following:
  - Line is Unprovisioned
  - Invalid State to Init Diags
  - Power/Clock Failure
  - SLAC Revision Failure
  - MPI Failure
  - PCM Failure
  - Standby Hook Failure
  - Active Hook Failure
  - VF Failure
  - Ringing Failure

Action:

Use the reason code to determine the course of action as follows:

- Line is Unprovisioned—Provision the line and re-try the diagnostics.
- Invalid State to Init Diags—Set the line state to **oos** and re-try the diagnostics.
- others—Reset the eDVA and re-try diagnostics. If the problem persists, replace the E-UE.

Voice Line Diag Passed

Indicates that the eDVA has successfully completed manual diagnostics on the specified line.

Format:

Voice Line Diag Passed, Line Number = *line*

Fields:

line indicates the line that passed diagnostics. The first line is line 1.

Action:

None.

Voice Line State Change

The eDVA received an operator-requested state change for the specified line.

Format:

Voice Line State Change, Line Number = *line*, Prev State = *old_state*, New State = *new_state*

Fields:

The fields are as follows:

- *line*—the line number that changed state. The first line is line 1.
- *old_state*, *new_state*—the previous and current line states; see *eDVA Line States* (page 163) for details.

Action:

If the new state indicates a trouble condition, correct the problem.

Voice Line Protection State Change

The specified line has detected or cleared a protection fault.

Format:

Voice Line Protection State Change, Line Number = *line*, New State = *new_state*

Fields:

The fields are as follows:

- *line*—the line number that changed state. The first line is line 1.
- *new_state*—the current protection state; one of:

- o Fault DETECTED
- o Fault CLEARED

Action:

Monitor the E-UE for service issues and possible replacement.

Power Supply Telemetry Log

The E-UE has detected a change in its battery telemetry state.

Format:

Power Supply Telemetry - *state*

Fields:

The *state* field reflects the telemetry state; see *E-UE Battery States* (page 164) for details. If a Replace Battery condition persists for 24 hours, the E-UE generates another Power Supply Telemetry log.

Action:

Replace the E-UE battery if indicated by the telemetry state.

MTA TFTP: Successful

The E-UE successfully downloaded its eDVA provisioning file.

Format:

MTA TFTP: Successful

Action:

None.

MTA PROV: Successful!

The E-UE successfully completed its eDVA provisioning.

Format:

MTA PROV: Successful!

Action:

None.

SIP General Failure

Severity:

Major, service-affecting

Cause:

The registration request failed with a 4xx, 5xx, or 6xx error (other than 401 or 407) and authorization has already been sent.

Impact:

The eDVA is unable to register. Data communications are not necessarily affected.

Action:

Check the server logs for the error message associated with the registration request and correct the problem as required.

SIP Network Failure

Severity

Major, service-affecting

Cause

(SIP only) An invalid domain name for registration was specified and the DNS server could not resolve the name.

Impact

The eDVA is unable to register. Data communications are not necessarily affected.

Action

Correct the domain name, or add the domain name to the DNS server.

SIP Authentication Failure

Severity:

Major, service-affecting

Cause:

(SIP loads only) Telephony Modem can not be authenticated by the P-CSCF.

Impact:

Telephony is disabled. Data communications are not necessarily affected.

Action:

Verify that the authentication strings in the eDVA configuration file match those expected by the P-CSCF.

SIP Registration Timeout

Severity:

Major, service-affecting

Cause:

(SIP loads only) Telephony Modem lost communications to the SIP registry server.

Impact:
Telephony is disabled. Data communications are not necessarily affected.

Action:
Check the communications path between the eDVA and the SIP registry server.

SIP Proxy Loss of Communications

Severity:
Major, service-affecting

Cause:
(SIP loads only) Telephony Modem lost communications to the P-CSCF and can not initiate calls.

Impact:
eDVA cannot initiate calls. Data communications are not necessarily affected.

Action:
Check the communications path between the eDVA and the P-CSCF.

SSH LOGIN ACCEPTED

A user has successfully logged into the Telephony Modem using SSH.

Format:
SSH LOGIN ACCEPTED FROM IP [*ipaddr*] . USERNAME - (*name*).

Fields:
The *ipaddr* is the IP address (e.g. 192.168.42.42) of the client logging into the Telephony Modem. The *name* field is the user name entered. The *name* field is the user name entered.

Action:
None.

SSH LOGIN REJECTED

A user has unsuccessfully attempted to log into the Telephony Modem using SSH.

Format:
SSH LOGIN REJECTED FROM IP [*ipaddr*] . USERNAME - (*name*).

Fields:
The *ipaddr* is the IP address (e.g. 192.168.42.42) of the client logging into the Telephony Modem. The *name* field is the user name entered.

Action:

None. However, a large number of unsuccessful login attempts may indicate a potential intrusion attempt.

SSH LOGIN REJECTED - MAX ATTEMPTS REACHED

A user unsuccessfully attempted to log in too many times. The session was disconnected.

Format:

SSH LOGIN REJECTED - MAX ATTEMPTS REACHED

Action:

Continue to monitor the logs for “SSH LOGIN REJECTED” messages. Blocking the IP addresses of any attempted intruders may be necessary to protect your network.

Touchstone Firmware Upgrade Failed Before Download Attempt

Touchstone Firmware Upgrade failed before attempting a download.

Format:

Touchstone Firmware Upgrade Failed Before Download Attempt: *reason*

Fields:

The *reason* description is one of the following:

Provisioned upgrade, bad IP address in **arrisCmDevSwTable**

Download initiated via configuration file has failed. A match was found in the ARRIS table but the IP address was invalid. No download was attempted.

Provisioned upgrade, bad filename in **arrisCmDevSwTable**

Download initiated via configuration file has failed. A match was found in the ARRIS table but the filename was invalid. No download was attempted.

Provisioned upgrade, no match in **arrisCmDevSwTable**

Download initiated via configuration file has failed. No match was found in the ARRIS table. No download was attempted.

Manual upgrade, bad IP address in **arrisCmDevSwTable**

Download initiated via remote SNMP browser has failed. A match was found in the ARRIS table but the IP address was invalid. No download was attempted.

Manual upgrade, bad filename in **arrisCmDevSwTable**

Download initiated via remote SNMP browser has failed. A match was found in the ARRIS table but the file name was invalid. No download was attempted.

Manual upgrade, no match in **arrisCmDevSwTable**

Download initiated via remote SNMP browser has failed. No match was found in the ARRIS table. No download was attempted.

Manual upgrade, device not in operational state
Download initiated via remote SNMP browser has failed. Device not in the operational state. This is also the error returned when **arrisCmDevSwAdminStatus** is set to `upgradeFromArrisMgt` in a configuration file.

Action:
Correct the data in the upgrade table or on the TFTP server as necessary, then retry the download.

Touchstone Firmware Upgrade Failed

Touchstone Firmware Upgrade has failed. Standard DOCSIS download logs should also be present to further describe the exact download failure reason.

Format:
Touchstone Firmware Upgrade Failed: *type*

Fields:
The *type* code is one of the following:

Provisioned upgrade
Download initiated via configuration file has failed.

Manual upgrade
Download initiated via remote SNMP browser has failed.

Action:
Make sure the upgrade table and the specified TFTP servers are configured properly, then retry the download.

Touchstone Firmware Upgrade Successful

Touchstone Firmware Upgrade has succeeded. Standard DOCSIS download logs may also be present.

Format:
Touchstone Firmware Upgrade Successful: *type*

Fields:
The *type* code is one of the following:

Provisioned upgrade
Download initiated via configuration file has succeeded.

Manual upgrade
Download initiated via remote SNMP browser has succeeded.

Action:
None.

Touchstone SW Upgrade Aborted due to Battery AC-FAIL condition

Touchstone Firmware Upgrade has failed since the E-UE is running on battery power.

Format:

Touchstone SW Upgrade Aborted due to Battery AC-FAIL condition

Fields:

None

Action:

Wait for AC power to be restored before attempting the firmware upgrade again.

Touchstone SW Upgrade Aborted due to Call in Progress

Touchstone Firmware Upgrade has failed since one or more lines are off-hook and the **arrisMtaDevSwDnldNoSvcImpact** object is set to `StrictEnable`(2).

Format:

Touchstone SW Upgrade Aborted due to Call in Progress

Fields:

None

Action:

Wait for all lines to go on-hook before attempting the firmware upgrade again.

Touchstone SW Upgrade Reboot Delayed due to Call in Progress

Touchstone Firmware Upgrade has succeeded, but the E-UE has not rebooted since one or more lines are off-hook and the **arrisMtaDevSwDnldNoSvcImpact** object is set to `enable`(1).

Format:

Touchstone SW Upgrade Reboot Delayed due to Call in Progress

Fields:

None

Action:

None; the E-UE will reboot once all lines to go on-hook.

MTA DHCP RENEW: Lease Renewal delay; Voice line offhook

The eDVA DHCP RENEW sequence has been delayed because the **arrisMtaDevDhcpNoSvcImpact** object is enabled (value is either `dontSend` or `sendIgnore`) and one of the voice lines is offhook.

Format:

MTA DHCP RENEW: Lease Renewal delay; Voice line offhook

Action:

None; the eDVA will not RENEW its IP address until all voice lines are on-hook.

MTA DHCP REBIND: Lease Renewal delay; Voice line offhook

The eDVA DHCP RENEW period has timed out, and the eDVA DHCP REBIND sequence has been delayed, because the **arrisMtaDevDhcpNoSvcImpact** object is enabled (value is either `dontSend` or `sendIgnore`) and one of the voice lines is offhook.

Format:

MTA DHCP REBIND: Lease Renewal delay; Voice line offhook

Action:

None; the eDVA will not RENEW its IP address until all voice lines are on-hook.

Power Supply Telemetry Alarm

Severity:

Major

Cause:

The E-UE has lost AC power or has encountered a problem in the battery charging circuitry. The alarm includes one of the following battery status codes:

- AC Fail—the E-UE has detected an AC power failure.
- Replace Battery—the battery has deteriorated to about 75% of its off-the-shelf capacity and should be replaced.
- Battery Missing—the battery was not installed, has been removed, or cannot be detected.

The event log provides more information about the battery status.

Impact:

None at time of alarm. Depending on the condition of the battery and the nature of the power failure, the E-UE may exhaust the battery before AC power is restored.

Action:

Depends on the scope of the power outage.

Gateway has reset

Severity:

Informational

Cause:

The Gateway was reset either through the web pages or by pressing the “Router Reset” button.

Impact:

Data communications (Ethernet or wifi) may be disrupted for a few seconds. Telephony is not affected.

Action:

None.

Unit has been restored to factory defaults

Severity:

Informational

Cause:

The Gateway was restored to factory defaults either through the web pages or by holding the “Router Reset” button for more than 15 seconds.

Impact:

Data communications (Ethernet or wifi) may be disrupted for a few seconds. The subscriber may have to reconfigure the Gateway or restore previous settings. Telephony is not affected.

Action:

None.

Speedtest Results

Format:

SpeedTest Results: Download Status: Complete, Speed: *dl_speed*, Upload Status: Complete, Speed: *ul_speed*;CM-MAC=*cm_mac_addr*;CMTS-MAC=*cmts_mac_addr*;CM-QOS= *cm_qos*;CM-VER= *cm_hw_ver*;

Severity:

Informational

Cause:

A speed test was conducted, and the **arrisCmDoc30SpeedtestConfigSyslogReports** object is set to `true`(1) so results would be sent to the Syslog server.

Impact:

None.

Action:

Check the results for expected performance. Note that rate limiting can affect the results. If the performance is not as expected, monitor the network for congestion or noise and correct any problems.

PacketCable Events

The following are PacketCable-related events. PacketCable events use the enterprise number **4491**. All supported PacketCable events are related to the battery and charger system.

Battery Not Low

The E-UE battery has recharged to over 25% of its maximum capacity.

Battery Low

The E-UE battery run time is less than or equal to the value of the **upsConfigLowBattTime** object (which also triggers the **upsAlarmLowBattery** alarm).

Battery Present

A missing battery has been replaced.

If a large number of “Battery Missing” and “Battery Present” messages appear in the logs for a single E-UE, this may indicate a problem with either the battery or the E-UE.

Battery Missing

The E-UE battery has been either removed or is undetectable. If the battery is still installed, it may be defective.

If a large number of “Battery Missing” and “Battery Present” messages appear in the logs for a single E-UE, this may indicate a problem with either the battery or the E-UE.

Battery Good

An E-UE battery that had previously shown a “Replace Battery” state is now good. This may indicate that the battery has been replaced.

Replace Battery

The E-UE battery has deteriorated to about 75% of its off-the-shelf capacity and should be replaced.

AC Restored

AC power has been restored to the indicated E-UE.

If a large number of “AC Restored” and “AC Fail” messages appear in the logs for a single E-UE, this may indicate that the E-UE is connected to a switched outlet or the power connection may be intermittent.

AC Fail

AC power has been removed from the E-UE, and the E-UE is running on battery power.

If a large number of “AC Restored” and “AC Fail” messages appear in the logs for a single E-UE, this may indicate that the E-UE is connected to a switched outlet or the power connection may be intermittent.

Network Failure Recovery

Touchstone firmware can automatically recover from cable cuts or degraded downstream RF conditions. No provisioning or control is necessary to take advantage of recovery features.

Recovery from Extreme Plant Conditions

In a situation where the downstream RF signal degrades to the point where the upstream transmit buffers are full, and the E-UE cannot send packets, an automatic recovery feature:

- resets the DOCSIS layer
- restarts the DSP
- notifies Call Processing to restore dialtone functionality

The result is a cleaner recovery from extreme RF degradation.

Working with Message Trace Logs

Touchstone E-UEs keep logs of:

- CM DHCP messages
- eDVA DHCP messages
- CallP messages
- MGCP
- DSX

Log messages are available only for NCS loads, through the CLI, SNMP, or the troubleshooting pages. Use this procedure to display message trace logs.

Message Capacity

The message log stores up to 250 messages. The buffer size is 25K bytes. SIP loads generate larger messages, reducing the number of messages actually stored.

The DHCP log stores up to 50 CM messages and up to 50 eDVA DHCP messages. Each DHCP buffer is 5K bytes.

Note: The E-UE captures the original DHCP Discover-Offer exchange, and all subsequent Renew exchanges.

The eDVA uses a circular buffer scheme to store messages. When a new message exceeds the available buffer space, the eDVA deletes the oldest messages as needed.

SNMP Overview

Log messages can be up to 4K bytes in length. Since an SNMP string is limited to 256 bytes, retrieving a message through SNMP requires breaking the message up. The objects **arrisMtaDevSignalingLastMsg1** through **arrisMtaDevSignalingLastMsg16** contain the selected message.

Action

Perform the following tasks as needed.

Enabling or Disabling Message Tracing

Follow these steps to enable or disable message tracing using SNMP. Trace logs are enabled by default.

1. Specify which message type that you want to enable or disable by setting the **arrisMtaDevLoggingContext** object as follows:

Value	Log Type
0	MGCP (default)
1	CM DHCP
2	eDVA DHCP
3	DSX

2. Enable or disable message tracing for the selected message type by setting the **arrisMtaDevEnableLogging** object: either **`disable`**(0) or **`enable`**(1).

Viewing Logs Using the CLI

Follow these steps to view message trace logs using the command-line interface.

1. Access the CLI as described in Enabling and Accessing the Troubleshooting CLI.
2. To view the eDVA DHCP logs:
   `/ dhcp mtaDispLog`
3. To view the CM DHCP logs:
   `/ dhcp cmDispLog`
4. To view the CallP logs:
   `/ voice callp callpDispLog`
5. To view the DSx logs:
   `/ voice callp dsxDispLog`

Viewing Logs Using SNMP

Follow these steps to read message logs using SNMP.

1. Specify which message type that you want to view by setting the **arrisMtaDevLoggingContext** object as follows:

Value	Log Type
0	MGCP (default)
1	CM DHCP
2	eDVA DHCP
3	DSX

2. Write a **`0`** to **arrisMtaDevEnableCallSigLastMsgRpt**. This creates a snapshot of the selected message log and writes the content of the last message to the **arrisMtaDevSignalingLastMsg1** through **arrisMtaDevSignalingLastMsg16** objects.
3. Read the **arrisMtaDevSignalingLastMsg1** through **arrisMtaDevSignalingLastMsg16** objects to view the last log of the selected type. If the message does not require all 16 objects, the unused objects return empty strings.
4. To read older log messages, write a higher value to the **arrisMtaDevEnableCallSigLastMsgRpt** object — for example, a **`1`** reads the second newest message, **`2`** reads the message before that, and so on. To refresh the log, write another **`0`**.

   See "Message Capacity" above for the maximum number of each log type that the Telephony Modem stores.

Viewing Logs Using the Web-based Interface

Follow these steps to view message logs using the web-based troubleshooting interface.

1. Use a web browser to access the web-based interface as described in see "*Accessing the Standard Pages* (page 304).
2. If you are currently viewing the Basic pages, click the **Advanced** link. Enter the Password of the Day if necessary.
3. To view either CM or eDVA DHCP logs:
   a. Select the **DHCP** link at the top of any Advanced web page to access the DHCP Parameters page.
   b. At the bottom of the DHCP Parameters page (scroll down to the bottom of the page if necessary), select either the **CM-DHCP** or **MTA-DHCP** link.

      The E-UE displays the selected DHCP log.
4. To view MGCP and DSX logs:
   a. Select the **QoS** link at the top of any Advanced web page to access the QoS Statistics page.

b. To view MGCP logs, select the **Show Call Signalling Log** link (about midway down the page).

c. To view DSX logs, select the **Show DSX Log** link at the bottom of the page (scroll down to the bottom of the page if necessary).

The E-UE displays the selected log.

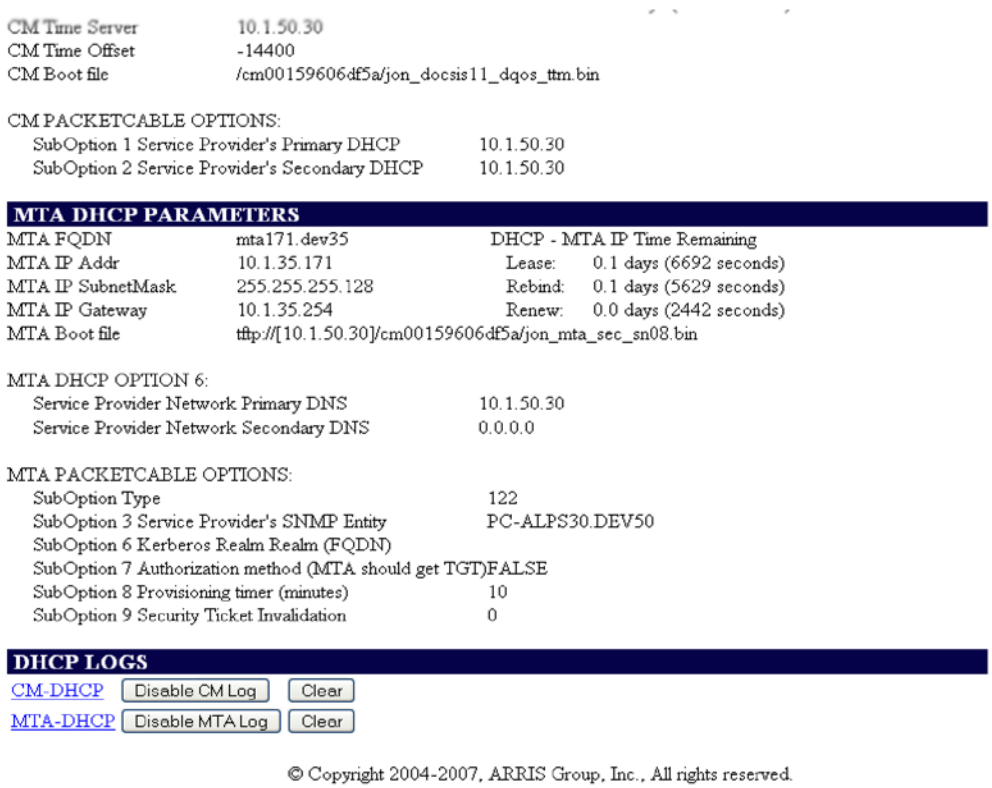
Controlling Logs Using the Web-Based Interface

Follow these steps to enable, disable, or clear message logs using the web-based interface.

1. Access the Advanced web pages as described in see "*Accessing the Advanced Pages* (page 305).
2. To work with DHCP logs:

a. Click the **DHCP** link to access the DHCP page.

b. Scroll down to the bottom of the screen to view the logging controls:



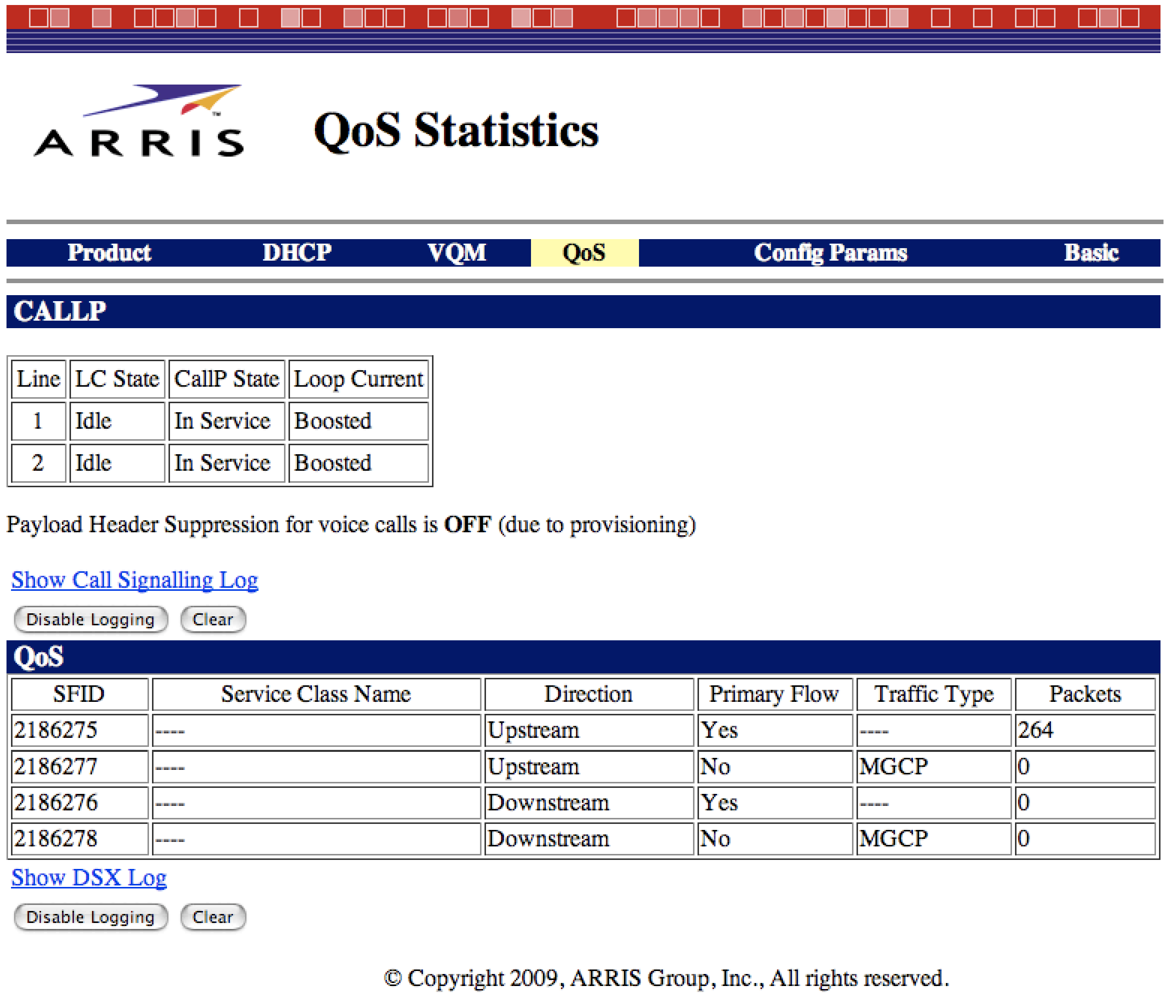
c. To clear the CM DHCP or eDVA DHCP logs, click the **Clear** button for the desired log.

d. To enable or disable CM DHCP logs, click the **Disable CM Log** button. This button changes to **Enable CM Log** when logging is disabled.

e. To enable or disable eDVA DHCP logs, click the **Disable MTA Log** button. This button changes to **Enable MTA Log** when logging is disabled.

3. To work with MGCP and DSX logs:

a. Click the **QoS** link to access the Qos Statistics page:



QoS Statistics

Product | DHCP | VQM | QoS | Config Params | Basic

CALLP

Line	LC State	CallP State	Loop Current
1	Idle	In Service	Boosted
2	Idle	In Service	Boosted

Payload Header Suppression for voice calls is **OFF** (due to provisioning)

Show Call Signalling Log

Disable Logging | Clear

QoS

SFID	Service Class Name	Direction	Primary Flow	Traffic Type	Packets
2186275	----	Upstream	Yes	----	264
2186277	----	Upstream	No	MGCP	0
2186276	----	Downstream	Yes	----	0
2186278	----	Downstream	No	MGCP	0

Show DSX Log

Disable Logging | Clear

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b. To clear the MGCP log, click the **Clear** button midway down the screen.

c. To enable or disable CallP logging, click the **Disable Logging** button midway down the screen. This button changes to **Enable Logging** when logging is disabled.

d. To enable or disable DSX logging, click the **Disable Logging** button at the bottom of the screen. This button changes to **Enable Logging** when logging is disabled.

e. To clear the DSX log, click the **Clear** button at the bottom of the screen.

Clearing Message Trace Logs Using the CLI

Follow these steps to clear message trace logs using the CLI.

1. Access the CLI as described in Enabling and Accessing the Troubleshooting CLI.
2. To clear the eDVA DHCP logs:
   ```
   / dhcp mtaClearLog
   ```
3. To clear the CM DHCP logs:
   ```
   / dhcp cmClearLog
   ```

4. To clear the MGCP logs:
   `/ voice callp callpClrLog`
5. To clear the DSx logs:
   `/ voice callp dsxClrLog`

Capturing Signaling Traces

CAUTION
Potential security breach
This feature allows NCS signaling messages to enter the network as clear text. This breaks security as defined in the PacketCable Security specification. IPsec is used to secure the message link between the eDVA and CMS to, among other things, keep the voice keys exchanged between endpoints secure. Unauthorized personnel may potentially be able to monitor a subscriber's voice traffic.

Touchstone firmware can generate a Syslog report that contains a full signaling trace on a Touchstone E-UE. Individual signaling messages may be up to 4000 bytes in length. Since messages of this size would violate the maximum message size limitations of the Syslog server, long messages are broken into blocks of 128 bytes, time stamped, and numbered for reassembly. The eDVA then sends the blocks to the Syslog server IP address defined during normal E-UE provisioning.

Signaling tracing is controlled using the **arrisMtaDevEnableCallpSigTrace** MIB object. You can enable or disable message tracing output on an E-UE using an SNMP manager.

Note: Touchstone firmware supports this feature only for capturing NCS signaling traces. Support for capturing SIP signaling traces may be added in a future release.

Controlling Signaling Tracing

Use the **arrisMtaDevEnableCallpSigTrace** MIB object (part of the **arrisMtaDevBase** MIB) to enable or disable signal tracing. The default value for this object is **`disable`**(0).

CAUTION
Potential performance impacts
The number of messages expected as a result of enabling this feature can affect the real-time performance of the E-UE, and may cause network congestion.

Set the object as follows:

- To enable signaling message tracing, set the object to **`enable`**(1). Tracing continues until disabled using the MIB object, or the E-UE is reset.
- To disable signaling message tracing, set the MIB object to **`disable`**(0).

Note: The signaling trace feature cannot be enabled through either the CM configuration file or the eDVA configuration file.

Interpreting the Signaling Trace Output Data

The following is an example of a single part transmission from the E-UE to a NCS Call server:

```
Oct 21 10:55:04 10.1.61.17 Oct 21 10:55:03 2005 mta17.dev61 <44> <4115>
<37> <00:13:11:23:23:E7> <Xmit: (17: 1 of 1) - ’NTFY 8 aaln/2@mta17.dev61
MGCP 1.0 NCS 1.0 X: 26002 O: hu ’>
```

The following is an example of a single part Receive from a NCS Call Server to the E-UE:

```
Oct 21 10:55:04 10.1.61.17 Oct 21 10:55:03 2005 mta17.dev61 <45> <4115>
<38> <00:13:11:23:23:E7> <Rcv: (19: 1 of 1) - ’200 8 OK ’>
```

The following output is part of the Syslog header, and appears in both Transmit and Receive trace messages.

Oct 21 10:55:04
Syslog server Date and Time.

10.1.61.17
The IP address of the eDVA that sent the message.

Oct 21 10:55:03 2005
The E-UE generated Data and Time.

mta17.dev61
The FQDN of the E-UE.

<44>
The Syslog message Event Number. It is incremented for each message in the Syslog.

<4115>
The ARRIS Enterprise Number.

The following sections describe the transmit and receive data payloads.

Interpreting the Transmit Data Payload

The following is an example of a single part transmission from the E-UE to a NCS Call server:

```
Oct 21 10:55:04 10.1.61.17 Oct 21 10:55:03 2005 mta17.dev61 <44> <4115>
<37> <00:13:11:23:23:E7> <Xmit: (17: 1 of 1) - ’NTFY 8 aaln/2@mta17.dev61
MGCP 1.0 NCS 1.0 X: 26002 O: hu ’>
```

The transmit message data can be broken down into two parts, the Header and the signaling data itself. The header indicates that this is a transmitted message from the E-UE; it provides the sequence number of the message, the block number, total number of blocks in this message, and the payload of the signaling message.

The following table shows the data payload that is part of the transmit message trace. The first two parts comprise the header; the next two parts are the actual message.

<37>
An Index number indicating that this is a transmitted message. All transmitted messages are of type 37.

<00:13:11:23:23:E7>
The MAC address of the E-UE that transmitted the message.

Xmit: (17: 1 of 1) -
All transmitted messages start with Xmit. The **17** is a message sequence number. All transmitted messages have a unique sequence number that increases by one for each complete message transmitted. This internal value is a 32-bit unsigned integer value that increments only when tracing is active. Sequence numbers start at zero. The sequence numbers increment only when a message is sent to the Syslog. The “1 of 1” indicates that this is part 1 of a one part message. All parts of the same message have the same sequence number.

’NTFY 8 aaln/2@mta17.dev61

MGCP 1.0 NCS 1.0 X: 26002 O: hu ’
The actual signaling message data. The data is surrounded by single quotes. All signaling messages are NULL terminated strings.

Interpreting the Receive Data Payload

The following is an example of a single part Receive from a NCS Call Server to the eDVA:

```
Oct 21 10:55:04 10.1.61.17 Oct 21 10:55:03 2005 mta17.dev61 <45> <4115>
<38> <00:13:11:23:23:E7> <Rcv: (19: 1 of 1) - ’200 8 OK ’>
```

The receive message data can be broken down into two parts, the Header and the signaling data. The header indicates that this is a signaling message received by the eDVA; it provides the sequence number of the message, the block number, total number of blocks in this message, and the signaling data received.

The following table shows the data payload that is part of a receive message trace:

<38>
An index number indicating that this is a received signaling message. All received messages are type 38.

<00:13:11:23:23:E7>
The MAC address of the Message Destination.

Rcv: (19: 1 of 1)
All received messages start with `Rcv:`. The **19** is a message sequence number. All received messages have a unique sequence number that increases by one for every complete message received by the eDVA. This internal value is a 32-bit unsigned integer value and only increments when the tracing is active. Sequence numbers start at zero. All messages have a unique sequence number. The “1 of 1” indicates that this is part 1 of a 1-part message. All parts of the same message have the same sequence number.

’200 8 OK ’
The actual received signaling data. The data is surrounded by single quotes.

Signaling Trace Feature Example Output

Below is a small sample output for a typical off-hook and on-hook sequence in NCS. In a real world situation on an actual network Syslog server, there could be messages unrelated to this feature interleaved with these messages:

```
Oct 21 10:55:00 10.1.61.17 Oct 21 10:54:59 2005
mta17.dev61 <37> <4115> <37> <00:13:11:23:23:E7>
<Xmit: (14: 1 of 1) - ’NTFY 7 aaln/2@mta17.dev61 MGCP 1.0 NCS 1.0 X: 25888
O: hd ’>
Oct 21 10:55:00 10.1.61.17 Oct 21 10:54:59 2005 mta17.dev61 <38> <4115>
<38> <00:13:11:23:23:E7>
<Rcv: (16: 1 of 1) - ’200 7 OK ’>
Oct 21 10:55:00 10.1.61.17 Oct 21 10:54:59 2005 mta17.dev61 <39> <4115>
<38> <00:13:11:23:23:E7>
<Rcv: (17: 1 of 1) - ’RQNT 3752 aaln/2@mta17.dev61 MGCP 1.0 NCS 1.0 X:
26001 S: Q: loop R: hf(I), hu(N) ’>
Oct 21 10:55:01 10.1.61.17 Oct 21 10:54:59 2005 mta17.dev61 <40> <4115>
<37> <00:13:11:23:23:E7>
<Xmit: (15: 1 of 1) - ’200 3752 OK ’>
Oct 21 10:55:01 10.1.61.17 Oct 21 10:54:59 2005 mta17.dev61 <41> <4115>
<38> <00:13:11:23:23:E7>
<Rcv: (18: 1 of 2) - ’RQNT 3753 aaln/2@mta17.dev61 MGCP 1.0 NCS 1.0 X:
26002 D: (#|A|D|[2-9]11|0[2-9]11|0T|00|010|11X|[2-9]XXXXXX|[01][2-
9]XXXXXXXXX|0’>
Oct 21 10:55:01 10.1.61.17 Oct 21 10:54:59 2005 mta17.dev61 <42> <4115>
<38> <00:13:11:23:23:E7>
<Rcv: (18: 2 of 2) - ’1[1-9]XXXXX|10XXXX|X.#) S: dl R: hf(I,K), hu(N), oc,
of, [0-9*#T](D) ’>
Oct 21 10:55:01 10.1.61.17 Oct 21 10:54:59 2005 mta17.dev61 <43> <4115>
<37> <00:13:11:23:23:E7>
<Xmit: (16: 1 of 1) - ’200 3753 OK ’>
Oct 21 10:55:04 10.1.61.17 Oct 21 10:55:03 2005 mta17.dev61 <44> <4115>
<37> <00:13:11:23:23:E7>
<Xmit: (17: 1 of 1) - ’NTFY 8 aaln/2@mta17.dev61 MGCP 1.0 NCS 1.0 X: 26002
O: hu ’>
Oct 21 10:55:04 10.1.61.17 Oct 21 10:55:03 2005 mta17.dev61 <45> <4115>
<38> <00:13:11:23:23:E7>
<Rcv: (19: 1 of 1) - ’200 8 OK ’>
Oct 21 10:55:04 10.1.61.17 Oct 21 10:55:03 2005 mta17.dev61 <46> <4115>
<38> <00:13:11:23:23:E7>
<Rcv: (20: 1 of 1) - ’RQNT 3788 aaln/2@mta17.dev61 MGCP 1.0 NCS 1.0 X:
26025 S: Q: loop R: hd(N) ’>
Oct 21 10:55:04 10.1.61.17 Oct 21 10:55:03 2005 mta17.dev61 <47> <4115>
<37> <00:13:11:23:23:E7>
<Xmit: (18: 1 of 1) - ’200 3788 OK’> Oct 21 10:55:33 10.1.61.17 Oct 21
10:55:32 2005 mta17.dev61 <48> <4115> <37> <00:13:11:23:23:E7>
<Xmit: (19: 1 of 1) - ’NTFY 9 aaln/2@mta17.dev61 MGCP 1.0 NCS 1.0 X: 26025
O: hd ’>
Oct 21 10:55:33 10.1.61.17 Oct 21 10:55:32 2005 mta17.dev61 <49> <4115>
<38> <00:13:11:23:23:E7>
<Rcv: (21: 1 of 1) - ’200 9 OK ’> Oct 21 10:55:33 10.1.61.17 Oct 21
10:55:32 2005 mta17.dev61 <50> <4115> <38> <00:13:11:23:23:E7>
<Rcv: (22: 1 of 1) - ’RQNT 3940 aaln/2@mta17.dev61 MGCP 1.0 NCS 1.0 X:
26060 S: Q: loop R: hf(I), hu(N) ’>
```

Configuring SNMP Coexistence

The firmware provides several SNMPv3 security-related features. SNMP Coexistence is a feature that allows SNMPv1 and SNMPv2c network management systems to function within the context of SNMPv3 security and view-based MIB access. The NMS can use an SNMPv1 or SNMPv2 community string to access the eMTA's MIB or to receive traps.

Overview

This procedure provides details on adding the necessary MIBs and TLVs to a cable modem configuration file.

To configure the eMTA for coexistence mode, you must create a row entry in the **snmpCommunityTable**, to map the community string to an SNMPv3 security name. You can optionally modify the following tables to provide extended access control.

vacmSecurityToGroupTable
: Two row entries, containing group name information. One entry supports SNMPv1 access and the other entry supports SNMPv2 access.

vacmAccessTable
: Two row entries, which map the community name, security name, and group name information to an SNMPv3 security name. The DOCSIS-standard default security name for cable modems is **`docsisManager`**. One entry supports SNMPv1 access and the other entry supports SNMPv2 access.

Enabling SNMP Access

SNMP access is enabled and open to all users until applying any restricted settings in the CM configuration file. The following example fragment can be placed in a CM configuration file to restrict SNMP access:

```
SnmpMib = docsDevNmAccessCommunity.1 "public"
SnmpMib = docsDevNmAccessControl.1 read
SnmpMib = docsDevNmAccessInterfaces.1 hexstr: 40
SnmpMib = docsDevNmAccessStatus.1 createAndGo
SnmpMib = docsDevNmAccessCommunity.2 "private"
SnmpMib = docsDevNmAccessControl.2 readWrite
SnmpMib = docsDevNmAccessInterfaces.2 hexstr: 40
SnmpMib = docsDevNmAccessStatus.2 createAndGo
```

The firmware also allows SNMP access through the RF interface when the CM configuration file does not specify **docsDevNmAccessInterfaces** but includes all other NmAccess entries. For example, the following fragment enables SNMP access through the RF interface:

```
SnmpMib = docsDevNmAccessIp.10 192.168.31.0
SnmpMib = docsDevNmAccessIpMask.10 255.255.255.0
SnmpMib = docsDevNmAccessCommunity.10 "public"
SnmpMib = docsDevNmAccessControl.10 read
SnmpMib = docsDevNmAccessStatus.10 createAndGo
```

Note: TS11.1 allows read-only SNMP access to certain objects from the LAN interfaces before the modem has ranged and registered, in accordance with CM-SP-OSSIv3.0-I08-090121.

Configuration File Notes

Keep the following notes in mind when creating or editing configuration files.

- A MIB object whose type is “StorageType” must always have a value of **`volatile`**.
- A MIB object whose type is “RowStatus” should have a value of **`createAndGo`**. The eMTA automatically changes its value to **`active`** after successfully adding the row.

SNMP Access Mode

The following examples configure SNMP access to the eMTA for SNMPv1v2c coexistence mode. This allows an NMS (i.e. a MIB Browser) to access the eMTA’s MIBs with a simple community string using either SNMPv1 or SNMPv2.

The SNMP requests GET, GET-NEXT, and SET are all supported. The examples use the community name **my_password**.

SNMP Trap Transmission

SNMP trap transmission uses the DOCSIS TLV-38 (SNMPv3 Notification Receiver) configuration file element.

The example configures two trap destinations, each with a different IP address. One destination supports SNMPv1 traps and the other destination supports SNMPv2 traps. The parameters for each trap destination are:

- Trap destination #1:
  - IP Address: 10.1.50.100
  - Trap Type: SNMPv1
- Trap destination #2:
  - IP Address: 10.1.50.80
  - Trap Type: SNMPv2

Note: Change the trap IP address to the IP address of your specific trap server in the configuration file.

This example starts with a basic DOCSIS 1.1 CM configuration file, containing enough information to allow a cable modem to range and register, and then add the coexistence MIB elements to it. If you have a CM configuration file that you are already using, start with that file and add the coexistence elements to it.

snmpCommunityTable Parameters

The following table shows the example row to add to the **snmpCommunityTable**. The index for this table is an octet string; the example uses the string **comm1** as the index value. You can use a different string if you desire. Italicized values in the table are default values that are created automatically.

The following examples use the ARRIS PacketACE Configuration Editor to create the configuration file, covering only the details needed to add the desired functionality. See the *PacketACE Configuration Tools User's Guide* for more information about using PacketACE.

Object Name	Value	Required?
snmpCommunityName.*comm1*	my_password	Yes
snmpCommunitySecurityName.*comm1*	`rwAccess`	Yes
snmpCommunityContextEngineID.*comm1*	*local snmpEngineID*	No
snmpCommunityContextName.*comm1*	(zero-length)	No
snmpCommunityTransportTag.*comm1*	(zero-length)	No
snmpCommunityStorageType.*comm1*	`volatile`(2)	No
snmpCommunityStatus.*comm1*	`createAndGo`(4)	Yes

Note: Avoid adding table index objects to the configuration file; the firmware fills in the index object using the index value supplied with the object name (.comm1 in this example). See see "*Adding the snmpCommunityTable* (page 192) for the proper way to add rows.

Object order is not important.

vacmSecurityToGroupTable Parameters

The following table shows the example row to add to the **vacmSecurityToGroupTable**. This table has an index consisting of two objects:

vacmSecurityModel
: Corresponds to the SNMP version in use; in this example; it takes the values **1** and **2** to allow support for both SNMPv1 and SNMPv2 requests.

vacmSecurityName
: Corresponds to the **snmpCommunitySecurityName** object in the **snmpCommunityTable** (`rwAccess` in this example).

Object Name	Value (row 1)	Value (row 2)
vacmSecurityModel (row index 1)	1 (SNMPv1)	2 (SNMPv2)
vacmSecurityName (row index 2)	**rwAccess**	**rwAccess**
vacmGroupName	**rwAccess1**	**rwAccess2**
vacmSecurityToGroupStorageType	`volatile`(2)	`volatile`(2)
vacmSecurityToGroupStatus	`createAndGo`(4)	`createAndGo`(4)

vacmAccessTable Parameters

The following table shows the example row to add to the **vacmAccessTable**. This table has an index consisting of four objects:

vacmGroupName
: Corresponds to the **vacmGroupName** object in the **vacmSecurityToGroupTable** (`rwAccess` in our example). CM and MTA group names should be unique.

vacmAccessContentPrefix
: An octet string; in this example we use an empty (zero length) string. This is shown as “” in the table below.

vacmAccessSecurityModel
: Corresponds to the SNMP version in use; in this example; it takes the values `1` and `2` to allow support for both SNMPv1 and SNMPv2 requests.

vacmAccessSecurityLevel
: For SNMP coexistence, use a value of `1` (noAuthnoPriv). This means that the eMTA has no configured SNMPv3 USM security users/keys.

Object Name	Value (row 1)	Value (row 2)
GroupName (row index 1)	**rwAccess**	**rwAccess**
ContentPrefix (row index 2)	“”	“”
SecurityModel (row index 3)	1 (SNMPv1)	2 (SNMPv2)
SecurityLevel (row index 4)	1 (noAuthnoPriv)	1 (noAuthnoPriv)
vacmAccessContextMatch	**exact**(1)	**exact**(1)
vacmAccessReadViewName	**docsisManagerView**	**docsisManagerView**
vacmAccessWriteViewName	**docsisManagerView**	**docsisManagerView**
vacmAccessNotifyViewName	**docsisManagerView**	**docsisManagerView**
vacmAccesssStorageType	**volatile**(2)	**volatile**(2)
vacmAccessStatus	**createAndGo**(4)	**createAndGo**(4)

Action

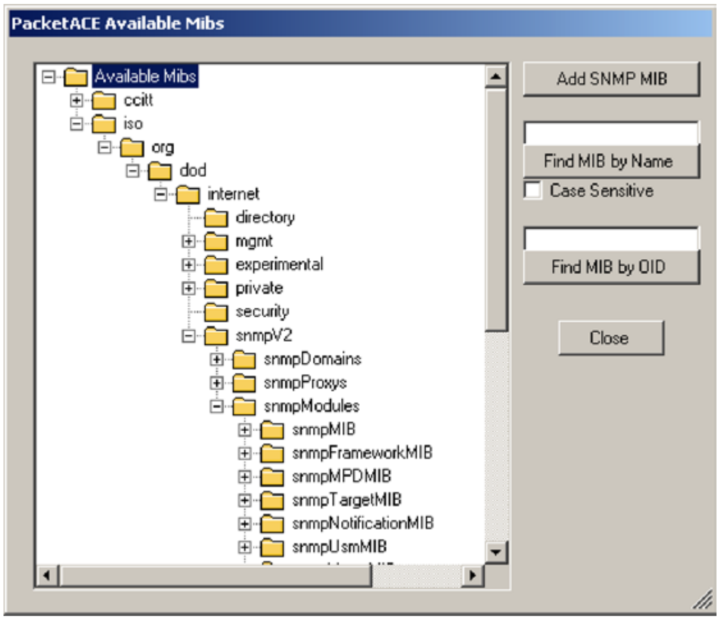
Perform the following tasks in the order shown.

Adding the snmpCommunityTable

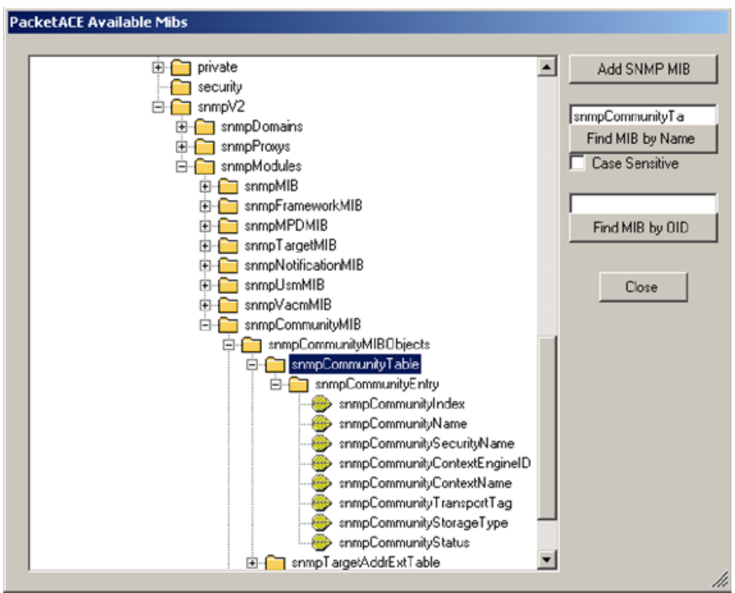
Follow these steps to add coexistence MIB objects to a CM configuration file using PacketACE.

1. Click the “Add SNMP MIB” icon or select **Edit menu → Add SNMP MIB**.

The Available MIBs tree appears:



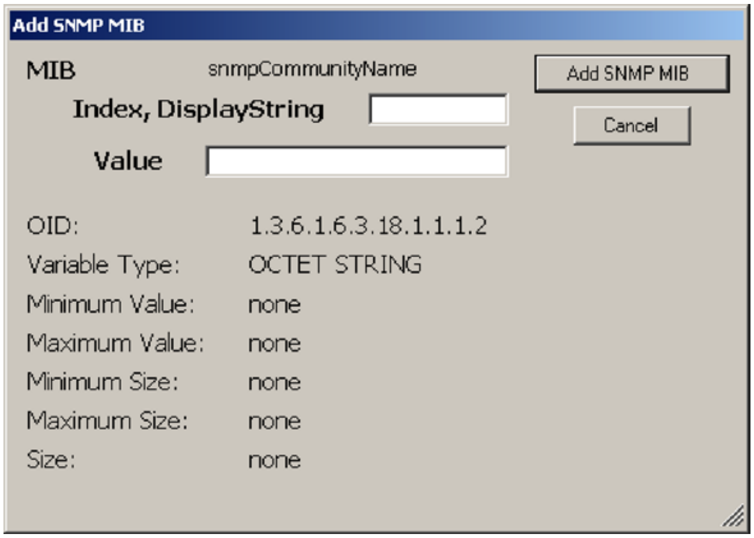
2. Locate the **snmpCommunityTable**. If you are not sure where the table is located in the tree, enter the name in the upper field along the right side of the PacketACE window, then click **Find MIB by Name**. The following figure shows the **snmpCommunityTable**.



3. Double-click on one of the objects listed in the following table.

Object Name	Value
snmpCommunityName	*my_password*
snmpCommunitySecurityName	**rwAccess**
snmpCommunityStatus	`createAndGo`(4)

The Add SNMP MIB window appears:

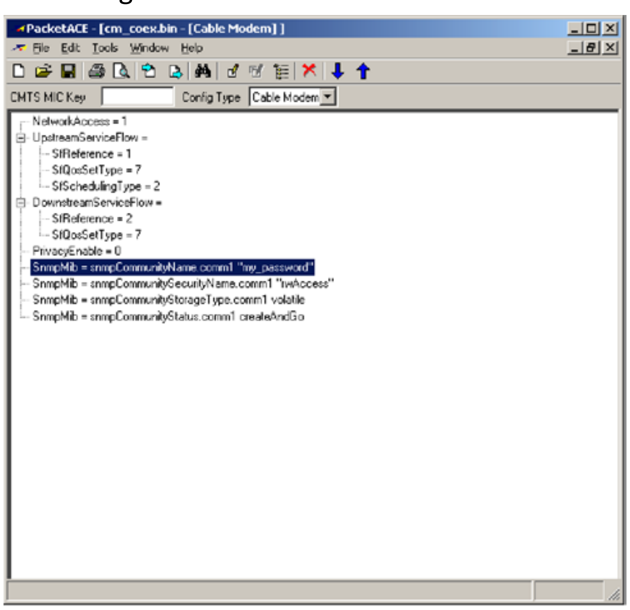


4. Enter the index name (**comm1**) in the **Index,DisplayString** field, and the value from the table in step 3 in the **Value** field.

   Some of the objects have a fixed set of values; this is indicated by the drop-down menu button at the end of the **Value** field. Click the drop-down to display a list of allowed values, and choose the correct value.

5. Repeat steps 3 and 4 until all values are filled in.

   The configuration file should now look similar to the following:



Note: The order of your MIB entries may be different than what is shown above.

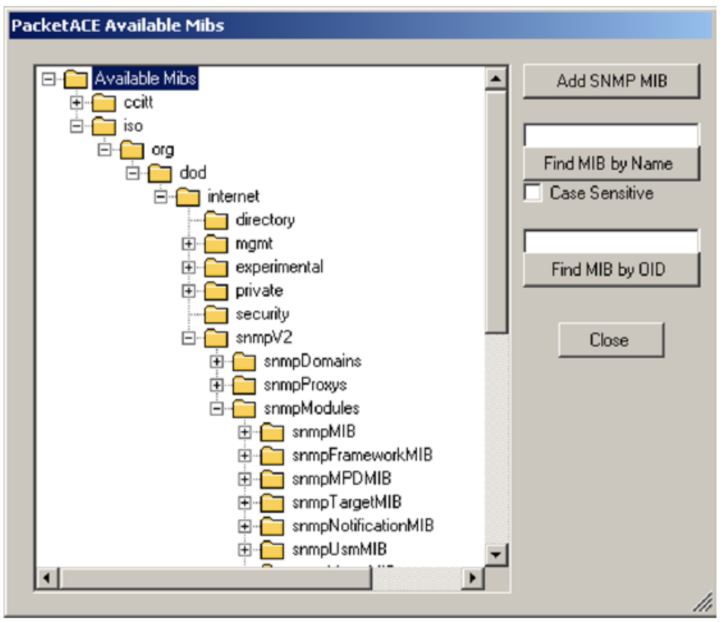
6. Proceed to see "*Adding the vacmSecurityToGroupTable* (page 195).

Adding the vacmSecurityToGroupTable

Follow these steps to add coexistence MIB objects to a CM configuration file using PacketACE.

1. Click the "Add SNMP MIB" icon or select **Edit menu → Add SNMP MIB**.

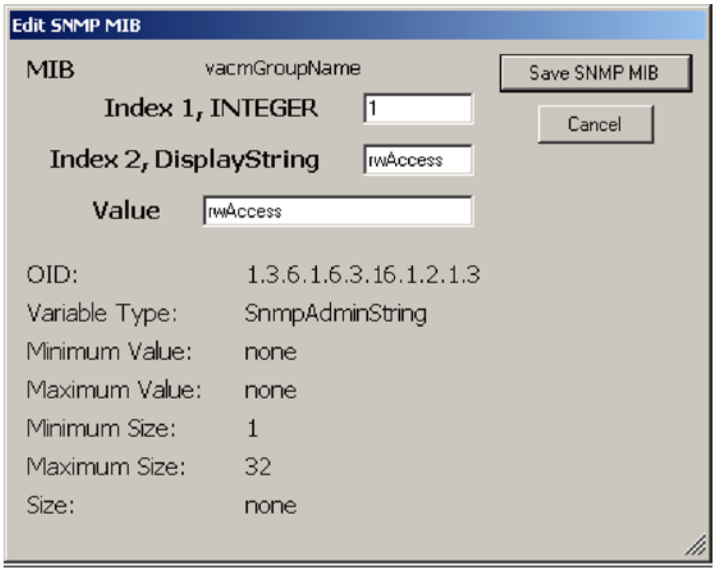
   The Available MIBs tree appears.



2. Locate the **vacmSecurityToGroupTable**. If you are not sure where the table is located in the tree, enter the name in the upper field along the right side of the PacketACE window and then click **Find MIB by Name**.
3. Double-click on one of the objects listed in the following table.

Object Name	Value (row 1)	Value (row 2)
vacmGroupName	`rwAccess`	`rwAccess`
vacmSecurityToGroupStorageType	`volatile`(2)	`volatile`(2)
vacmSecurityToGroupStatus	`createAndGo`(4)	`createAndGo`(4)

The Add SNMP MIB window appears:

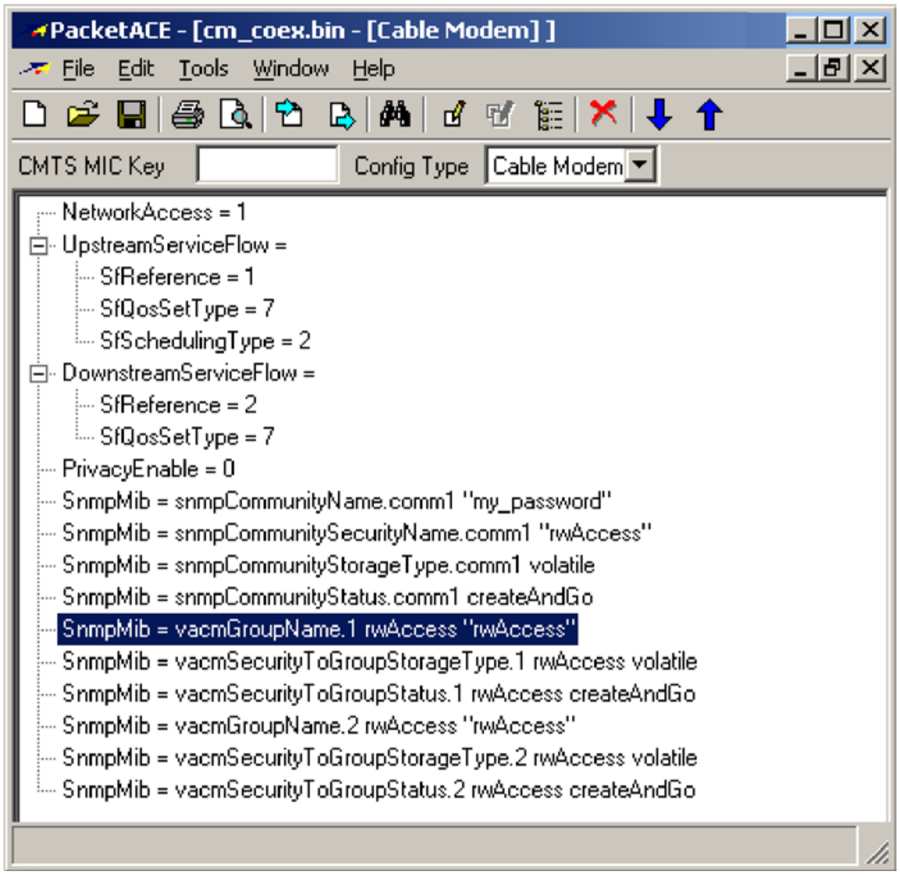


4. Enter the first index (**1** or **2**) in the **Index1,Integer** field, the second index (**rwAccess**) in the **Index2,DisplayString** field, and the value from the table in step 3 in the **Value** field.

   Some of the objects have a fixed set of values; this is indicated by the drop-down menu button at the end of the **Value** field. Click the drop-down to display a list of allowed values, and choose the correct value.

5. Repeat steps 3 and 4 until all values are filled in.

6. The configuration file should now be similar to the following:



Note: The order of your MIB entries may be different than what is shown above.

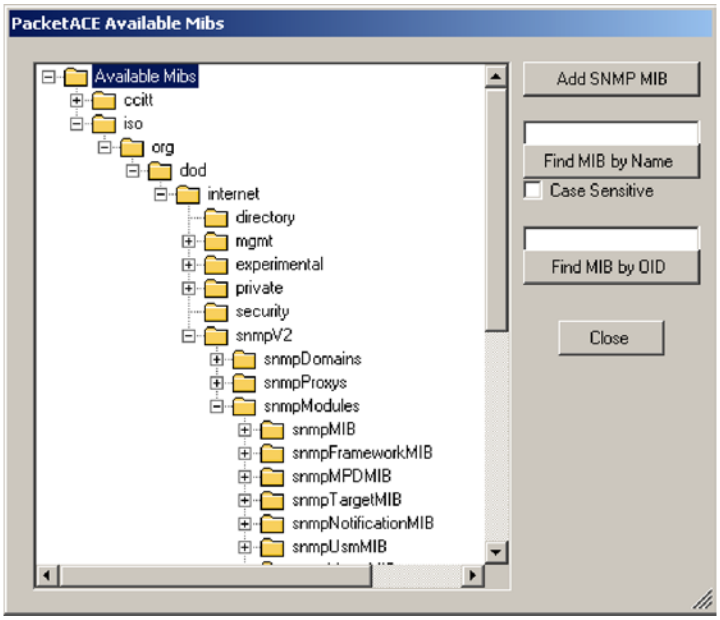
7. Proceed to see "*Adding the vacmAccessTable* (page 197).

Adding the vacmAccessTable

Follow these steps to add coexistence MIB objects to a CM configuration file using PacketACE.

1. Click the “Add SNMP MIB” icon or select **Edit menu → Add SNMP MIB**.

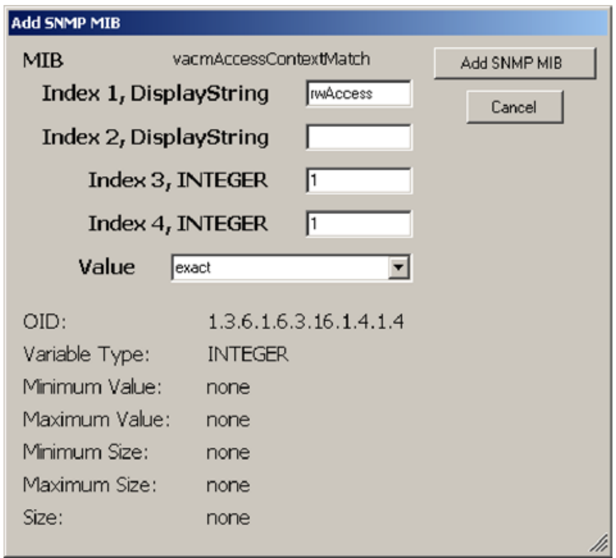
The Available MIBs tree appears:



2. Locate the **vacmAccessTable**. If you are not sure where the table is located in the tree, enter the name in the upper field along the right side of the PacketACE window, then click **Find MIB by Name**.
3. Double-click on one of the objects listed in the following table.

Object Name	Value (row 1)	Value (row 2)
vacmAccessContextMatch	`exact`(1)	`exact`(1)
vacmAccessReadViewName	`docsisManagerView`	`docsisManagerView`
vacmAccessWriteViewName	`docsisManagerView`	`docsisManagerView`
vacmAccessNotifyViewName	`docsisManagerView`	`docsisManagerView`
vacmAccesssStorageType	`volatile`(2)	`volatile`(2)
vacmAccessStatus	`createAndGo`(4)	`createAndGo`(4)

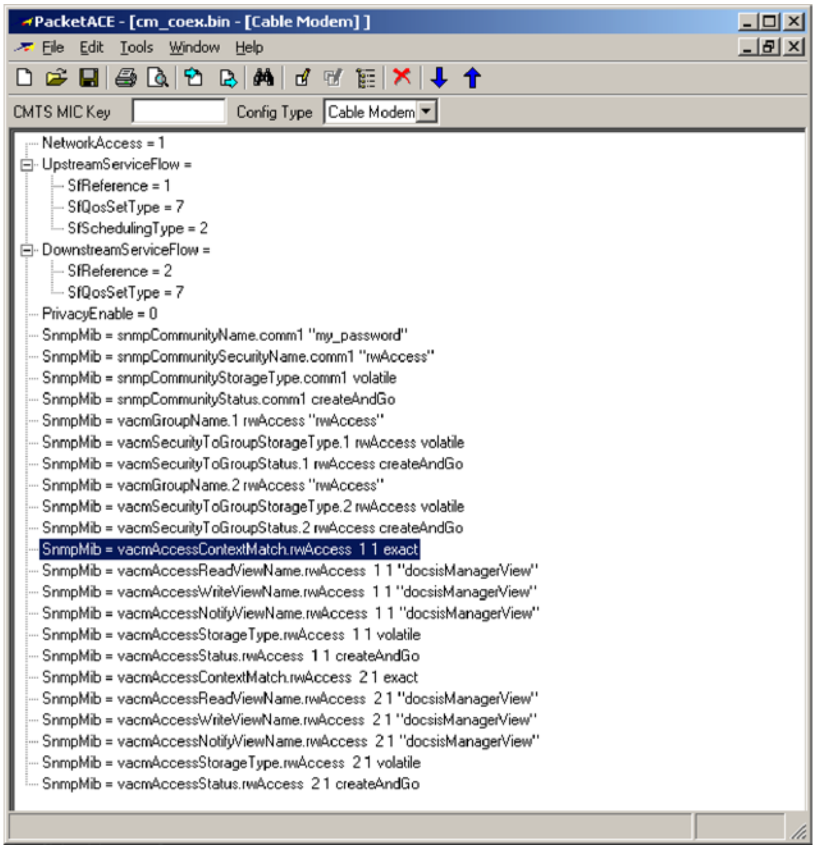
The Add SNMP MIB window appears:



4. Enter indexes as follows, and the value from the table in step 3 in the **Value** field.
   - **Index1,DisplayString**— **rwAccess** (or the value of the **vacmGroupName** object from the **vacmSecurityToGroupTable**)
   - **Index2,DisplayString**—leave blank
   - **Index3,Integer**— **1** or **2**, depending on the SNMP level row
   - **Index4,Integer**— **1** (noAuthnoPriv)

   Some of the objects have a fixed set of values; this is indicated by the drop-down menu button at the end of the **Value** field. Use the drop-down to display a list of allowed values, and choose the correct value.
5. Repeat steps 3 and 4 until all values are filled in.

The configuration file should now be similar to the following:



Note: MIB entries may be in a different order than what is shown above.

6. Proceed to see "*Configuring Trap Servers* (page 200).

Configuring Trap Servers

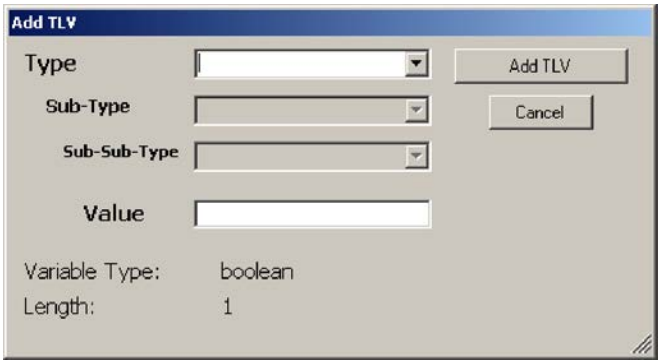
Use this procedure to configure trap destinations, using TLV-38 SNMPv3 Notification Receiver elements. The following procedure assumes that PacketACE is running and the coexistence MIBs are in the configuration file, but you can add the TLVs to the configuration file before adding the MIBs if you prefer.

Action

Follow these steps to configure trap servers.

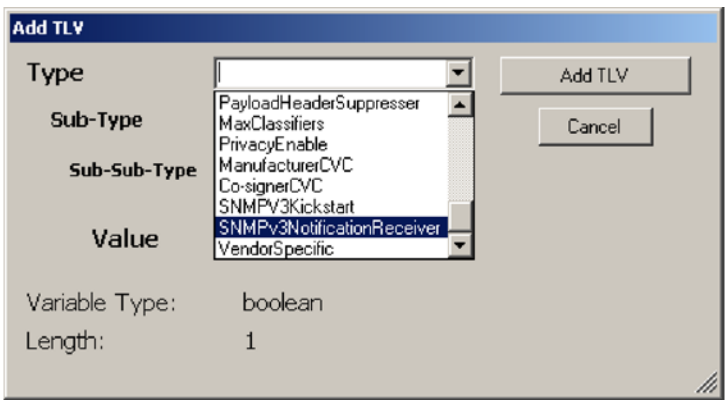
1. Click the "**Add TLV parameter**" icon or select **Edit menu → Add TLV.**

The following window appears:

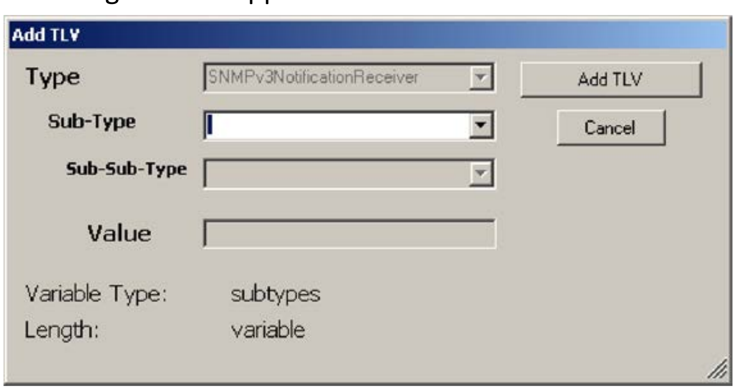


2. Select **SNMPv3NotificationReceiver** from the **Type** drop-down menu, then click **Add TLV**.

   The **SNMPv3NotificationReceiver** element appears in the main PacketACE window.



3. In the main PacketACE window, select the **SNMPv3NotificationReceiver** element, then click the "**Add TLV sub-parameter**" icon or select **Edit menu → Add Sub-TLV**. The following window appears:



4. Select **SNMPv3NrIpAddress** from the Sub-Type drop-down menu.
5. Enter the IP address of the trap server (10.1.50.100 in this example) in the Value box.
6. Click **Add TLV**.

   PacketACE adds the Trap IP address sub-type to the SNMPv3NotificationReceiver element.

7. Repeat steps 4 through 6, adding the SNMPv3NrTrapType sub-parameter and specifying a value of 1: SNMP v1 trap in an SNMP v1 packet.
8. Repeat steps 1 through 7 to create a second SNMPv3Notification-Receiver element with an IP address of 10.1.50.80 and a trap type of 2: SNMP v2c trap in an SNMP v2c packet.

   The configuration file should now resemble the following:

9. Save the configuration file and exit PacketACE.

Power Management

Touchstone TM3402 devices running TS11.1 support the TB130B and TB160B external UPS. A telemetry connection allows the Telephony Modem to monitor battery status and make use of enhanced low-power features to maximize battery hold-up time.

When the E-UE loses AC power, it starts the data shutdown timer with a default value of 5 minutes; use the **arrisMtaDevPwrSupplyDataShutdownTime** object to set the desired shutdown time (in seconds). If the timer expires before AC power is restored, the device:

- powers down Ethernet and wireless (if equipped) interfaces. To leave data services in place, set the **arrisMtaDevPwrSupplyEnableDataShutdown** object to `disabled`(2).
- switches to 1x1 unbonded mode after all lines go on-hook, informing the CMTS using a CM-STATUS message.

Once AC power is restored, the E-UE:

- activates data interfaces (as provisioned)
- attempts to restore its original bonding mode.
- clears codeword counters as a side effect

If the **arrisMtaDevPwrSupplyEnableDataShutdown** is set to `disabled`(2), the **arrisCmDoc30SetupPowerSaveWifiShutdownOnly** object allows shutting down the wifi interface on Gateway products while leaving the Ethernet interfaces active.

Touchstone Gateway products that support 802.11ac power down LAN-side interfaces immediately on loss of power, regardless of the MIB object settings. This affects the following interfaces:

- MoCA (if equipped)
- Ethernet
- Wi-Fi radios (both 2.4 GHz and 5.0 GHz)
- USB

In this case, the **arrisMtaDevPwrSupplyEnableDataShutdown** and **arrisCmDoc30SetupPowerSaveWifiShutdownOnly** objects control only when the DOCSIS interface switches to 1x1 unbonded operation. For 802.11ac-capable routers, the default time to switch to 1x1 unbonded operation is 30 seconds.

Recovery from Partial Service

The term *partial service* describes the situation where a DOCSIS 3.0 or newer device cannot acquire, or loses after acquisition, one or more of its bonded channels. Section 8.4 of CM-SP-MULPIv3.0-I12-100115 defines the recovery mechanisms to be used under the following scenarios:

When the channel...	The CM signals the CMTS using...
is not acquired during registration	REG-ACK
is not acquired during Dynamic Bonding Change	DBC-RSP
becomes unusable during normal operation	CM-STATUS

Some DOCSIS 3.0 CMTSs currently do not support all three partial service signaling methods. Use the information in this procedure to configure the Telephony Modem to work around the limitations of the CMTS, if necessary.

Consult the release notes for your CMTS firmware or software to determine whether one of the following workarounds are needed.

Action

Perform the following tasks as needed.

Identifying Partial Service Issues

Follow these steps to identify partial service issues on a Telephony Modem.

1. To identify partial service issues using an SNMP browser, read the **arrisCmDoc30ProvisionedChannelIDs** object. This object displays the channel IDs of the provisioned downstreams and upstreams. An example display is:

   ```
   DS: 8, 5, 6, (7) -- US: 5, 6, 7, 8
   ```

   This example shows that the downstream channel DCID 7 is down (enclosed in parentheses) and the modem thus has partial service.
2. To identify partial service issues using the web-based interface, access the basic Status page. This page shows the provisioned downstreams and upstreams in the “RF Parameters” section.

   The Status page indicates down channels by replacing parameters with dashes (----) for all but the provisioned frequency.

If the CMTS does not Support CM-STATUS

If the CMTS supports the REG-ACK Partial Service Confirmation Code, but not CM-STATUS message reporting, follow these steps.

1. Set the **arrisCmDoc30SetupSecDsLossReinitEnable** object to **enable**(1), using an SNMP manager or by changing the configuration file.
2. If you made the change in the configuration file, reset the Telephony Modem so the change takes effect.

   If QAM or FEC lock is lost on a secondary downstream for 45 seconds, the Telephony Modem re-initializes the downstream MAC; this allows a REG-ACK to inform the CMTS which downstreams are available.
3. When the secondary downstream becomes available, manually reset the CM to restore full service.

If the CMTS does not Support REG-ACK

If the CMTS supports bonded channels, but does not support the REG-ACK Partial Service Confirmation Code, follow these steps.

1. Set the **arrisCmDoc30SetupPartServiceFallback20** object to **`enable`**(1), using an SNMP manager or by changing the configuration file.
2. If you made the change in the configuration file, reset the Telephony Modem so the change takes effect.

   If the Telephony Modem cannot acquire a secondary downstream, it re-registers in DOCSIS 2.0 (non-bonded) mode. Every 15 minutes, the Telephony Modem re-initializes the primary downstream DOCSIS MAC until it can acquire all downstream channels. No further operator intervention is required.

ARRIS DOCSIS 3.0 MIB

The **arrisCmDoc30** MIB provides information, access control, and DHCP settings for Touchstone products supporting DOCSIS 3.0. Objects in this MIB appear in the following groups:

- **arrisCmDoc30Base**
- **arrisCmDoc30Access**
- **arrisCmDoc30Setup**
- **arrisCmDoc30Dhcp**

arrisCmDoc30Base

Miscellaneous information and control objects:

arrisCmDoc30ResetFactoryDefaults

Set to **`1`** to reset the E-UE to factory default settings.

arrisCmDoc30FwImageName

A string containing the firmware image name.

arrisCmDoc30FwImageBuildTime
A string containing the firmware build date and time. This may be useful when calling ARRIS Technical Support.

arrisCmDoc30BondingMode
The registration mode of the Telephony Modem, how many downstreams and upstreams it uses, and whether (for DOCSIS 3.1 modems) how many OFDM channels are in use. Example: `DOCSIS3.1 32x4 (2 OFDM)`

arrisCmDoc30ResetAccessTime
Set to **`true`**(1) to reset access start times in non-volatile memory.

arrisCmDoc30ProvisionedChannelIDs
Displays the channel IDs of the provisioned upstreams and downstreams. Downstreams that are not currently locked are enclosed in parentheses, indicating that the modem has partial service.

arrisCmDoc30BaseReportDuplex
Displays the duplex status of the Ethernet port; one of: **`Full`**, **`Half`**, or **`Unavailable`**.

arrisCmDoc30Access

These objects control access to the CLI and web pages.

arrisCmDoc30AccessTelnetPassword
A string containing the password used to enable Telnet.

arrisCmDoc30AccessClientSeed
A string containing the seed used to generate the Password of the Day. If you change this value, you must also change the value of the seed used in the PacketACE Password of the Day generator. Clear this value to use the default seed.

arrisCmDoc30AccessHttpLan
Controls access to the web pages from the LAN (Ethernet and USB) interfaces.

arrisCmDoc30AccessHttpWan
Controls access to the web pages from the WAN (cable) interface.

arrisCmDoc30AccessHttpTimeout
The time, in minutes, that the Advanced web pages are accessible before the Telephony Modem requires re-entry of the Password of the Day. Use **`0`** to disable the timeout.

arrisCmDoc30CLITimeout
The time, in minutes, that a Telnet or SSH session can be idle before the Telephony Modem terminates the session. Valid range: **`1`** to **`65535`** minutes, or **`0`** to disable timeout.

arrisCmDoc30AccessHttpPwCtrl
Controls which web pages are password protected:

- **`none`**(0): no pages are protected
- **`advanced`**(1): advanced pages are protected (default)
- **`all`**(2): all pages are protected

arrisCmDoc30AccessSSHEnable

Enables or disables SSH access. To enable SSH, this object must be enabled, and the **arrisCmDoc30AccessTelnetPassword** object must also be set.

arrisCmDoc30Setup

These objects control cable modem features in Touchstone products.

arrisCmDoc30SetupDSTPolicy

Sets the Daylight Savings Time (DST) policy. This object contains a string, defining the starting and ending dates and times for DST. The format of the string is as follows:

start=*month*/*day*/*weekday*/*hour***;end=***month*/*day*/*weekday*/*hour*

where...	is...
month	the month: **1** for January, to **12** for December.
weekday	The day of the week that DST begins or ends: **1** for Monday, to **7** for Sunday, or **0** to ignore the weekday and use the exact date. If not zero, DST begins or ends on the specified weekday after the date if the date is positive, or before the date if negative.
day	The day: **–31** to **–1** to count backwards from the end of the month, **1** to **31** to count forward from the beginning of the month.
hour	The hour at which DST begins or ends: **00** to **23**.

Example:

```
start=3/8/7/02;end=11/1/7/02
```

Implements the U.S. DST policy in effect since March 2007: DST begins at 2 a.m. on the second Sunday in March and ends at 2 a.m. on the first Sunday in November.

arrisCmDoc30SetupTODTimeOffset

Sets the CM ToD time offset from GMT, in seconds. Valid range: **–43200** (–12 hours) to **46800** (+13 hours). This value may be overwritten when the TOD offset option is received through DHCP.

arrisCmDoc30SetupTODSyncTimeout

The interval, in hours, between ToD sync operations. Valid range: **0** (default, disable sync completely) to **4320**.

Setting this object restarts the timer.

arrisCmDoc30SetupIgnoreMddSymbolClockIndicator

When set to **enable**(1), the CM ignores the Symbol Clock Locking Indicator TLV in the MDD message and always uses asynchronous timing mode. In this mode, the cable modem does not lock to the downstream symbol clock on its Primary Downstream

Channel. It acquires the synchronization timebase for upstream burst timing from the SYNC messages.

When set to **disable**(0), the CM follows the timing mode specified by the Symbol Clock Indicator TLV in the MDD message.

arrisCmDoc30SetupTftpBlkSize

Controls the TFTP Blocksize Option, defined in RFC 2348, for firmware downloads. Valid settings:

- 0: use the default block size of 1428 for IPv6 connections or 1448 for IPv4 connections.
- 64–65464: the TFTP block size, in octets.

Note: Specifying extremely high or low values may affect TFTP performance.

arrisCmDoc30SetupTftpTimeout

Controls the TFTP Timeout Option, defined in RFC 2349, for firmware downloads. Valid settings:

- 0 (default): disable this option.
- 1–255: when smart download mode (**arrisCmDoc30EsafeSmartDownloadMode**) is enabled, the TFTP client uses the specified timeout value.

arrisCmDoc30SetupRCPBypass

Controls the RCP check bypass feature. Valid settings:

- **disable**(0): disables RCP check bypass, and overrides any setting of this object in the CM configuration file. If set in an SNMP browser, the value is persistent.
- **enable**(1): enables RCP check bypass, and overrides any setting of this object in the CM configuration file. If set in an SNMP browser, the value is persistent.
- **default**(2): disables RCP check bypass, unless specified otherwise in the CM configuration file. This value is not persistent.

arrisCmDoc30SetupTimeoffsetWrapper

Controls the Time Offset Wrapper feature. Valid settings:

- **disable**(0): disables Time Offset Wrapper, and overrides any setting of this object in the CM configuration file. If set in an SNMP browser, the value is persistent.
- **enable**(1): enables Time Offset Wrapper, and overrides any setting of this object in the CM configuration file. If set in an SNMP browser, the value is persistent.
- **default**(2), disables Time Offset Wrapper, unless specified otherwise in the CM configuration file. This value is not persistent.

arrisCmDoc30DiplexerControl

Displays or sets the diplexer. Valid settings:

- **band0**(0)
- **band1**(1)

Use the **arrisCmDoc30DiplexerFreqencyRanges** object to view the frequency ranges for each diplexer.

Note: You must reset the device to allow the diplexer setting to take effect.

arrisCmDoc30DiplexerFreqencyRanges

Displays the upstream and downstream frequency ranges for each diplexer.

Example: `Band0: 5-85MHz/108-1002MHz; Band1: 5-42MHz/108-1002MHz`

arrisCmDoc30AppSoftware

These objects provide information about the TFTP server used for firmware downloads.

arrisCmDoc30AppSwOperStatus

(read-only) The TFTP download status, one of:

- `inProgress`(1): A TFTP download is currently underway.
- `failed`(2): The last attempted download failed.
- `incorrectCRC`(3): The image to download has either a bad CRC.
- `incorrectType`(4): The **arrisMgSwImageType** does not match the hardware.
- `other`(5): The default state when arrisMgSw objects are not used.

arrisCmDoc30Dhcp

These objects provide DHCP status information.There are four groups under this MIB:

- **arrisCmDoc30DhcpLeaseParameters**
- **arrisCmDoc30DhcpSvrParameters**
- **arrisCmDoc30DhcpCmParameters**
- **arrisCmDoc30DhcpMtaParameters**

arrisCmDoc30DhcpLeaseParameters Objects

These objects provide information about the current DHCP lease.

arrisMtaDoc30DhcpOfferedLeaseTime

The offered lease time, in seconds.

arrisCmDoc30DhcpTimeUntilRenew

The time, in seconds, before the eDVA begins a RENEW exchange.

arrisCmDoc30DhcpTimeUntilRebind

The time, in seconds, before the eDVA begins a REBIND exchange.

arrisCmDoc30DhcpLeaseTimeRemaining

The remaining lease time, in seconds.

arrisCmDoc30DhcpSvrParameters Objects

These objects provide information about the DHCP server. All these objects are read-only.

arrisCmDoc30DhcpState

The current CM DHCP state; one of:

- `init-selecting`(0)
- `requesting`(1)
- `bound`(2)
- `renewing`(3)
- `rebinding`(4)
- `init-reboot`(5)
- `renew-requested`(6)
- `released`(7)

arrisCmDoc30DhcpPrimaryDhcpServerIpAddr

The primary DHCP server address.

arrisCmDoc30DhcpTftpSvrIpAddr

The current TFTP server IP address in use.

arrisCmDoc30DhcpTimeSvrIpAddr

The current Time server IP address in use.

arrisCmDoc30DhcpCmTimeOffset

The current time offset being used by the CM.

arrisCmDoc30DhcpPrimaryTeleDhcpSvr

The primary eDVA DHCP server address.

arrisCmDoc30DhcpSecondaryTeleDhcpSvr

The secondary eDVA DHCP server address.

arrisCmDoc30DhcpCmParameters Objects

These objects provide information about the current CM DHCP lease. All these objects are read-only.

arrisCmDoc30DhcpCmIpAddrType

The type (IPv4 or IPv6) of the currently leased IP address.

arrisCmDoc30DhcpCmIpAddr

The currently leased IP address.

arrisCmDoc30DhcpCmSubNetMask

The current IP subnet mask in use.

arrisCmDoc30DhcpCmGatewayIpAddr

The current IP gateway address in use.

arrisCmDoc30DhcpCmConfigFile

The CM configuration file to be retrieved.

arrisCmDoc30DhcpMtaParameters Objects

These objects provide information and control over eDVA DHCP operation.

arrisCmDoc30DhcpMtaOpt60Override
Enables or disables SIP advertisement in eDVA DHCP Option 60. The default is disabled. This object can be set only using the CM configuration file.

arrisCmDoc30DhcpExtended

These objects provide extended DHCP information. All these objects are read-only.

`arrisCmDoc30DhcpExtendedProvisionedMode`
Displays the CM provisioned mode:

- **`ipv4-only`**(0)
- **`ipv6-only`**(1)
- **`alternate-prov-mode`**(2)
- **`dual-prov-mode`**(3)

arrisCmDoc30DhcpExtendedPreferredMode
Displays the CM preferred IP mode: **`ipv4`**(0) or **`ipv6`**(1).

arrisCmDoc30DhcpExtendedActiveMode
Displays the CM active IP mode: **`ipv4`**(0) or **`ipv6`**(1).

arrisCmDoc30DhcpExtendedLeaseParametersTable Objects

This table provides DHCP lease information.

arrisCmDoc30DhcpExtendedLeaseParametersType
The DHCP type: **`dhcpv4`**(0) or **`dhcpv6`**(1).

arrisCmDoc30DhcpExtendedOfferedLeaseTime
The offered IP lease time, in seconds.

arrisCmDoc30DhcpExtendedTimeUntilRenew
The current time remaining, in seconds, before the eDVA starts the lease renewal process.

arrisCmDoc30DhcpExtendedTimeUntilRebind
The current time remaining, in seconds, before the CM starts the lease rebinding process.

arrisCmDoc30DhcpExtendedLeaseTimeRemaining
The remaining IP lease time, in seconds.

arrisCmDoc30DhcpExtendedRenewLease
Set to **`apply`**(1) to renew the WAN(0) DHCP lease.

arrisCmDoc30DhcpExtendedSvrParametersTable

This table provides extended information about the DHCP server.

arrisCmDoc30DhcpExtendedSvrParametersType
The DHCP type: **dhcpv4**(0) or **dhcpv6**(1).

arrisCmDoc30DhcpExtendedState
The current DHCP state of the CM:

- **`init-selecting`**(0)
- **`requesting`**(1)
- **`bound`**(2)
- **`renewing`**(3)
- **`rebinding`**(4)
- **`init-reboot`**(5)
- **`renew-requested`**(6)
- **`released`**(7)
- **`dhcp6c-init`**(8)
- **`dhcp6c-solicit`**(9)
- **`dhcp6c-inforeq`**(10)
- **`dhcp6c-request`**(11)
- **`dhcp6c-renew`**(12)
- **`dhcp6c-rebind`**(13)
- **`dhcp6c-release`**(14)
- **`dhcp6c-decline`**(15)
- **`dhcp6c-confirm`**(16)
- **`dhcp6c-idle`**(17)

arrisCmDoc30DhcpExtendedPrimaryDhcpServerIpAddr
The primary DHCP server address.

arrisCmDoc30DhcpExtendedTftpSvrIpAddr
The current TFTP server IP address in use.

arrisCmDoc30DhcpExtendedTimeSvrIpAddr
The current time server IP address in use.

arrisCmDoc30DhcpExtendedCmTimeOffset
The current time offset used by the CM.

arrisCmDoc30DhcpExtendedPrimaryTeleDhcpSvr
The primary eDVA DHCP server address.

arrisCmDoc30DhcpExtendedSecondaryTeleDhcpSvr
The secondary eDVA DHCP server address.

arrisCmDoc30DhcpExtendedSrvDUIDV6
(read-only) The server DUID.

arrisCmDoc30DhcpExtendedCmParametersTable Objects

This table provides extended DHCP information for the CM.

arrisCmDoc30DhcpExtendedCmParametersType
The DHCP type: **dhcpv4**(0) or **dhcpv6**(1).

arrisCmDoc30DhcpExtendedCmIpAddr
The currently leased IP address.

arrisCmDoc30DhcpExtendedCmSubNetMask
The current IP subnet mask.

arrisCmDoc30DhcpExtendedCmPrefix
The current IP Prefix.

arrisCmDoc30DhcpExtendedCmGatewayIpAddr
The current IP gateway address.

arrisCmDoc30DhcpExtendedCmConfigFile
The CM configuration file to be retrieved.

arrisCmDoc30DhcpExtendedCmDUIDV6
The CM DUID.

arrisCmDoc30DhcpExtendedDnsSvrTable

This table provides extended information for DNS servers.

arrisCmDoc30DhcpExtendedDnsSvrIPAddrType
The IP address type of the DNS server: **dhcpv4**(0) or **dhcpv6**(1).

arrisCmDoc30DhcpExtendedDnsSvrIPAddr
The IP address of the DNS server.

arrisCmDoc30ResetReasonLog

These objects contain the reset reason log. All these objects are read-only.

arrisCmDoc30LastHwResetReason
The last reset reason retrieved from the processor hardware. This reason is used to determine if hardware or firmware caused the reset.

arrisCmDoc30ResetReasonLogTable Objects

This table contains the last ten reset reasons. The valid index range is **1** to **10**.

arrisCmDoc30ResetReasonLogText
The reported reset reason log text.

arrisCmDoc30Software

See Using Enhanced Firmware Loading for a description of objects in this branch.

DOCSIS 3.0 MIB Object Mapping

The following table shows how to map certain ARRIS DOCSIS 2.0 MIB objects to their DOCSIS 3.0 counterparts.

DOCSIS 2.0 Object/OID	DOCSIS 3.0 Object/OID
arrisCmProdResetToFactoryDefaults	arrisCmDoc30ResetFactoryDefaults
1.3.6.1.4.1.4115.1.3.2.2.15	1.3.6.1.4.1.4115.1.3.4.1.1.6
arrisCmDevSwImageName	arrisCmDoc30FwImageName
1.3.6.1.4.1.4115.1.3.1.1.1.2	1.3.6.1.4.1.4115.1.3.4.1.1.7
arrisCmDevSwImageBuildTime	arrisCmDoc30FwImageBuildTime
1.3.6.1.4.1.4115.1.3.1.1.1.3	1.3.6.1.4.1.4115.1.3.4.1.1.8
arrisCmProdAccessPWD	arrisCmDoc30AccessTelnetPassword
1.3.6.1.4.1.4115.1.3.2.2.2	1.3.6.1.4.1.4115.1.3.4.1.2.1
arrisCmDevTelnetEnable	arrisCmDoc30AccessTelnetEnable
1.3.6.1.4.1.4115.1.3.1.1.2.3.23	1.3.6.1.4.1.4115.1.3.4.1.2.2
arrisCmDevHttpClientSeed	arrisCmDoc30AccessClientSeed
1.3.6.1.4.1.4115.1.3.1.1.2.3.5.5	1.3.6.1.4.1.4115.1.3.4.1.2.3
arrisCmDevHttpLanAccess	arrisCmDoc30AccessHttpLan
1.3.6.1.4.1.4115.1.3.1.1.2.3.5.3	1.3.6.1.4.1.4115.1.3.4.1.2.5
arrisCmDevHttpWanAccess	arrisCmDoc30AccessHttpWan
1.3.6.1.4.1.4115.1.3.1.1.2.3.5.4	1.3.6.1.4.1.4115.1.3.4.1.2.6
arrisCmDevSwCustomerLoadId	arrisCmDoc30SwCustomerLoadId
1.3.6.1.4.1.4115.1.3.3.1.4.1.1.2	1.3.6.1.4.1.4115.1.3.4.1.5.1.1.2
arrisCmDevSwHwModel	arrisCmDoc30SwHwModel
1.3.6.1.4.1.4115.1.3.3.1.4.1.1.3	1.3.6.1.4.1.4115.1.3.4.1.5.1.1.3
arrisCmDevSwHwRev	arrisCmDoc30SwHwRev
1.3.6.1.4.1.4115.1.3.3.1.4.1.1.4	1.3.6.1.4.1.4115.1.3.4.1.5.1.1.4
arrisCmDevSwFilename	arrisCmDoc30SwFilename
1.3.6.1.4.1.4115.1.3.3.1.4.1.1.6	1.3.6.1.4.1.4115.1.3.4.1.5.1.1.5
arrisCmDevSwServerAddressType	arrisCmDoc30SwServerAddressType
1.3.6.1.4.1.4115.1.3.3.1.4.1.1.7	1.3.6.1.4.1.4115.1.3.4.1.5.1.1.6
arrisCmDevSwServerAddress	arrisCmDoc30SwServerAddress
1.3.6.1.4.1.4115.1.3.3.1.4.1.1.8	1.3.6.1.4.1.4115.1.3.4.1.5.1.1.7

Supported eDVA MIB Objects

TS11.1 supports the following ARRIS eDVA MIB objects.

PACKETPORT-MIB Objects

The following objects are supported:

ppCfgMtaCountryTemplate
Specifies the country template for ring cadences and signaling tones.

ppCfgMtaCallpFeatureSwitch
The CallP Feature Switch. See *CallP Feature Switch* (page 46) for details.

ppCfgRfc2833DigitPayloadType
When bit 0x01000000 of the CallP Feature Switch is set, this object specifies the payload type used to send DTMF events. Valid range: **97** to **127**. Default: **101**.

ppCfgMtaFeatureSwitch
The MTA Feature Switch.

ppCfgMtaCallpFeatureSwitch2
The secondary CallP Feature Switch. See *CallP Feature Switch* (page 46) for details.

ppCfgPortTable Objects

The following objects are supported.

ppCfgPortLoopCurrent
Sets the loop current for the specified line: **`normal`**(1) or **`high`**(2).

ppCfgPortLocUserIndication
The tone applied during Loss of Communication (LoC): **`silence`**(0) or **`reorderTone`**(1) (default).

ppCfgPortT38MaxDatagram
The maximum datagram size for incoming T.38 packets. Valid range: **`160`** (default) to **`65535`**. If this object is set to a value higher than the default, the Call Agent must allocate more bandwidth accordingly.

ppSurvPortTable Objects

The following objects are supported.

ppSurvPortMaintState
The maintenance state of the line.

ppSurvPortLcDiagRequest
Set to **`true`**(2) to begin linecard diagnostics.

ppSurvPortLcDiagLastResult

The last result of linecard diagnostics performed on the line.

ARRIS-MTA-MIB (non-battery)

The following MIB objects from the ARRIS-MTA-MIB are supported. Battery-related objects are listed below.

arrisMtaDevControl Objects

The following **arrisMtaDevControl** objects are supported:

arrisMtaDevResetCallStats

Set this object to `true`(1) to reset the following objects to their default values:

- **arrisMtaDevRtpTxPktsTotal**
- **arrisMtaDevRtpRxPktsTotal**
- **arrisMtaDevRtpPercentPktsLostTotal**
- **arrisMtaDevSignalingAvgLatency**
- **arrisMtaDevSignalingTxSuccessfulMsgCnt**
- **arrisMtaDevSignalingRxSuccessfulMsgCnt**
- **arrisMtaDevSignalingTxNAKCnt**
- **arrisMtaDevSignalingRxNAKCnt**
- **arrisMtaDevSignalingRxNoACKCnt**

Setting this object to `false`(2) does nothing. Reading this object always returns `false`(2).

arrisMtaDevEnableCallpSigTrace

Controls CallP signaling message tracing in the Syslog. Take care when setting this object, as excessive messaging could adversely affect performance. The default value is `disable`(0).

arrisMtaDevEnableCallStatsSyslogRpt

Enables end-of-call statistics reporting, and CallP signaling last message reporting, to the Syslog.

When set to `enable`(1), end-of-call statistics are reported in the Syslog. If the **arrisMtaDevEnableCallSigLastMsgRpt** object is enabled, then the last 4K of signaling messages is also reported in the Syslog.

When set to `disable`(0) (the default), end-of-call statistics and the CallP last signaling messages are not reported in the Syslog.

arrisMtaDevSwDnldNoSvcImpact

Enables or disables the software download service impact feature. When set to `enable`(1) (the default), the eDVA accepts the load, but does not apply the load until all lines have been idle for at least 30 seconds after the load has been accepted.

arrisMtaDevEnableCallSigLastMsgRpt

Enables or disables reporting of the CallP signaling ''last message'' to the MIB objects **arrisMtaDevSignalingLastMsg1** through **arrisMtaDevSignalingLastMsg16**. Together the sixteen objects contain a signaling message up to 4000 bytes long. Each object contains a 255-byte segment of the message. If the message does not require all sixteen MIBs, then the empty objects display ''Buffer is empty.'' The default value is **`disable`**(0).

arrisMtaDevNsadSwDnldStatus

(read-only) Displays the current firmware download status:

`download-Idle`(0)
: Indicates that the firmware download has completed. This value is also set at startup.

`download-Acceptance-In-Progress`(1)
: The unit is currently downloading the firmware in the background.

`download-Application-Pending`(2)
: Indicates that the load has been downloaded and accepted but is waiting to be applied.

This MIB object is only valid if the **arrisMtaDevSwDnldNoSvcImpact** object is set to **`enabled`**(1).

arrisMtaDevRestoreNvmFactoryDefault

Set this object to **`true`**(1) to reset the NVM to default values.

arrisMtaDevTrace Objects

These objects control and display message trace results.

arrisMtaDevRtpTxPktsTotal

(read-only) The total number of RTP packets sent from the eDVA since it was last started up or reset. This value represents the total number of packets sent for all endpoints combined. It is calculated from values obtained from the signaling end of call statistics. Set the **arrisMtaDevResetCallStats** object to **`true`**(1) to clear this counter.

arrisMtaDevRtpRxPktsTotal

(read-only) The total number of RTP packets received by the eDVA since it was last started up or reset. This value represents the total number of packets received for all endpoints combined. It is calculated from values obtained from the signaling end of call statistics. Set the **arrisMtaDevResetCallStats** object to **`true`**(1) to clear this counter.

arrisMtaDevRtpPercentPktsLostTotal

(read-only) The percentage of RTP packets lost since the eDVA was last started up or reset. This value represents the total number of packets lost for all endpoints combined. It is calculated from values obtained from the signaling end of call statistics. Set the **arrisMtaDevResetCallStats** object to **`true`**(1) to clear this counter.

The value of this object is expressed in increments of 1/100 percent. For example, a value of **`1745`** means that 17.45% of the packets were lost.

arrisMtaDevRtpPktsLostTotal
(read-only) The number of RTP packets lost since the MTA was last started up or reset. This value represents the total number of packets lost for all endpoints combined. It is calculated from values obtained from the signaling end of call statistics. Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

arrisMtaDevLastCallStartTime
(read-only) The last call start time from the eDVA.

arrisMtaDevLastCallEndTime
(read-only) The last call end time from the eDVA.

arrisMtaDevSignalingAvgLatency
(read-only) The average latency or delay, in milliseconds, for responses to signaling messages. It is calculated from values obtained from the signaling end of call statistics. Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

arrisMtaDevSignalingTxSuccessfulMsgCnt
(read-only) The total number of successful signaling messages sent from the eDVA. Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

arrisMtaDevSignalingRxSuccessfulMsgCnt
(read-only) The total number of successful signaling messages received by the MTA. Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

arrisMtaDevSignalingTxNAKCnt
(read-only) The total number of negative acknowledgement signaling messages (NAKmessages) sent from the MTA. Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

arrisMtaDevSignalingRxNAKCnt
(read-only) The total number of negative acknowledgement signaling messages (NAKmessages) received by the MTA.Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

arrisMtaDevProvState
(read-only) The current provisioning state of the MTA:

- `dhcpBound`(1)
- `dnsReqProvSvrIP`(2)
- `kdcHostNameDnsReq`(3)
- `kdcHostNameDnsRply`(4)
- `kdcIpDnsReq`(5)
- `kdcIpDnsRply`(6)
- `asReqSent`(7)
- `asRplyRcvd`(8)
- `tgsReqSent`(9)
- `tgsRplyRcvd`(10)
- `apReqSent`(11)
- `apRplyRcvd`(12)

- **`enrollmentInform`**(13)
- **`cfgUrlSet`**(14)
- **`dnsReqTftpSvrIp`**(15)
- **`cfgFileReq`**(16)
- **`rcvCfgFile`**(17)
- **`syslogMsgProvComplete`**(18)
- **`statusInform`**(19)
- **`provcomplete`**(20)

arrisMtaDevSWUpgradeStatus

(read-only) The current software upgrade status of the device:

- **`inProgress`**(1)
- **`completeFromProvisioning`**(2)
- **`completeFromMgt`**(3)
- **`failed`**(4)
- **`other`**(5)

arrisMtaDevSignalingRxNoACKCnt

(read-only) The total number of 'no acknowledgement' signaling messages received by the MTA. Set the **arrisMtaDevResetCallStats** object to **`true`**(1) to clear this counter.

arrisMtaDevSignalingLastMsg1–16

(read-only) These objects contain a 255-byte segment of the CallP last signaling message sent or received. The sixteen objects together can display a signaling message as large as 4000 bytes. If the message does not require all sixteen objects, then the empty objects display the value ''Buffer is empty.'' Use **arrisMtaDevEnableCallSigLastMsgRpt** to enable or disable reporting of the CallP signaling last message.

arrisMtaDevCallStatsTable

This table reports various end of call statistics. All objects in this table are read-only, and are indexed by line number.

arrisMtaDevCallStatsRtpTxPkts

The total number of RTP packets sent from the endpoint during the most recent call. This value is obtained from the signaling end-of-call statistics.

arrisMtaDevCallStatsRtpRxPkts

The total number of RTP packets received by the endpoint during the most recent call. This value is obtained from the signaling end-of-call statistics.

arrisMtaDevCallStatsRtpPercentPktsLost

The percentage of RTP packets lost during the most recent call. This value is obtained from the signaling end-of-call statistics.

The value of this object is expressed in increments of 1/100 percent. For example, a value of **`1745`** means that 17.45% of the packets were lost.

arrisMtaDevCallStatsAvgJitter

The average jitter measurement, in milliseconds, during the most recent call. This value is obtained from the signaling end-of-call statistics.

arrisMtaDevCallStatsMaxJitter

The maximum jitter measurement, in milliseconds, during the most recent call. This value is obtained from DSP statistics.

arrisMtaDevCallStatsAvgLatency

The average latency, in milliseconds, observed during the most recent call. This value is obtained from the RTCP signaling end-of-call statistics.

arrisMtaDevCallStatsHookStatus

The hook status for each endpoint: **onHook**(0) or **offHook**(1).

Note: The disconnected state is not a valid return value for this MIB object.

arrisMtaDevCallStatsSLICStatus

The over temperature condition of the SLIC chips: **normal**(0) or over **temp**(1).

arrisMtaDevCallStatsEndPntOpStatus

The current operational status for each endpoint: **up**(1) (ready to pass packets), **down**(2), or **testing**(3) (in some test mode). This object obtains its value from **ifAdminOperStatus**.

arrisMtaDevCallStatsLineSubState

The current sub-state for each line: **normal**(0), **diagsPending**(1), **diagsFailed**(2), **lcProtection**(3), or **dspFail**(4). This object obtains its value from the lineRec.

arrisMtaDevParameters

These objects provide information about E-UE and line parameters. All objects in this group are read-only.

arrisMtaDevMaxCpeAllowed

Reflects the ''MaxCpeAllowed'' parameter in the CM config file.

arrisMtaDevNetworkAccess

Reflects the ''NetworkAccess'' parameter set in the CM config file: **no**(0) or **yes**(1).

arrisMtaDevQosMode

Reflects the setting of the DSx DQoS bit (0x00004000) in the CallP Feature Switch: **bestEffort-FullDQos-PCMM**(0) or **dsxMode**(1).

arrisMtaDevEventFormat

The PacketCable event format in use: **pktc10**(0) (PacketCable 1.0) or **pktc15**(1) (PacketCable 1.5).

arrisMtaDevLineParameterTable

This table reports various line parameters on the eMTA.

arrisMtaDevInterfaceIndex

The **ifIndex** object for a particular line.

arrisMtaDevPktcDevEvEndpointName

The endpoint name in the following format: **AALN/** *`line`*:*`FQDN`***/** *`ipaddr`*.

arrisMtaDevActiveConnections

The number of active connections for a particular line. Active connections include receive-only, send-only, and send/receive; but not inactive.

arrisMtaDevLineMWIActive

The MWI status for the line.

arrisMtaDevLineRTCPXR

Indicates whether or not RTCP-XR is configured: **`disabled`**(0) or **`enabled`**(1). The setting is based on the value of the **arrisMtaDevVqmEnableRemote** and **pktcEDVACodecRTCPXR** objects according to the following table:

arrisMtaDevVqmEnableRemote	pktcEDVACodecRTCPXR(1)	arrisMtaDevLineRTCPXR		
		PC2.0	LegacySIP	NCS
normal	true	enabled	disabled	disabled
normal	false	disabled(2)	disabled(2)	disabled(3)
forceDisable	true	disabled	disabled	disabled
forceDisable	false	disabled	disabled	disabled
forceEnable	true	enabled	enabled	enabled
forceEnable	false	enabled	enabled	enabled

(1)The **pktcEDVACodecRTCPXR** object is applicable only to PC2.0.

(2)Though **arrisMTADevLineRTCPXR** is set to disabled, the remote may still request RTCP-XR to be transmitted which will be honored.

(3) Though **arrisMTADevLineRTCPXR** is set to disabled, the Call Agent may instruct the MTA to request RTCP-XR through NCS signaling.

arrisMtaDevLineRTCPXRNegotiatedConnectionA

The negotiation status of RTCP-XR for a particular line's A (i.e. first) connection. The status is one of the following:

`callNotInProgress`(0)

no active call is utilizing this connection/call leg.

`notNegotiated`(1)

RTCP-XR is not negotiated to be sent on this call leg.

`negotiated`(2)

RTCP-XR is negotiated to be sent on this call leg.

arrisMtaDevLineRTCPXRNegotiatedConnectionB
The negotiation status of RTCP-XR for a particular line's B (i.e. second) connection. The status is identical to the A connection above.

arrisMtaDevUpSvcFlowParameterTable

This table reports various Upstream Service Flow parameters on the eMTA.

arrisMtaDevDocsQosParamUpSvcFlowSFID
The upstream service flow SFID for a valid service flow index.

arrisMtaDevDocsQosParamUpSvcFlowSchedulingType
The upstream service flow scheduling type for a valid service flow index.

arrisMtaDevVqm Objects

The **arrisMtaDevVqm** MIB provides objects for controlling the Voice Quality Monitoring (VQM) feature and for retrieving VQM data.

The following objects are available.

arrisMtaDevVqmLine
Specifies the line for Voice Quality Metrics reporting.

arrisMtaDevVqmClear
Clears Voice Quality Metrics: **`single-line`**(0) or **`all-lines`**(1). To clear a single line, specify the line using **arrisMtaDevVqmLine**.

arrisMtaDevVqmEnable
Enables or disables recording of Local Voice Metrics: **`disable`**(0) or **`enable`**(1) (the default).

arrisMtaDevVqmCallNumberTable
Voice Quality Metric history on a per call number basis. This table contains up to ten entries, consisting of the following object:

arrisMtaDevVqmCallNumberIds
Provides a history of call number identifiers for the specified line.

arrisMtaDevVqmCallNumberIdentifier
Specifies the call number for Voice Quality Metrics reporting. Valid IDs are obtained from **arrisMtaDevVqmCallNumberIds**.

arrisMtaDevVqmMetricTable
Voice Quality Metrics pertaining to a particular call number. Entries are indexed by metric number, and contain the following objects:

arrisMtaDevVqmMetricValues
The call data. Walk the **arrisMtaDevVqmMetricTable** to retrieve the data in the buffer.

arrisMtaDevVqmThresholds

The thresholds associated with this metric. The eDVA generates a Syslog message if VQM data scores fall below the specified thresholds.

arrisMtaDevVqmEnableRemote

Determines the policy to control Remote Voice Metrics (XR):

- **normal**(0) (default) — obey directives from the Call Server.
- **forceDisable**(1) — disables XR, overriding any directive from the Call Server.
- **forceEnable**(2) — enables XR, overriding any directive from the Call Server.
- **default**(3) — resets this object to the default and removes persistence.

The value of this object is persistent if it is set after configuration.

arrisMtaDevVqmThresholdEnable

The maximum number of logs allowed to be sent in a report when a threshold is exceeded. The more logs that are allowed, the more metrics that can be reported. A value of **0** (default) disables threshold reporting.

The eDVA reports metrics in the order specified by the **arrisMtaDevVqmMetricIndex**. To minimize network traffic, logs are sent only when a threshold is first exceeded. Thereafter until the metric returns to normal, logs are inhibited. To send logs each time the threshold is exceeded, add 100 to the value.

See see "*Setting VQM Thresholds* (page 264) for more details.

arrisMtaDevVqmHistorySize

Sets the size of the VQM history buffer. Valid range: **2** to **50** records. Default: **50**.

arrisMtaDevVqmCallNumberIdentifierLastCall

(read-only) The last call identifier for Voice Quality Metrics reporting.

arrisMtaDevOperationalSetup Objects

arrisMtaDevVPNomJitterBuffer

The Voice Playout nominal jitter buffer size, in terms of packetization rate: **packetizationRatex1**(1) (default), **packetizationRatex2**(2), **packetizationRatex3**(3), or **packetizationRatex4**(4).

arrisMtaDevVPJitterBufferMode

The Voice Playout jitter buffer mode: **adaptive**(1) (default) or **fixed**(2).

arrisMtaDevRTPTxQueueSize

Sets the RTP transmit queue size. Valid range: **2** to **4**. Default: **2**.

arrisMtaDevEchoCancellerTailLength

The length, in milliseconds, of the echo canceller tail: **eightMs**(1) or **thirtyTwoMs**(2) (default). This object can be set only in the eDVA configuration file.

arrisMtaDevDspHandleNonPhaseReversedTone

Configures handling of the DSP non-phase reversed tone detection:

off(1)

ignores tone detection.

onECANEnable(2)
(default) the non-phase reversed CED tone is handled with the echo canceler enabled.

onECANDisabled(3)
the non-phase reversed CED tone is handled with the echo canceler disabled.

arrisMtaDevProvMethodIndicator
The method used to provision the device. This object can be set only in the configuration file. The following provisioning methods are supported:

docsisOnly(0)
DOCSIS-only provisioning.

fullPacketCable(1)
PacketCable 1.5 flows. The **arrisMtaDevPacketcableProvisioningFlow** object specifies which flow is in use.

packetCableMinusKDC(2)
same as full PacketCable, except with IPSEC and SNMPv3 disabled.

singleMAC(5)
single configuration file (SNMPv2, single IP address, single MAC address, no SNMP Informs, IPsec disabled).

basic1(6)
Basic1 SNMPv2, without SNMP Enrollment, Status Informs, or Hash.

basic2(7)
ARRIS Basic2 SNMPv2, without SNMP Enrollment Inform or Hash.

arrisMtaCfgRTPDynPortStart
The starting value for a range of ports that is used dynamically when sending out SIP RTP voice packets. This object and **arrisMtaCfgRTPDynPortEnd** define the port range.

Valid range: **1024** to **65535**. Default: **49152**.

arrisMtaCfgRTPDynPortEnd
The ending value for a range of ports that is used dynamically when sending out SIP RTP voice packets. The value of this object must be higher than **arrisMtaCfgRTPDynPortStart**.

Valid range: **1024** to **65535**. Default: **65535**.

arrisMtaDevVPMaxJitterBuffer
Indicates the Voice Playout maximum jitter buffer: **packetizationRatex1**(1), **packetizationRatex2**(2), **packetizationRatex3**(3) (default), or **packetizationRatex4**(4).

arrisMtaDevPacketcableProvisioningFlow
(read-only) Indicates the PacketCable simplified provisioning flow for PacketCable 1.5 compliance:

secure(0)
PacketCable 1.5 Secure Flow

hybrid2(1)
PacketCable 1.5 Hybrid 2 Flow

hybrid1(2)
PacketCable 1.5 Hybrid 1 Flow

basic2(3)
PacketCable 1.5 Basic 2 Flow

basic1(4)
PacketCable 1.5 Basic 1 Flow

none(5)
ARRIS non-PacketCable 1.5 Flow

arrisMtaDevEnableIndexTenEleven
Set to **enable**(1) to use 10 and 11 as the **ifIndex** for lines 1 and 2. The default behavior uses 9 and 10 for the **ifIndex**.

arrisMtaDevDspCpsSetting
Enables or disables ECAN fast constant power signal detection. The default value of **on**(2) makes the echo canceller disengage immediately when a high level constant power signal is detected.

arrisMtaDevVbdOverwriteLineBitmap
A bitmask that defines which lines use the VbdOverwriteJitterBuffer values for fax/modem jitter buffer settings. A value of **0** (the default) affects no lines; **0x01** enables line 1, and so on.

arrisMtaDevVbdOverwriteMinJitterBuffer
When **arrisMtaDevVbdOverwriteLineBitmap** is set for the line, this value is used as the minimun jitter buffer setting in all modem/fax calls. Valid range: **10** to **160**. Default: **20**.

arrisMtaDevVbdOverwriteNomJitterBuffer
When **arrisMtaDevVbdOverwriteLineBitmap** is set for the line, this value is used as the nominal jitter buffer setting in all modem/fax calls. Valid range: **10** to **160**. Default: **20**.

arrisMtaDevVbdOverwriteMaxJitterBuffer
When **arrisMtaDevVbdOverwriteLineBitmap** is set for the line, this value is used as the maximum jitter buffer setting in all modem/fax calls. Valid range: **10** to **160**. Default: **20**.

arrisMtaDevEventHideFQDNandIPAddress
Set to **enable**(1) to hide the MTA FQDN and IP address on the Event Log web page. Logging into the Advanced pages allows an operator to view this information.

arrisMtaDevDhcpOptionOverride
Set to **on**(2) to disable DHCP option code 122/177 sub-option 3 value enforcement.

The default value, **off**(1), enforces DHCP option 122/177 suboption 3 value comparison checking. Typically, the value received in the DHCP OFFER should not change in the DHCP ACK. DHCP RENEW/REBIND values should be consistent with the value received in the DHCP OFFER.

arrisMtaDevDefaultReasonNoCIDName

The reason sent to the CPE when the Caller ID Name is not included in the signal request. The default value is **unavailable**(0). The following table shows the reason sent for each setting.

Value	Reason	Description
unavailable(0)	‘O’	Out of area
private(1)	‘P’	Private caller
sendnothing(2)	NULL	No reason sent
sdmf(3)	number	Number in NA SDMF format
excludeName(4)	nothing	No name parameters or reason

arrisMtaDevSipConfigFileURL

The URL of the SIP configuration file for re-downloading provisioning and configuration parameters to this device.

arrisMtaDevSipDwnldConfig

Set to **on**(2) to enable a re-download of the SIP configuration file parameters specified in the configuration file URL obtained from the **arrisMtaDevSipConfigFileURL** object.

arrisMtaDevSpecialConfigurationOverrideEnable

A bitfield that enables proprietary features of the Arris eDVA. Currently, only 0x80000000 is supported, to enable DHCP Option 60 sub-option 18 override. The default value is **0**.

arrisMtaDevRtcpTosValue

The value used in the IP ToS byte for RTCP packets. Valid range: **0** to **63**.

arrisMtaDevAutomaticOsiDelay

The time, in 100ms increments, to wait after a DLCX is received before determining whether an automatic OSI should be generated to force a line disconnect.

Valid range: **0** to **100**. Default: **50** (5 seconds). Use a value of **0** to send OSI immediately (if there are no other connections on the line).

Note: This object takes effect only if the **ppCfgMtaCallpFeatureSwitch** has bit 0x20000000 set.

arrisMtaDevCustomJitterBufferEnabled

Set to **on**(1) to customize jitter buffer settings. When this setting is off (the default), the jitter buffer size is set using the **arrisMtaDevVPNomJitterBuffer** and **arrisMtaDevVPMaxJitterBuffer** objects.

The default jitter buffer range depends on the setting of this object:

- When **off**(0), the defaults are (in ms):

minimum = [(packet rate *1) + 5]
nominal = minimum
maximum = [(packet rate * 3) + 5]

- When **on**(1), the defaults are:

  minimum: 5
  nominal: 10
  maximum: 60

arrisMtaDevCustomMinJitterBuffer
The customized voice playout minimum jitter buffer size to use when the **arrisMtaDevCustomJitterBufferEnabled** object is enabled. Valid range: **5** (default) to **160**, in increments of 5.

arrisMtaDevCustomNomJitterBuffer
The customized voice playout nominal jitter buffer size to use when the **arrisMtaDevCustomJitterBufferEnabled** object is enabled. Valid range: **5** to **160**, in increments of 5. Default: **10**.

arrisMtaDevCustomMaxJitterBuffer
The customized voice playout maximum jitter buffer size to use when the **arrisMtaDevCustomJitterBufferEnabled** object is enabled. Valid range: **5** to **160**, in increments of 5. Default: **60**.

arrisMtaDevEnableDHCPLog
Enables or disables eDVA DHCP logging.

arrisMtaDevEnableMGCPLog
Enable or disables CallP signaling logging.

arrisMtaDevClearDHCPLog
Set to **clear**(1) to clear the eDVA DHCP Logs.

arrisMtaDevClearMGCPLog
Set to **clear**(1) to clear the eDVA MGCP Logs.

arrisMtaDevTDDReportToCMS
Enables or disables reporting of TDD detection events to the CMS. The default is **enable**(1).

arrisMtaDevAutomaticCallResourceRecovery
The time, in seconds, to delay after an on-hook event before detecting whether resources acquired while a call was active need to be recovered on a line in the idle state. Set to **0** to disable this feature.

arrisMtaDevOffHookFskDelay
The time, in milliseconds, to delay before sending the FSK to the CPE. The delay starts upon receiving the ACK (DTMF D) from the CPE in response to the CAS tone for Call Waiting (or Type 2) Caller ID. Valid range: **0** to **500**.

arrisMtaDevT38Timeout
The T.38 timeout, in seconds. The audio is muted for this period before reporting T.38 failure events. Valid range: **1** to **30**. Default: **15**.

arrisMtaDevSuperG3FaxRelay

Set to **`enable`**(1) to allow SuperG3 fax processing upon detection of the V.21 CM or V.8 data signal. When enabled, the MTA handles the detection of SuperG3 signaling to start the T.38 process. The negotiation during call setup determines whether or not T.38 can be used to send the fax.

When disabled (the default), the MTA still detects signaling for SuperG3 fax, and the signal is used to setup the endpoint for SuperG3 pass-thru fax transmission via G.711.

arrisMtaDevDTMFEndEventForceAscending

Enables or disables RFC 2833 DTMF end event duration force ascending. The default is disabled.

arrisMtaDevDspHandleBellModemTone

Set to **`enable`**(1) to detect the Bell Modem Tone (2225 kHz). The Bell Modem Tone is frequently used by older data modems, usually in low speed setups. When enabled, the DSP detects the Bell Modem Tone from either the local or network end. When disabled (the default), the tone is ignored.

arrisMtaDevDhcpSubOpt3Immediate

Set to **on**(2) to enable immediate comparison and handling of MTA DHCP Option 122 sub-option 3. SNMP notifications are sent to the new Provisioning SNMP Entity. The default value is **`off`**(1).

arrisMtaDevMaxCallPServiceFlows

Used to limit the number of active calls (service flows). Outgoing calls, incoming calls, and conference call legs are included in this count. The eDVA ports can call each other without this limitation. Valid range: **`0`** to **`64`**. The default value is **`0`**.

arrisMtaDevCmIpTable

Provides a way to read the CM IP address from the eDVA. The table has one entry, containing the following objects:

arrisMtaDevCmIpAddressType

The CM IP address type: **`ipv4`**(1) or **`ipv6`**(2).

arrisMtaDevCmIpAddress

The CM IP address.

arrisMtaDevCmIpPhysAddress

The CM MAC address.

arrisMtaDevEndPntTable Objects

This table provides per-line provisioning details.

arrisMtaDevEndPntDialingMethod

The method used to dial the digits for this endpoint:

`tone`(1)

(default) tone dialing (DTMF)

`pulse`(2)
dial-pulse signaling (DTMF is disabled)

`toneAndPulse`(3)
tone dialing (DTMF) and dial-pulse signaling

`pulseWithDTMFRelay`(4)
DTMF is disabled, and pulse dialed digits are relayed in-band to the media gateway

`toneAndPulseWithDTMFRelay`(5)
DTMF is enabled, and pulse dialed digits are relayed in-band to the media gateway.

Note: The values **`pulseWithDTMFRelay`**(4) and **`toneAndPulseWithDTMFRelay`**(5) require an IPDT solution as well as DTMF support by the media gateway.

arrisMtaDevEndPntRingingWaveform
The voltage waveform used when ringing this endpoint: **`normal`**(1) (default) or **`sinusoidal`**(2). Sinusoidal waveform is for use with telephones that exhibit increased sensitivity to ring voltage waveform.

arrisMtaDevEndPntFaxOnlyLineTimeout
Set to a non-zero value to define a line as a fax-only line for NCS. The default value is **`0`**.

In fax-only mode, this object defines a timer (in seconds) that is started after the RTP mode becomes sendReceive. If this timer expires without detecting fax/modem tones on the connection, the call is dropped.

arrisMtaDevPersistentLineStatus
Controls persistent line status. This object and **ifAdminStatus** impact the line status.

- Set to **`ignore`**(0) (the default) to base the line status on the setting of **ifAdminStatus** in the MTA configuration file. If **ifAdminStatus** is not set in the MTA configuration file, then it uses the default value: **up**(1) if the line is provisioned or **down**(2) if the line is not provisioned.
- Set to **`forceDisable`**(1) to force the service status of the line to **down**(2) and the line state to EP_OOS after a reset.

This object is always ignored in a configuration file.

arrisMtaDevEndPntCallWaitingRepeatSteady
Set to **`enabled`**(1) to repeat the call waiting tone forever. The default value **`disabled`**(0) uses the normal repeat rules.

arrisMtaDevEndPntCIDEnable
Set to **`disable`**(0) to disable sending of all Caller ID fields.

arrisMtaDevEndPntCIDNameEnable
Set to **`disable`**(0) to disable sending the CallerID Name field.

arrisMtaDevEndPntCIDDateTimeEnable
Set to **`disable`**(0) to disable sending the CallerID Date/Time field.

arrisMtaDevEndPntLoopReversal
When set to **`enabled`**(1), the line reverses to normal polarity once the originating party

hangs up first. The default value, **disabled**(0), maintains reverse loop polarity while the originating party hangs up.

arrisMtaDevEndPntGainControlTxVoice

The transmit digital gain adjustment, in dBm, for voice calls. When set to a value other than **disabled**(-128), this object supersedes **arrisMtaDevGainControlTxVoice**. Valid range: **–16** to **16**. Default: **0**. A value of **2** increases the voice level by 2 dBm. A value of **–2** decreases the voice level by 2 dBm.

This object does not affect the levels of local tones or FSK.

Note: Use caution when changing this object. Increasing or decreasing the voice level by the larger numbers allowed in the range may compromise voice quality.

arrisMtaDevEndPntGainControlRxVoice

The receive digital gain adjustment, in dBm, for voice calls. When set to a value other than **disabled**(-128), this objects supersedes **arrisMtaDevGainControlRxVoice**. Valid range: **–16** to **16**. Default: **0**. A value of **2** increases the voice level by 2 dBm. A value of **–2** decreases the voice level by 2 dBm.

This object does not affect the levels of local tones or FSK.

Note: Use caution when changing this object. Increasing or decreasing the voice level by the larger numbers allowed in the range may compromise voice quality.

arrisMtaDevGainControl Objects

These objects provide digital gain adjustments for each endpoint.

arrisMtaDevGainControlFSK

The transmit digital gain adjustment, in dBm, for MTA-generated FSK tones (CID and VMWI). Valid range: **–10** to **2**. Default: **0**.

arrisMtaDevGainControlCAS

The transmit digital gain adjustment, in dBm, for MTA-generated CAS tones. Valid range: **–2** to **2**. Default: **0**.

arrisMtaDevGainControlLocalTone

The transmit digital gain adjustment, in dBm, for all MTA-generated Call Progress tones (Dialtone, Busytone, Ringback, etc.) toward the CPE. Valid range: **–2** to **2**. Default: **0**. This object does not effect CAS tone levels; use **arrisMtaDevGainControlCAS** to adjust the CAS tone.

arrisMtaDevGainControlNetworkTone

The transmit digital gain adjustment, in dBm, for MTA-generated Call Progress tones toward the network (Ringback). Valid range: **–2** to **2**. Default: **0**.

arrisMtaDevGainControlLocalDTMF
The transmit digital gain adjustment, in dBm, for MTA-generated DTMF tones toward the local CPE (ex. DTMF CID). Valid range: **–15** to **9**. Default: **0**.

arrisMtaDevGainControlNetworkDTMF
The transmit digital gain adjustment, in dBm, for MTA-generated DTMF tones toward the network (ex. IPDT Pulse Dialing). Valid range: **–9** to **9**. Default: **0**.

arrisMtaDevGainControlTxVoice
The transmit digital gain adjustment, in dBm, for voice. This value does not effect the levels of local tones or FSK. Valid range: **–16** to **16**. Default: **0**.

Note: Use caution when changing this object. Increasing or decreasing the voice level by the larger numbers allowed in the range may compromise voice quality.

arrisMtaDevGainControlRxVoice
The receive digital gain adjustment, in dBm, for voice. Valid range: **–16** to **16**. Default: **0**.

Note: Use caution when changing this object. Increasing or decreasing the voice level by the larger numbers allowed in the range may compromise voice quality.

arrisMtaDevLevelControl Objects

These objects control off-hook tone gain.

arrisMtaDevLevelControlOffHookEnable
Set to **enable**(1) to allow use of the **arrisMtaDevLevelControlOffHookFSK** and **arrisMtaDevLevelControlOffHookCAS** objects, instead of **arrisMtaDevLevelControlFSK** and **arrisMtaDevLevelControlCAS**, in off-hook situations.

arrisMtaDevLevelControlOffHookFSK
The transmit digital gain setting, in dBm, for MTA-generated FSK tones (CID and VMWI) while the line is off-hook. Valid range: **–32** to **–10**. Default: **–15**.

arrisMtaDevLevelControlOffHookCAS
The transmit digital gain setting, in dBm, for MTA-generated CAS tones (CID and VMWI) while the line is off-hook. Valid range: **–32** to **–10**. Default: **–15**.

arrisMtaDevDiagLoopTable Objects

These objects provide per-line Loop Diagnostic (patent pending) details.

arrisMtaDevDiagLoopTime
(read-only) The time and date when loop diagnostics were last run on the selected line.

arrisMtaDevDiagLoopRequest
Set this value to **true**(2) to start loop diagnostics on the line.

arrisMtaDevDiagLoopLastResult
(read-only) Current loop diagnostics status; one of:

- **diagnostics-passed**(1)
- **hazardous-potential-test-failure**(2)
- **foreign-emf-test-failure**(3)
- **resistive-faults-test-failure**(4)
- **receiver-offhook-test-failure**(5)
- **ringer-test-failure**(6)
- **invalid-state-to-init-diags**(7)
- **line-is-unprovisioned**(8)
- **diagnostics-results-pending**(9)
- **not-started**(10)
- **unsupported**(11)
- **ringer-test-warning**(12)

When complete, the status is either **diagnostics-passed**(1) or the first test failed. During the test, the status is always **diagnostics-results-pending**(9).

arrisMtaDevDiagLoopHazardousPotentialTest
(read-only) The Hazardous Potential Test result for the last time that loop diagnostics were run.

arrisMtaDevDiagLoopForeignEmfTest
(read-only) The Foreign EMF Test result for the last time that loop diagnostics were run.

arrisMtaDevDiagLoopResistiveFaultsTest
(read-only) The Resistive Faults Test result for the last time that loop diagnostics were run.

arrisMtaDevDiagLoopReceiverOffHookTest
(read-only) The Receiver Off-hook Test result for the last time that loop diagnostics were run.

arrisMtaDevDiagLoopRingerTest
(read-only) The Ringer Test result for the last time that loop diagnostics were run.

ARRIS-MTA-MIB (battery telemetry items)

Battery-related objects appear in the following groups:

- **arrisMtaDevPwrSupplyBase**
- **arrisMtaDevPwrSupplyControl**
- **arrisMtaDevPwrSupplyTimers**
- **arrisMtaDevPwrSupplyStats**
- **arrisMtaDevPwrSupplyAlarm**

arrisMtaDevPwrSupplyBase Objects

These read-only objects provide basic information.

arrisMtaDevBatteryChargerFWRev

The firmware revision of the battery charger.

arrisMtaDevPwrSupplyControl Objects

These objects provide control and status information for the charger system.

arrisMtaDevPwrSupplyEnableDataShutdown

Set to **`disabled`**(2) to allow CPE interfaces (Ethernet and wifi, if equipped) to provide service during a power outage. This reduces battery hold-up time.

arrisMtaDevPwrSupplyLowBatteryThresh

Sets the low battery threshold, in 10 watt*minutes increments. A charge below this threshold indicates a low battery condition. The initial default value is equivalent to 1 hour of holdup time.

arrisMtaDevPwrSupplyTypicalIdlePwr

(read-only) The typical idle power, in 50 mW increments. A nominal value is loaded when the modem powers up. This value is used in conjunction with the Tested Battery Capacity, Low Battery Threshold, and the Replace Battery Threshold to determine when to raise the Replace Battery or Low Battery alarms.

arrisMtaDevPwrSupplyReplaceBatThresh

The minimum acceptable battery charge, in 10 watt*minutes increments, needed to achieve the desired End of Life hold-up time, based on typical idle power. If the Tested Battery Capacity minus the Charge Hysteresis loss (20%) is less than this value, the modem raises the Replace Battery alarm. The initial default value provides 1 hour of holdup time.

arrisMtaDevPwrSupplyChargeState

The ‘‘full charge’’ state, in 10 watt*minutes increments. The initial default value is 80% of the Rated Battery Capacity. Writing to this object forces the charger to start a discharge/charge cycle, to charge the battery to the specified level.

Be careful when changing this value. Specifying a higher charge level may increase hold-up times, but can reduce overall battery life.

arrisMtaDevPwrSupplyBatteryTest

Controls the battery test schedule:

`testScheduled`(0)
: Resumes the battery test scheduler at its current value. When read, this value indicates that the battery test runs when the scheduled time expires.

`disableAutoTesting`(1)
: Freezes the battery test scheduler at its current value. When read, this value indicates that the battery test is suspended. Removing and replacing the battery forces the charger to run the test one time. To resume normal operation, set the value to **`testScheduled`**(0) or **`testInProgress`**(2).

testInProgress(2)
Starts the battery test cycle immediately, and resets the test scheduler to its default value of 180 days. When read, this value indicates a test in progress.

testPending(3)
(read-only) A battery test was in progress, and either AC power was lost or a full charge was requested.

arrisMtaDevPwrSupplyConfigRunTime

The estimated battery hold-up time, in minutes, based on the typical idle power and the programmed battery charge setting. The battery hold-up time may be adjusted using this object. By setting the holdup time to a lower value, the total service life of the battery is extended. Increasing the hold-up time decreases the total service life of the battery.

Note: This value can only be set in multiples of 5 minutes. Setting a value greater than **arrisMtaDevPwrSupplyBatAvailableMinutes** does not extend the hold-up time beyond that specified by **arrisMtaDevPwrSupplyBatAvailableMinutes**.

arrisMtaDevPwrSupplyConfigReplaceBatTime

The replace battery threshold, in terms of minutes of hold-up time. If a battery's capacity has degraded to a point where its hold-up time is below this threshold, the Replace Battery condition becomes active.

This value can only be set to multiples of 5 minutes. The default value at power up is 60 minutes.

arrisMtaDevPwrSupplyOverTempAlarmControl

Set to **enable**(1) to issue an Over Temperature Alarm if the charger exceeds the temperature specified in **arrisMtaDevPwrSupplyOverTempAlarmThreshold**, and to shut down the charger if the temperature exceeds 90°C.

arrisMtaDevPwrSupplyOverTempAlarmThreshold

The temperature threshold, in degrees C, for the Over Temperature alarm. Valid range: **50** to **70**.

arrisMtaDevPwrSupplyTemperature

(read-only) The current charger temperature, in degrees C. This is available only when arrisMtaDevPwrSupplyOverTempAlarmControl is enabled.

arrisMtaDevPwrSupplyHiTempBatteryShutdownControl

Set to **enable**(1) to turn off the battery if the temperature reaches 75°C.

arrisMtaDevPwrSupplyHighestTemperature

(read-only) The highest recorded battery charger temperature, in degrees C. This is available only when **arrisMtaDevPwrSupplyOverTempAlarmControl** is enabled.

arrisMtaDevPwrSupplyHighestTemperatureTime

(read-only) The time and date when the highest temperature was recorded.

arrisMtaDevPwrSupplyHighestTemperatureClear

Set to **clear**(1) to clear current values of **arrisMtaDevPwrSupplyHighestTemperature** and **arrisMtaDevPwrSupplyHighestTemperatureTime**. These objects reset to current values within 4 seconds.

arrisMtaDevPwrSupplyControlChargerReset
Set to `true`(1) to reset the battery charger.

Note: Resetting the battery charger during an AC Fail condition immediately shuts down the unit.

arrisMtaDevPwrSupplyTimers Objects

These objects provide timers for the battery charging system.

arrisMtaDevPwrSupplyDataShutdownTime
The timeout period, in seconds, until the device terminates data services (Ethernet, USB, and wifi if equipped) after loss of AC power. The default value depends on the gateway type:

- 802.11ac-capable Gateway products: 30 seconds.
- all others: **`900`** seconds (15 minutes).

arrisMtaDevPwrSupplyFullChargeTime
The number of days to maintain the battery at 100% of its rated voltage. Any ongoing or schedule battery tests are stopped. This may be useful when widespread power outages are expected (for example, an approaching storm system). After the time has elapsed, the charger allows the battery to return to the value specified by **arrisMtaDevPwrSupplyChargeState**.

Valid range: **`1`** to **`16`**. A value of **`0`** may also be read, which indicates that the charger is operating normally.

arrisMtaDevPwrSupplyStats Objects

These object provide battery and charger statistics. All objects in this group are read-only.

arrisMtaDevPwrSupplyBatteryTestTime
The present value of the test timer scheduler, in days. If the value is 0xFF (255), the timer has been paused.

arrisMtaDevPwrSupplyRatedBatCapacity
The rated capacity, in 10 watt*minutes increments, of the battery.

arrisMtaDevPwrSupplyTestedBatCapacity
The measured battery capacity, in 10 watt*minutes increments, as measured by the last battery test cycle.

arrisMtaDevPwrSupplyBatStateOfCharge
The present battery state of charge, in 10 watt*minutes increments. This value is approximate and is re-calibrated following a battery test cycle.

arrisMtaDevPwrSupplyReadBatteryPwr

The present load power, in 50mW increments, over an eight-second moving average. This is the power being removed (when running on battery power) or applied to the battery (when charging).

arrisMtaDevPwrSupplySecondsOnBattery

The time, in seconds, that the modem has been using battery power.

arrisMtaDevPwrSupplyBatRatedMinutes

The estimated rated hold-up time, in minutes, based on typical idle power and the rated capacity of the battery when fully charged.

arrisMtaDevPwrSupplyBatAvailableMinutes

The estimated available hold-up time, in minutes, based on typical idle power and the tested capacity of the battery when fully charged.

arrisMtaDevPwrSupplyTelemetryValues

Power supply telemetry values, used by ARRIS technical support when troubleshooting a battery issue.

arrisMtaDevBatteryStatusTable

The Power Supply telemetry table. All objects in this table are read-only.

arrisMtaDevBatteryOperState

The current operational state of the battery:

- `unavailable`(0)
- `invalid`(1)
- `shutdownWarning`(2)
- `batteryReversedShorted`(3)
- `batteryLow-replaceBattery-acFail`(4)
- `batteryLow-replaceBattery`(5)
- `batteryLow-acFail`(6)
- `batteryLow`(7)
- `batteryMissing`(8)
- `acFail-replaceBattery`(9)
- `replaceBattery`(10)
- `acFail`(11)
- `normal`(12)
- `testInProgress`(13)
- `chargerFailure`(14)

arrisMtaDevBatteryLastStateChange

The value of **sysUpTime** when the battery entered its current operational state.

arrisMtaDevBatteryOperSubState

The current sub-state of the battery. The sub-state is not designed to match the

arrisMtaDevBatteryOperState but to provide additional information about the charger status.

arrisMtaDevBatteryOrderingCode

The ARRIS ordering code for the battery.

arrisMtaDevBatteryEprom

EPROM information for the battery.

ARRIS-MTA-DOC30-DEVICE-MIB

This MIB supplies basic proprietary management objects for eDVA components on Touchstone DOCSIS 3.x devices.

arrisMtaDoc30EmergencyNumber

Specifies the emergency number. A call to the emergency number sets the QoS priority to the highest level.

arrisMtaDoc30RootCertType

The iptel root certificate type used for MTA security: **`testRoot`**(1) or **`realRoot`**(2).

The object value can be changed only in the CM configuration file.

arrisMtaDoc30AdjustCallpFeatureSwitch

When set to **`enable`**(1), the devices maps the value of **ppCfgMtaCallpFeatureSwitch** to a value that uses the same settings as TS6.x firmware releases. The default value is **`disable`**(0).

arrisMtaDoc30InvalidateTickets

When set, the eDVA invalidates (and flushes from NVRAM) all Provisioning and CMS server Kerberos Tickets when eDVA provisioning has stopped and the provisioning timer has expired. The setting controls the timing:

- **`enable`**(1): the eDVA immediately invalidates the Kerberos Application Ticket for the Provisioning Application and all CMS Application Tickets.
- **`disable`**(0) (default): the eDVA uses the Provisioning and CMS Application Tickets currently stored in NVRAM before invalidating them.

ARRIS Speedtest MIB

The Speedtest MIB objects are collected in the **arrisSpeedTestMib**. The following objects are defined, separated into configuration, control, and results.

Configuration Objects

The uplink and downlink configurations are set using URL formatted strings.

arrisSpeedTestConfigEndUserGui

Set to **`true`**(1) to include the Speedtest web page in the Basic (subscriber-accessible) pages.

arrisSpeedTestConfigSyslogReports

Set to **`true`**(1) to report results to Syslog and the event log.

arrisSpeedTestConfigCpeAccess

Set to **`true`**(1) to allow CPE access to the Internet while running a speed test.

arrisSpeedTestConfigDownlinkURL

Defines the downlink configuration, in URL format (see below for an example).

arrisSpeedTestConfigUplinkURL

Defines the uplink configuration, in URL format (see below for an example).

Download Examples

The following examples show typical download URLs for FTP and HTTP tests. In these examples, the user ID and password are both **`speedtest`**.

```
ftp://speedtest:speedtest@10.1.59.18/movie.mpg
?ftpmode=passive&tcpwindow=2048&tcpmss=1460
&tcptimestamps=1&pmtud=1&cpeaccess=1
http://speedtest:speedtest@10.1.59.18:80/movie.mpg
?tcpwindow=63&tcpmss=1460&tcptimestamps=1&pmtud=1 &cpeaccess=1
```

Upload Examples

The following examples show typical upload URLs for FTP and HTTP tests. In these examples, the user ID and password are both **`speedtest`**.

```
ftp://speedtest:speedtest@10.89.255.203:21/uptest.txt
?ftpmode=active&uploadsize=10&tcpwindow=63&tcpmss=1460
&tcptimestamps=1&pmtud=1&cpeaccess=1
http://speedtest:speedtest@10.89.255.203:80/uptest.txt
?uploadsize=10&script=cgi-bin/upload.cgi&postfields
=ac~upload;login~admin;pass~admin&filefield=file
&tcpwindow=63&tcpmss=1460&tcptimestamps=1&pmtud=1 &cpeaccess=1
```

Control Object

Use the following object to start and stop the speed tests:

arrisSpeedTestConfigStartStopTest

One of the following:

- **`stopTest`**(0)
- **`startDownlinkTest`**(1)
- **`startUplinkTest`**(2)

Results Table

Use the following object to retrieve speed test results:

arrisSpeedTestResultsStatus

Use an index of **1** to **3** to retrieve a test result. **arrisSpeedTestResultsStatus**.1 contains the latest result.

The string is formatted as shown in the following example:

```
Status=complete&starttime=Thu Aug 11 13:19:04 2011 &endtime=Thu Aug 11
13:19:04 2011&vlstart=0&vlend=0 &cpedata=1 Download=complete&speed=
99.40Mbps &bytes=105541632&ohspeed=103.33&ohbytes=109734528
&totalspeed=103.33&totalbytes=109735469&pkts=72302
&pkterrors=0&pktdropped=0&setuptime=60 UploadStatus=complete&speed=
36.52Mbps&bytes=10486058 &ohspeed= 37.44&ohbytes=10874607&totalspeed= 37.44
&totalbytes=10874607&pkts=7190&pkterrors=0 &pktdropped=0&setuptime=20
```

If the **arrisSpeedTestConfigSyslogReports** object is set to `true`(1), the device also reports the results to Syslog. The following message (Event ID 2417164311, Informational) is an example:

```
SpeedTest Results: Download Status: Complete, Speed: 86.91, Upload Status:
Complete, Speed: 37.27;CM-MAC=00:15:d1:ad:96:9d; CMTS-
MAC=00:01:5c:23:62:82;CM-QOS=1.1;CM-VER=3.0;
```

Chapter 7

Administration

Administration involves collecting performance statistics, capacity planning, and maintaining system reliability.

Administration Objects

Touchstone supports both standard and ARRIS-proprietary SNMP MIBs for administration and other purposes. This section describes several generic objects. See the Operations chapter for ARRIS-proprietary objects.

System Description Objects

The **system** objects can be used to identify a Touchstone product and find general information about it. Most of these objects are generic to all devices, but those described below have values unique to ARRIS products.

sysDescr Object

Touchstone firmware supports the **sysDescr** MIB object. This object provides firmware version and product description information in the format specified in section 4.2.1 of the DOCSIS *Operations Support System Interface Specification*, CM-SP-OSSIv3.0-I15-100115. The specification requires the **sysDescr** object to be in the following format:

```
any text <<HW_REV: hardware version information;
VENDOR: vendor name; BOOTR: BootROM version;
SW_REV: firmware version; MODEL: hardware model information>> any text
```

Since the content is in a consistent format, the object can be automatically parsed and used for various functions such as determining when firmware upgrades are required. The fields in the **sysDescr**.0 object are:

Field	Meaning	Description
HW_REV	Hardware Revision	The hardware revision of the Telephony Modem. ARRIS updates this field as needed to reflect significant hardware changes or improvements to the product.
VENDOR	Vendor Name	The vendor name; in this case, “Arris Interactive, L.L.C.”
BOOTR	Boot ROM	The BootROM image version that is embedded in the product, and used to load the application firmware image.

Field	Meaning	Description
SW_REV	Firmware Revision	The firmware version of the application firmware image currently loaded on the device. **Note**: In addition to the System Descriptor MIB object (**sysDescr**.0), the SW_REV information is also available in the **docsDevSwCurrentVers** MIB object.
MODEL	Model Number	The hardware model number of the device.

The following is an example of the **sysDescr**.0 contents for a modem device.

```
ARRIS DOCSIS 3.1 / PacketCable 1.5 Touchstone Residential Gateway <<HW_REV:
3; VENDOR: ARRIS Group, Inc.; BOOTR: 1.2.7.491938; SW_REV: 11.1.X; MODEL:
TG3402A>>
```

sysORTable Objects

The following **sysORTable** objects provide useful information:

sysORID
Equivalent to the OID representing **modemAgentDocsis20**.

sysORDescr
Contains the string “DOCSIS 3.0 Cable Modem agent.”

sysObjectID Object

The **sysObjectID** object provides a condensed version of the data found in the **sysDescr** object. The data in this object is in OID format, consisting of the following fields:

Field(s)	Description
1–9	Equivalent to the OID representing **arris**: **`1.3.6.1.4.1.4115`**. Some SNMP software may display this as **arris** or **enterprises.4115**.
10	The model number; for example, **`802`**
11	Hardware release
12–16	Not used, display as 0

Bridging and Routing Objects

Depending on provisioning and model-specific capabilities, Touchstone hardware can function either as a bridge or a router. The MIBs described here provide information about each function.

dot1dBridge Objects

The **dot1dBridge** MIB provides information about the bridging function. The following objects can be of use:

dot1dBaseBridgeAddress.0
In Touchstone hardware, the MAC address of the Ethernet interface.

dot1dBaseNumPorts.0
The number of interfaces connected to the bridge. Touchstone hardware uses the **ifIndex** of the interface as indexes to various tables in the **dot1dBridge** MIB.

dot1dBaseType.0
The bridge type.

dot1dTpFdbTable
The forwarding database. This table uses the MAC address of supported interfaces as the index.

rip2 Objects

The **rip2** MIB provides routing information when the E-UE is provisioned for RIPv2 routing support. The following objects can be of use:

rip2IfStatTable
Statistics for each interface enabled for RIP routing.

rip2IfConfTable
RIP interface configuration for each interface enabled for RIP routing.

End of Call Connection Statistics

Touchstone firmware supports the PacketCable 1.0-defined call connection statistics, with clarifications as defined in ECN EC-MGCP-N-04.0175-7. The EC clarifies the requirement to ensure that the statistics represent the actual packets sent/received regardless of the current connection state of the call.

Note: End-of-call connection statistics and Voice Quality Monitoring (VQM) statistics are different features. For information about the VQM feature, see “see "*Managing Voice Quality Monitoring* (page 250).”

Touchstone firmware makes end-of-call connection statistics available through proprietary MIB objects and Syslog messages. The **arrisMtaDevEnableCallStatsSyslogRpt** MIB object controls end-of-call statistics reporting through Syslog. The **arrisMtaDevEnableCallSigLastMsgRpt** MIB object controls reporting of the last 4K of signaling messages through Syslog. Each object may be set independently, and interact as follows:

CallStats MIB	LastMsg MIB	CMS LoC Alarm	Messages Sent
Disabled	Disabled	Inactive	None
		Active	None
	Enabled	Inactive	None
		Active	None
Enabled	Disabled	Inactive	None
		Active	Last messages sent/received
	Enabled	Inactive	Last 4K of messages sent/received
		Active	Last 4K of messages sent/received

NCS Behavior

Touchstone NCS loads make end-of-call connection statistics available through proprietary MIB objects and Syslog messages. Touchstone firmware supports the PacketCable-defined call connection statistics, with clarifications as defined in ECN EC-MGCP-N-04.0175-7. Previously, the PacketCable NCS specification implied that these statistics were related to the connection mode requested by the CMS. The EC clarifies the requirement to ensure that the statistics represent the actual packets sent/received regardless of the current connection state of the call.

The E-UE sends connection statistics to the call server (and optionally, Syslog servers) during the call tear-down procedure. PacketCable-compliant call servers provide a method of reporting these captured statistics. See the call server documentation for instructions on accessing the statistics.

SIP Behavior

Touchstone SIP loads support end-of-call statistics reporting through the SIP PUBLISH mechanism (defined in RFC 3903). The content of the PUBLISH message uses the session report format.

To enable end-of-call statistics reporting, set the following MIB objects:

pktcEDVACodecPubRepAddrType
: Specifies the IP address type (IPv4 or IPv6) of the device that receives statistics reports.

pktcEDVACodecPubRepAddr
: The IP address of the device that receives statistics reports. If this object is not specified, end-of-call statistics are disabled.

pktcEDVACodecRTCPXR
: Determines whether the eDVA includes far-end statistics in the report. The default is `true`(1).

End-of-Call Statistics MIB Objects

Touchstone firmware provides MIB objects for monitoring eDVA end-of-call statistics.

The monitoring MIB is broken into groups under **arrisMtaDevTrace**:

- Objects that report on a per-call level (the counter values represent the total count for the most recently completed call on an endpoint). These objects are indexed by the endpoint number in the table **arrisMtaDevCallStatsEntry**.
- Objects that report on a device level (the counter values represent the sum total for all endpoints), located under **arrisMtaDevTrace**.
- Objects that control and reset counters, located under the **arrisMtaDevControl** MIB tree.

Device Level End-of-Call Statistics

Device-level MIB objects represent combined end-of-call statistics for all lines on the eDVA.

arrisMtaDevRtpTxPktsTotal

(read-only) The total number of RTP packets sent from the eDVA since it was last started up or reset. This value represents the total number of packets sent for all endpoints combined. It is calculated from values obtained from the signaling end of call statistics. Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

arrisMtaDevRtpRxPktsTotal

(read-only) The total number of RTP packets received by the eDVA since it was last started up or reset. This value represents the total number of packets received for all endpoints combined. It is calculated from values obtained from the signaling end of call statistics. Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

arrisMtaDevRtpPercentPktsLostTotal

(read-only) The percentage of RTP packets lost since the eDVA was last started up or reset. This value represents the total number of packets lost for all endpoints combined. It is calculated from values obtained from the signaling end of call statistics. Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

The value of this object is expressed in increments of 1/100 percent. For example, a value of `1745` means that 17.45% of the packets were lost.

arrisMtaDevRtpPktsLostTotal

(read-only) The number of RTP packets lost since the MTA was last started up or reset. This value represents the total number of packets lost for all endpoints combined. It is calculated from values obtained from the signaling end of call statistics. Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

arrisMtaDevLastCallStartTime

(read-only) The last call start time from the eDVA.

arrisMtaDevLastCallEndTime

(read-only) The last call end time from the eDVA.

arrisMtaDevSignalingAvgLatency

(read-only) The average latency or delay, in milliseconds, for responses to signaling messages. It is calculated from values obtained from the signaling end of call statistics. Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

arrisMtaDevSignalingTxSuccessfulMsgCnt

(read-only) The total number of successful signaling messages sent from the eDVA. Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

arrisMtaDevSignalingRxSuccessfulMsgCnt

(read-only) The total number of successful signaling messages received by the MTA. Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

arrisMtaDevSignalingTxNAKCnt

(read-only) The total number of negative acknowledgement signaling messages (NAKmessages) sent from the MTA. Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

arrisMtaDevSignalingRxNAKCnt

(read-only) The total number of negative acknowledgement signaling messages (NAKmessages) received by the MTA.Set the **arrisMtaDevResetCallStats** object to `true`(1) to clear this counter.

Line Level Statistics

Line-level MIB objects, collected in the **arrisMtaDevCallStatsTable**, report end-of-call statistics and general status for the most recently-completed call on each line in the eDVA. Entries in this table are indexed by the endpoint number (1 to N), where N is the maximum number of lines supported by the eDVA; for example, **arrisMtaDevCallStatsAvgJitter**.2 shows the average jitter for the second line.

Other MIB objects in this table provide information about endpoint temperature, hook status, and operational status.

arrisMtaDevCallStatsRtpTxPkts

The total number of RTP packets sent from the endpoint during the most recent call. This value is obtained from the signaling end-of-call statistics.

arrisMtaDevCallStatsRtpRxPkts

The total number of RTP packets received by the endpoint during the most recent call. This value is obtained from the signaling end-of-call statistics.

arrisMtaDevCallStatsRtpPercentPktsLost

The percentage of RTP packets lost during the most recent call. This value is obtained from the signaling end-of-call statistics.

The value of this object is expressed in increments of 1/100 percent. For example, a value of `1745` means that 17.45% of the packets were lost.

arrisMtaDevCallStatsAvgJitter

The average jitter measurement, in milliseconds, during the most recent call. This value is obtained from the signaling end-of-call statistics.

arrisMtaDevCallStatsMaxJitter

The maximum jitter measurement, in milliseconds, during the most recent call. This value is obtained from DSP statistics.

arrisMtaDevCallStatsAvgLatency

The average latency, in milliseconds, observed during the most recent call. This value is obtained from the RTCP signaling end-of-call statistics.

arrisMtaDevCallStatsHookStatus

The hook status for each endpoint: **`onHook`**(0) or **`offHook`**(1).

Note: The disconnected state is not a valid return value for this MIB object.

arrisMtaDevCallStatsSLICStatus

The over temperature condition of the SLIC chips: **`normal`**(0) or over **`temp`**(1).

arrisMtaDevCallStatsEndPntOpStatus

The current operational status for each endpoint: **`up`**(1) (ready to pass packets), **`down`**(2), or **`testing`**(3) (in some test mode). This object obtains its value from **ifAdminOperStatus**.

arrisMtaDevCallStatsLineSubState

The current sub-state for each line: **`normal`**(0), **`diagsPending`**(1), **`diagsFailed`**(2), **`lcProtection`**(3), or **`dspFail`**(4). This object obtains its value from the lineRec.

Clearing Counters

To clear the device-level end-of-call statistics counters, set the **arrisMtaDevResetCallStats** MIB to **`true`**(1). Setting this object to a value of **`false`**(2) has no effect. Reading this object always returns **`false`**(2).

This object clears the following counters:

- **arrisMtaDevRtpTxPktsTotal**
- **arrisMtaDevRtpRxPktsTotal**
- **arrisMtaDevRtpPercentPktsLostTotal**
- **arrisMtaDevSignalingAvgLatency**
- **arrisMtaDevSignalingTxSuccessfulMsgCnt**
- **arrisMtaDevSignalingRxSuccessfulMsgCnt**
- **arrisMtaDevSignalingTxNAKCnt**
- **arrisMtaDevSignalingRxNAKCnt**
- **arrisMtaDevSignalingRxNoACKCnt**

Last Signaling Message Sent

To make a short trace of the last messages available at the end of the most recent call, Touchstone firmware allocates a 4K byte circular buffer to store the last signaling messages sent on the most recent call. To enable collection of signaling messages, set the **arrisMtaDevEnableCallSigLastMsgRpt** object to **enable**(1). The default value is **disable**(0).

The following table shows the behavior of the buffer under certain conditions.

arrisMtaDevEnable CallSigLastMsgRpt	CMS LoC Alarm	Buffer Contents Stored
Disable	Inactive	None
	Active	Last messages sent/received
Enable	Inactive	Last 4k of messages
	Active	Last 4k of messages

Since the maximum PDU size for a MIB object response is 256 bytes, the message is split into 256-byte segments and stored in MIB objects **arrisMtaDevSignalingLastMsg1** through **arrisMtaDevSignalingLastMsg16**. Empty objects in this group report "Buffer is empty." When reading these objects, the eDVA sets the **arrisMtaDevEnableCallSigLastMsgRpt** object to **disable**(0) to prevent another message from overwriting these MIB objects while reading the contents. After reading the objects, you must manually set the **arrisMtaDevEnableCallSigLastMsgRpt** object to **enable**(1) to capture the next message.

Per-Call Syslog Reporting

Touchstone firmware can generate a Syslog report of call status and monitoring information on a per-call basis.

In addition to statistical data, the Syslog report can also include up to the last 4K bytes of signaling messages associated with the last call in a circular buffer. The Syslog report can be configured to supply only statistical data, or statistical data and signaling messages.

Enable or disable Syslog reporting using the **arrisMtaDevEnableCallStatsSyslogRpt** object, which is part of the **arrisMtaDevBase** MIB. The default value for this object is **disable**(0); set to **enable**(1) to enable reporting.

Enable or disable the signaling message buffer using the **arrisMtaDevEnableCallSigLastMsgRpt** object. The default value for this object is **disable**(0); set to **enable**(1) to enable the buffer. The contents of the buffer depends on the setting of this MIB, the **arrisMtaDevEnableCallStatsSyslogRpt** object, and the state of the CMS LoC alarm.

The following table shows how the MIB settings interact with the Loss of Comms alarm to affect the buffer contents.

arrisMtaDevEnable CallStatsSyslogRpt	arrisMtaDevEnable CallSigLastMsgRpt	CMS LoC Alarm	Buffer contents contained in Syslog
Disable	Disable	Inactive	None
		Active	None

arrisMtaDevEnable CallStatsSyslogRpt	arrisMtaDevEnable CallSigLastMsgRpt	CMS LoC Alarm	Buffer contents contained in Syslog
	Enable	Inactive	None
		Active	None
Enable	Disable	Inactive	None
		Active	Last messages sent/received
	Enable	Inactive	Last 4k of messages
		Active	Last 4k of messages

Per Call Syslog Report Example

Below is an example of a per-call Syslog report (statistical output only):

```
09-06-2007        10:11:02        Local0.Warning        10.1.36.205
Sep 06 09:39:27 2007 AALN/1:mta205.dev36/10.1.36.205 <6> <4115> <37>
<00:00:CA:CB:23:E7> <Call Stats: L1, HW: TM402G, SW: 5.2.21, RTP Tx 0,
RTP Rx 0, RTP Lost 0.00%, Prov State: MTA Prov Complete. (20),
Avg Jtr 0, Max Jtr 0, Avg Ltc RTP 0, Avg Ltc Sig Msg 11, No ACKs 0,
Batt 100%:INACTIVE, CMS LOC:INACTIVE>
```

The fields in the Syslog report are as follows:

- HW: Hardware Model Number
- SW: Firmware Rev Version Number
- Ttl Rx RTP: Total number of RTP packets received. This only reports RTP packets and not data packets.
- Ttl Tx RTP: Total number of RTP packets sent. This only reports RTP packets and not data packets.
- RTP Lost: Percentage of RTP packets lost. This only reports RTP packets and not data packets.
- Prov State: The step of the eDVA Provisioning Flow that the eDVA is currently in.
- Avg Jtr: Average Jitter in milliseconds.
- Max Jtr: Maximum Jitter in milliseconds.
- Avg Ltc RTP: Average latency for RTP packets, in milliseconds.
- Avg Ltc Sig Msg: average latency for a response to signaling messages, in milliseconds. This value is calculated from end-of-call signaling statistics and is a running average during eDVA uptime.
- No ACKs: Count of the number of “negative acknowledgment” messages received from the Call Agent.
- SNMP Traps: Battery Percentage; CMS LOC.

Message Trace Example

The following is an example Syslog report showing Send (Xmit) and Receive (Rcv) Messages:

```
Nov 17 18:34:11 2005 mta161.dev35 <13>
<4115> <42> <00:00:CA:CB:22:FB> <Rcv: (3: 1 of 1) -
<010>Q: loop<010>R: hf(I), hu(N)’>
```

```
Nov 17 18:34:11 2005 mta161.dev35 <14> <4115> <41> <00:00:CA:CB:22:FB>
<Xmit: (3: 1 of 1) - ’200 31855 OK’>
```

See see "*Capturing Signaling Traces* (page 184) for a breakdown of send and receive message formats.

Managing Voice Quality Monitoring

You can retrieve stored Voice Quality Monitoring (VQM) data using either SNMP or the web-based interface.

Touchstone firmware is compatible with draft PacketCable requirements for VQM data, as described in *RTCP XR VoIP Metrics Package for the Media Gateway Control Protocol*, IETF draft-auerbach-mgcp-rtcpxr-06 for NCS.

About Voice Quality Monitoring

The Voice Quality Monitoring (VQM) feature provides a per-line history of voice call quality metrics. VQM data is available from SNMP, CLI, and HTTP. The information provided is an extension of RFC 3611 recommendations.

Metrics are provided for both the near-end (eDVA) and far-end sides of the call (the far-end eDVA must also support RTCP-XR). Remote VQM is disabled by default.

Note: Touchstone firmware does not support reporting VQM data to the CMS. The eDVA ignores any VQM requests from the CMS.

A monitoring feature provides a Syslog notification if MOS scores fall below a threshold specified by the **arrisMtaDevVqmThresholds** objects.

VQM Data

The following list briefly describes the supported VQM metrics, noting metrics that are extensions of RFC 3611. See RFC 3611 for full descriptions of those metrics.

Call Start Time (extension)
(ARRS02) A timestamp defining the time the call began.

Call End Time (extension)
(ARRS03) A timestamp defining the time the call ended.

Call Duration (extension)
(ARRS04) The total length of the call time, in hours, minutes, and seconds.

Line Number (extension)
(ARRS05) The phone line used for this call.

Remote IP Address (extension)
(IPAD) The IP address of the destination endpoint.

CW Errors(extension)
(ARRS06) The number of codeword errors that occurred during the call.

CW Error Rate (extension)

(ARRS07) The ratio of errored codewords to total codewords.

SNR (extension)

(ARRS08) The signal-to-noise ratio, in dB.

MicroReflections (extension)

(ARRS09) Microreflection power, in dBc. A microreflection is RF energy that is reflected back toward the point of origin due to an impedance mismatch. It is not a measure of audio echo.

Downstream Power (extension)

(ARRS10) The downstream power, in dBmV, measured at the eDVA.

Upstream Power (extension)

(ARRS11) The upstream power, in dBmV, measured at the eDVA.

EQI Average (extension)

(ARRS12) The average Echo Quality Index (EQI). The EQI is an estimate of the quality of echo signals presented to the LEC. The value displayed is between 0 and 1, with higher values indicating higher echo quality. Low EQI measurements represent a high risk for echo-related problems.

EQI Minimum (extension)

(ARRS13) The minimum measured Echo Quality Index.

EQI Maximum (extension)

(ARRS14) The maximum measured Echo Quality Index.

EQI Instantaneous (extension)

(ARRS15) The last measured Echo Quality Index value.

MOS-LQ

(MLQ) The estimated Mean Opinion Score for Listening Quality (MOS-LQ). Valid range: **`10`** to **`50`**, corresponding to MOS × 10. For example, a value of **`35`** corresponds to an estimated MOS score of 3.5. MOS-LQ does not include the effects of delay and can be compared to MOS scores obtained from listening quality (ACR) tests.

A value of **`127`** indicates that this parameter is unavailable.

MOS-CQ

(MCQ) The estimated Mean Opinion Score for Conversational Quality (MOS-CQ). Valid range: **`10`** to **`50`**, corresponding to MOS × 10, as for MOS-LQ.

A value of **`127`** indicates that this parameter is unavailable.

Echo Return Loss

(RERL) The residual echo return loss (RERL), in decibels. RERL may be measured directly by the VoIP-end system's echo canceller or may be estimated by adding the echo return loss (ERL) and echo return loss enhancement (ERLE) values reported by the echo canceller.

RERL(dB) = ERL (dB) + ERLE (dB)

In the case of a VoIP gateway, the source of echo is typically line echo that occurs at 2–4 wire conversion points in the network. This can be in the 8–12 dB range. A line echo canceler can provide an ERLE of 30 dB or more and hence reduce this to 40–50 dB. A typical handset would result in 40–50 dB of echo loss due to acoustic feedback.

Signal Level

(SL) The voice signal relative level — the ratio of the signal level to a 0dBm0 reference, expressed in decibels. Signal level is measured only for packets containing speech energy. This measurement provides a real time indication that the signal level may be excessively high or low.

signal level = 10 Log_{10} (rms talkspurt power(mW))

A value of **`127`** indicates that this parameter is unavailable. Typical values are in the −15 to −20 dBm range.

Noise Level

(NL) The ratio of the silent period background noise level to a 0 dBm0 reference, expressed in decibels.

noise level = 10 Log_{10} (rms silence power (mW))

A value of **`127`** indicates that this parameter is unavailable.

Loss Rate

(NLR) The fraction of RTP data packets from the source, lost since the beginning of reception. Valid range: **`0`** (no packet loss) to **`255`** (near-complete packet loss). Duplicated packets and discarded packets are not counted as lost. However, significantly late packets may be counted as lost.

Pkt Loss Concealment

(PLC) The Packet Loss Concealment (PLC) method; describes how the eDVA is concealing lost packets. The value is one of the following:

Standard

A simple replay or interpolation algorithm is being used to fill-in the missing packet; this approach typically conceals isolated lost packets at low packet loss rates.

Enhanced

An enhanced interpolation algorithm is being used; algorithms of this type can conceal high packet loss rates effectively.

Disabled

Silence is being inserted in place of lost packets.

Unspecified

No information is available concerning the use of PLC; however, for some CODECs this may be inferred.

Discard Rate

(JDR) The fraction of RTP data packets from the source that have been discarded since the beginning of reception, due to late or early arrival, under-run, or overflow at the receiving jitter buffer. Valid range: **`0`** (no packets discarded) to **`255`** (nearly all packets discarded).

Burst Density

(BLD) The fraction of RTP data packets within burst periods since the beginning of reception that were either lost or discarded. Valid range: **`0`** (no packets lost/discarded, or no packets received) to **`255`** (nearly all packets lost/discarded).

Gap Density

(GLD) The fraction of RTP data packets, within inter-burst gaps since the beginning of reception, that were either lost or discarded. Valid range: **`0`** (no packets lost/discarded, or no packets received) to **`255`** (nearly all packets lost/discarded).

Burst Duration

(BD) The mean duration, in milliseconds, of the burst periods that have occurred since the beginning of reception. The duration of each period is calculated based upon the packets that mark the beginning and end of that period. If there have been no burst periods, the burst duration value is zero.

Gap Duration

(GD) The mean duration, in milliseconds, of the gap periods that have occurred since the beginning of reception. The duration of each period is calculated based upon the packet that marks the end of the prior burst and the packet that marks the beginning of the subsequent burst.

In the case of a gap that occurs at the beginning of reception, the sum of the timestamp of the prior burst packet and the duration of the prior burst packet are replaced by the reception start time. In the case of a gap that occurs at the end of reception, the timestamp of the subsequent burst packet is replaced by the reception end time. If there have been no gap periods, the gap duration value is zero.

Round Trip Delay

(RTD) The most recently calculated round trip time between RTP interfaces, in milliseconds.

End System Delay

(ESD) The most recently estimated end system delay, in milliseconds. End system delay is the sum of the total sample accumulation and encoding delay associated with the sending direction and the jitter buffer, decoding, and playout buffer delay associated with the receiving direction.

Gmin

(GMN) The gap threshold, the value used for this report to determine if a gap exists. The recommended value of **`16`** corresponds to a burst period having a minimum density of 6.25% of lost or discarded packets, which may cause noticeable degradation in call quality. During gap periods, if packet loss or discard occurs, each lost or discarded packet would be preceded by and followed by a sequence of at least 16 received non-discarded packets.

Lost or discarded packets that occur within Gmin packets of a report being generated may be reclassified as part of a burst or gap in later reports.

R Factor

(NSR) A voice quality metric describing the segment of the call that is carried over this RTP session. Valid range: **`0`** to **`100`**, with a value of **`94`** corresponding to toll quality voice and values of **`50`** or less regarded as unusable. This metric is defined as including the effects of delay, consistent with ITU-T G.107 [6] and ETSI TS 101 329-5 [3].

A value of **`127`** indicates that this parameter is unavailable.

External R Factor

(XSR) A voice quality metric describing the segment of the call that is carried over a network segment external to the RTP segment. The valid range and interpretation are the same as defined for the RTP R factor described above. This metric includes the effects of delay, consistent with ITU-T G.107 and ETSI TS 101 329-5, and relates to the outward voice path from the VoIP termination for which this metrics block applies.

A value of **`127`** indicates that this parameter is unavailable.

A value of **`0`** indicates that no measurement was obtained.

Jitter Buf Adaptive

(JBA) Describes the jitter buffer adaptation; one of the following:

Adaptive

The jitter buffer size is being dynamically adjusted to deal with varying levels of jitter. See the jitter buffer size parameters below for details.

Non-adaptive

The jitter buffer size is maintained at a fixed level. See the jitter buffer size parameters below for details.

Jitter Buf Rate

(JBR) The adjustment rate (J) in adaptive mode, referred to as “JB rate” in RFC 3611. Valid range: **`0`** to **`15`**. This parameter is defined in terms of the approximate time taken to fully adjust to a step change in peak to peak jitter from 30 ms to 100 ms such that:

adjustment time = 2 × J × frame size (ms)

A value of **`0`** indicates that the adjustment time is unknown.

JB Nominal Delay

(JBN) The current nominal jitter buffer delay, in milliseconds, which corresponds to the nominal jitter buffer delay for packets that arrive exactly on time.

JB Max Delay (JB max)

(JBM) The current maximum jitter buffer delay in milliseconds, which corresponds to the earliest arriving packet that would not be discarded. In adaptive jitter buffer implementations, this value may dynamically vary up to JB abs max (see below).

JB Abs. Max Delay (JB abs max)

(JBS) The absolute maximum delay in milliseconds, that the adaptive jitter buffer can reach under worst case conditions. If this value exceeds 65535 milliseconds, the report shows a value of **`65535`**. This parameter MUST be provided for adaptive jitter buffer implementations and its value MUST be set to JB max for fixed jitter buffer implementations.

Tx Packets

(PS) The number of RTP packets transmitted during this call.

Tx Octets

(OS) The number of octets transmitted during this call.

Rx Packets

(PR) The number of RTP packets received during this call.

Rx Octets

(OR) The number of octets received during this call.

Packet Loss

(PL) The percentage of RTP packets lost during this call.

Jitter

(IAJ) The estimated statistical variance of the RTP data packet interarrival time, in milliseconds.

When remote VQM is enabled, the eDVA also displays remote VQM data. Remote metrics have the same definitions as local metrics. See “Collecting Remote Metrics” for details.

Interpreting Results

The following table describes the VQM metrics and (where applicable) acceptable ranges. See RFC 3611 for full descriptions of those metrics. Metric names marked with an asterisk (*) are extensions to RFC 3611.

Metric	Description	Ranges	
		Allowed	Good
CW Errors*	Codeword error count	n/a	n/a
CW Error Rate*	Errored codeword ratio	n/a	$\leq 9.0^{-7}$
SNR*	Signal to noise ratio	n/a	≥ 35 dB
Microreflections*	Microreflection power[Note 1]	n/a	Multiple echoes: −20 dBc @ ≤1.5 µsec −30 dBc @ ≥1.5 µsec −10 dBc @ ≤0.5 µsec −15 dBc @ ≥1.0 µsec Single echo: −10 dBc @ ≤0.5 µsec −20 dBc @ ≤1.0 µsec −30 dBc @ >1.0 µsec
Downstream Power*	Downstream power at the eDVA	n/a	−3 to 3
Upstream Power*	Upstream power at the eDVA	n/a	50 to 60
EQI Average*	Average Echo Quality Index	0 to 1	> 0.6[Note 2]
EQI Minimum*	Minimum measured EQI	0 to 1	> 0.3[Note 2]

Metric	Description	Ranges	
EQI Maximum*	Maximum measured EQI	0 to 1	> 0.7[Note 2]
EQI Instantaneous*	Last measured EQI	0 to 1	> 0.6[Note 2]
MOS-LQ	Listening Quality score	1.0 to 5.0	3.5+
MOS-CQ	Conversational Quality score	1.0 to 5.0	3.5+
Echo Return Loss	Residual echo return loss, in dB	0 to 255	40+
Signal Level	Relative voice signal level, in dB, compared to dBm0	−128 to 127	−15 to −20
Noise Level	Silent period background noise level, in dB, compared to dBm0	−128 to 127	< −70
Loss Rate	Fraction of RTP data packets lost	0 to 255	0
Pkt Loss Concealment	How the eDVA is concealing lost packets	n/a	n/a
Discard Rate	Fraction of RTP data packets discarded	0 to 255	0
Burst Density	Fraction of RTP data packets lost or discarded within burst periods	0 to 255	0
Gap Density	Fraction of RTP data packets lost or discarded within inter-burst gaps	0 to 255	0
Burst Duration	Mean duration, in milliseconds, of burst periods	0 to 65535	0
Gap Duration	Mean duration, in milliseconds, of gap periods	0 to 65535	near Call Duration
Round Trip Delay	Most recently calculated round trip time between RTP interfaces, in milliseconds	0 to 65535	0 to 60
End System Delay	Most recently calculated end system delay, in milliseconds	0 to 65535	0 to 120
Gmin	Gap threshold	0 to 255	16
R Factor	Voice quality of the call carried over this RTP session	0 to 100	Voice: 90+ Fax: 70–90
External R Factor	Voice quality of the call carried over network segments external to this RTP session	0 to 100	90+
Jitter Buf Adaptive	Jitter Buffer adaptation	n/a	Voice calls: Adaptive Fax calls: Fixed
Jitter Buf Rate	The adjustment rate in adaptive mode	0 to 15	0 to 15
JB Nominal Delay	Current nominal jitter buffer delay, in milliseconds	0 to 65535	Voice: approx. 20ms Fax: approx. 70ms
JB Max Delay	Current maximum jitter buffer delay, in milliseconds	0 to 65535	0 to 60
JB Abs Max Delay	Absolute maximum jitter buffer delay, in milliseconds	0 to 65535	0 to 60
Tx Packets	Number of RTP packets transmitted	n/a	n/a
Tx Octets	Number of octets transmitted	n/a	n/a

Metric	Description	Ranges	
Rx Packets	Number of RTP packets received	n/a	n/a
Rx Octets	Number of octets received	n/a	n/a
Packet Loss	Percentage of packets lost	0 to 100	< 1
Jitter	Estimated jitter, in milliseconds	n/a	approx. 2 ms

Note 1: The value for microreflections is displayed as a positive number, but is actually negative.

Note 2: EQI measures the *potential* for poor echo quality. Minimum values in particular may be less than 0.2 for brief durations, especially near the beginning of a call, without subscribers noticing any problems. Short calls (less than 30 seconds) in particular may not allow for accurate EQI measurements.

Voice Quality Monitoring MIB Objects

The **arrisMtaDevVqm** MIB provides objects for controlling the Voice Quality Monitoring (VQM) feature and for retrieving VQM data.

The following objects are available.

arrisMtaDevVqmLine
Specifies the line for Voice Quality Metrics reporting.

arrisMtaDevVqmClear
Clears Voice Quality Metrics: **`single-line`**(0) or **`all-lines`**(1). To clear a single line, specify the line using **arrisMtaDevVqmLine**.

arrisMtaDevVqmEnable
Enables or disables recording of Local Voice Metrics: **`disable`**(0) or **`enable`**(1) (the default).

arrisMtaDevVqmCallNumberTable
Voice Quality Metric history on a per call number basis. This table contains up to ten entries, consisting of the following object:

arrisMtaDevVqmCallNumberIds
Provides a history of call number identifiers for the specified line.

arrisMtaDevVqmCallNumberIdentifier
Specifies the call number for Voice Quality Metrics reporting. Valid IDs are obtained from **arrisMtaDevVqmCallNumberIds**.

arrisMtaDevVqmMetricTable
Voice Quality Metrics pertaining to a particular call number. Entries are indexed by metric number, and contain the following objects:

arrisMtaDevVqmMetricValues

The call data. Walk the **arrisMtaDevVqmMetricTable** to retrieve the data in the buffer.

arrisMtaDevVqmThresholds

The thresholds associated with this metric. The eDVA generates a Syslog message if VQM data scores fall below the specified thresholds.

arrisMtaDevVqmEnableRemote

Determines the policy to control Remote Voice Metrics (XR):

- `normal`(0) (default) — obey directives from the Call Server.
- `forceDisable`(1) — disables XR, overriding any directive from the Call Server.
- `forceEnable`(2) — enables XR, overriding any directive from the Call Server.
- `default`(3) — resets this object to the default and removes persistence.

The value of this object is persistent if it is set after configuration.

arrisMtaDevVqmThresholdEnable

The maximum number of logs allowed to be sent in a report when a threshold is exceeded. The more logs that are allowed, the more metrics that can be reported. A value of `0` (default) disables threshold reporting.

The eDVA reports metrics in the order specified by the **arrisMtaDevVqmMetricIndex**. To minimize network traffic, logs are sent only when a threshold is first exceeded. Thereafter until the metric returns to normal, logs are inhibited. To send logs each time the threshold is exceeded, add 100 to the value.

See see "*Setting VQM Thresholds* (page 264) for more details.

arrisMtaDevVqmHistorySize

Sets the size of the VQM history buffer. Valid range: `2` to `50` records. Default: `50`.

arrisMtaDevVqmCallNumberIdentifierLastCall

(read-only) The last call identifier for Voice Quality Metrics reporting.

Collecting Remote Metrics

Touchstone firmware supports recording and display of far-end voice quality metrics. The following conditions must be true for the eDVA to receive far-end metrics:

- The remote eDVA must support the RTCP-XR protocol.
- The call must be up long enough to receive remote data (remote metrics not received display a value of “Call too Short”).

By default, Touchstone firmware complies with PacketCable 1.5 NCS specifications (PKT-SP-NCS1.5-I03-070412) for enabling and sending remote VQM data. For NCS loads and default operation, eDVAs send VQM data:

1. if the Call Server has not instructed the eDVA to disable remote VQM; and
2. the remote eDVA requests VQM data.

Note: The **arrisMtaDevVqmEnableRemote** MIB object can force remote metric data, either on or off, for all loads. In NCS operation, this object can override directives from the Call Server.

Action

Perform the following tasks as needed.

Configuring Local VQM Reporting

Follow these steps to configure local Voice Quality Monitoring using SNMP (the web-based interface does not support enabling or disabling VQM). VQM is enabled by default with a history buffer size of 10.

1. Set the **arrisMtaDevVqmEnable** object to control local collection of VQM data. Use `0` to disable local VQM, or `1` to enable local VQM.
2. Set the **arrisMtaDevVqmEnableRemote** object to control remote reporting of VQM data. Use `normal`(0) to use PacketCable rules; `forceDisable`(1) to force-disable remote VQM; or `forceEnable`(2) to force-enable remote VQM. When set through SNMP, the setting persists across reboots.
3. To set the size of the VQM buffer, set the **arrisMtaDevVqmHistorySize** object.

   Valid range: `2` to `50` (calls). Default: `10`.

Retrieving the Last Call Using SNMP

Follow these steps to retrieve data for the last completed call using SNMP. See see "*Retrieving VQM Data Using SNMP* (page 260) to retrieve arbitrary call data.

1. Read the **arrisMtaDevVqmCallNumberIdentifierLastCall** object.

   The eDVA updates the **arrisMtaDevVqmCallNumberIdentifier** object and loads the **arrisMtaDevVqmMetricTable** with the data for the last call.
2. Walk the **arrisMtaDevVqmMetricTable** to read the metrics from the specified buffer.

Retrieving VQM Data Using SNMP

Retrieving arbitrary VQM data through SNMP is more complex than web-based methods, requiring a multi-step process. To retrieve data for the last completed call, see see "*Retrieving the Last Call Using SNMP* (page 259) above.

Follow these steps to retrieve VQM data through SNMP. See Voice Quality Monitoring MIB Objects for detailed descriptions of VQM-related MIB objects.

1. Write the desired line number to the **arrisMtaDevVqmLine** object. Valid range: `1` to the number of lines supported by the eDVA.

   The eDVA loads the **arrisMtaDevVqmCallNumberTable** with pointers to reports for up to the last 10 calls.

2. Walk the **arrisMtaDevVqmCallNumberTable** to read the buffer IDs:

```
***** SNMP QUERY STARTED *****
1: arrisMtaDevVqmCallNumberIds.1 (integer) 19
2: arrisMtaDevVqmCallNumberIds.2 (integer) 16
3: arrisMtaDevVqmCallNumberIds.3 (integer) 14
4: arrisMtaDevVqmCallNumberIds.4 (integer) 12
5: arrisMtaDevVqmCallNumberIds.5 (integer) 11
6: arrisMtaDevVqmCallNumberIds.6 (integer) 9
7: arrisMtaDevVqmCallNumberIds.7 (integer) 6
8: arrisMtaDevVqmCallNumberIds.8 (integer) 4
9: arrisMtaDevVqmCallNumberIds.9 (integer) 2
10: arrisMtaDevVqmCallNumberIds.10 (integer) 1
***** SNMP QUERY FINISHED *****
```

   Entry 1 is the buffer containing the most recent call (in the above example, buffer 19). Entries containing `Empty` represent no data for that call (for example, less than 10 calls have been made since the MTA started or since VQM data was cleared).

3. Write the desired buffer number to the **arrisMtaDevVqmCallNumberIdentifier** object.

   The eDVA loads the **arrisMtaDevVqmMetricTable** with the contents of the buffer.

4. Walk the **arrisMtaDevVqmMetricTable** to read the metrics from the specified buffer:

```
***** SNMP QUERY STARTED *****
1: arrisMtaDevVqmMetricValues.1 (octet string) Call End Time: THU JAN 10
16:59:37 2008
2: arrisMtaDevVqmMetricValues.2 (octet string) Call Start Time: THU JAN
10 16:58:21 2008
3: arrisMtaDevVqmMetricValues.3 (octet string) Call Duration: 0h 1m 16s
4: arrisMtaDevVqmMetricValues.4 (octet string) Line Number: 2
5: arrisMtaDevVqmMetricValues.5 (octet string) Remote IP Address:
10.1.36.151
6: arrisMtaDevVqmMetricValues.6 (octet stri CW Errors: 0
7: arrisMtaDevVqmMetricValues.7 (octet string) CW Error Rate: 0.00e+00
8: arrisMtaDevVqmMetricValues.8 (octet string) SNR: 36 dB
9: arrisMtaDevVqmMetricValues.9 (octet string) MicroReflections: 29 dBc
10: arrisMtaDevVqmMetricValues.10 (octet string) Downstream Power: -6.0
dBmV
11: arrisMtaDevVqmMetricValues.11 (octet string) Upstream Power: 44.0
dBmV
12: arrisMtaDevVqmMetricValues.12 (octet string) EQI Average: 0.468
13: arrisMtaDevVqmMetricValues.13 (octet string) EQI Minimum: 0.389
14: arrisMtaDevVqmMetricValues.14 (octet string) EQI Maximum: 0.595
15: arrisMtaDevVqmMetricValues.15 (octet string) EQI Instantaneous:
0.389
16: arrisMtaDevVqmMetricValues.16 (octet string) MOS-LQ: 3.3
```

```
17: arrisMtaDevVqmMetricValues.17 (octet string) MOS-CQ: 3.3
18: arrisMtaDevVqmMetricValues.18 (octet string) Echo Return Loss:
Unavailable
19: arrisMtaDevVqmMetricValues.19 (octet string) Signal Level: -21 dBm0
20: arrisMtaDevVqmMetricValues.20 (octet string) Noise Level: -58 dBm0
21: arrisMtaDevVqmMetricValues.21 (octet string) Loss Rate: 0 %
22: arrisMtaDevVqmMetricValues.22 (octet string) Pkt Loss
Concealment:Standard
23: arrisMtaDevVqmMetricValues.23 (octet string) Discard Rate: 0 %
24: arrisMtaDevVqmMetricValues.24 (octet string) Burst Density: 0 %
25: arrisMtaDevVqmMetricValues.25 (octet string) Gap Density: 12 %
26: arrisMtaDevVqmMetricValues.26 (octet string) Burst Duration: 0 ms
27: arrisMtaDevVqmMetricValues.27 (octet string) Gap Duration: 65+
seconds
28: arrisMtaDevVqmMetricValues.28 (octet string) Round Trip Delay: 5 ms
29: arrisMtaDevVqmMetricValues.29 (octet string) End System Delay: 57 ms
30: arrisMtaDevVqmMetricValues.30 (octet string) Gmin: 16 packets
31: arrisMtaDevVqmMetricValues.31 (octet string) R Factor: 67
32: arrisMtaDevVqmMetricValues.32 (octet string) External R Factor: 0
33: arrisMtaDevVqmMetricValues.33 (octet string) Jitter Buf Adaptive:
Adaptive
34: arrisMtaDevVqmMetricValues.34 (octet string) Jitter Buf Rate: 15
35: arrisMtaDevVqmMetricValues.35 (octet string) JB Nominal Delay: 12 ms
36: arrisMtaDevVqmMetricValues.36 (octet string) JB Max Delay: 25 ms
37: arrisMtaDevVqmMetricValues.37 (octet string) JB Abs. Max Delay: 60
ms
38: arrisMtaDevVqmMetricValues.38 (octet string) Remote MOS-LQ: No
Remote Data
39: arrisMtaDevVqmMetricValues.39 (octet string) Remote MOS-CQ: No
Remote Data
40: arrisMtaDevVqmMetricValues.40 (octet string) Remote Echo Return
Loss: No Remote Data
41: arrisMtaDevVqmMetricValues.41 (octet string) Remote Signal Level: No
Remote Data
42: arrisMtaDevVqmMetricValues.42 (octet string) Remote Noise Level: No
Remote Data
43: arrisMtaDevVqmMetricValues.43 (octet string) Remote Loss Rate: No
Remote Data
44: arrisMtaDevVqmMetricValues.44 (octet string) Remote Pkt Loss
Concealment:No Remote Data
45: arrisMtaDevVqmMetricValues.45 (octet string) Remote Discard Rate: No
Remote Data
46: arrisMtaDevVqmMetricValues.46 (octet string) Remote Burst Density:
No Remote Data
47: arrisMtaDevVqmMetricValues.47 (octet string) Remote Gap Density: No
Remote Data
48: arrisMtaDevVqmMetricValues.48 (octet string) Remote Burst Duration:
No Remote Data
49: arrisMtaDevVqmMetricValues.49 (octet string) Remote Gap Duration: No
Remote Data
50: arrisMtaDevVqmMetricValues.50 (octet string) Remote Round Trip
Delay: No Remote Data
51: arrisMtaDevVqmMetricValues.51 (octet string) Remote End System
Delay: No Remote Data
52: arrisMtaDevVqmMetricValues.52 (octet string) Remote Gmin: No Remote
Data
53: arrisMtaDevVqmMetricValues.53 (octet string) Remote R Factor: No
Remote Data
54: arrisMtaDevVqmMetricValues.54 (octet string) Remote External R
Factor: No Remote Data
55: arrisMtaDevVqmMetricValues.55 (octet string) Remote Jitter Buf
Adaptive: No Remote Data
56: arrisMtaDevVqmMetricValues.56 (octet string) Remote Jitter Buf Rate:
No Remote Data
```

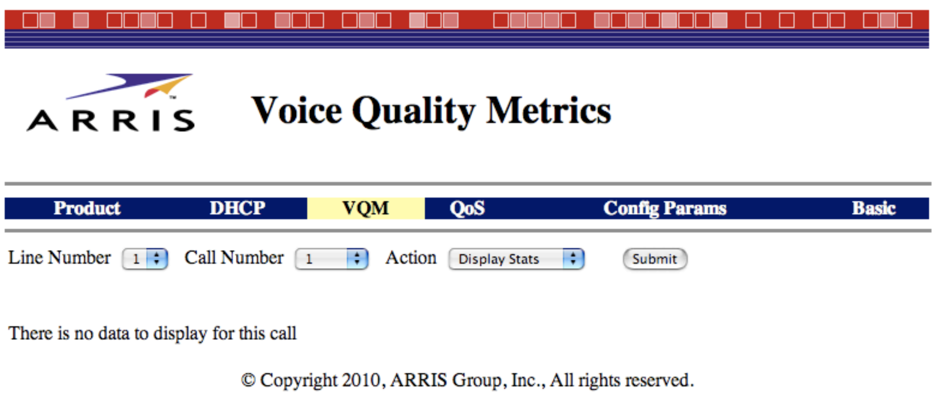
```
57: arrisMtaDevVqmMetricValues.57 (octet string) Remote JB Nominal
Delay: No Remote Data
58: arrisMtaDevVqmMetricValues.58 (octet string) Remote JB Max Delay: No
Remote Data
59: arrisMtaDevVqmMetricValues.59 (octet string) Remote JB Abs. Max
Delay: No Remote Data
60: arrisMtaDevVqmMetricValues.60 (octet string) Tx Packets: 7584
61: arrisMtaDevVqmMetricValues.61 (octet string) Tx Octets: 606720
62: arrisMtaDevVqmMetricValues.62 (octet string) Rx Packets: 7582
63: arrisMtaDevVqmMetricValues.63 (octet string) Rx Octets: 606560
64: arrisMtaDevVqmMetricValues.64 (octet string) Packet Loss: 1
65: arrisMtaDevVqmMetricValues.65 (octet string) Interval Jitter: 0 ms
***** SNMP QUERY FINISHED *****
```

Retrieving VQM Data Using the Web-based Interface

Follow these steps to retrieve VQM data using the web-based interface.

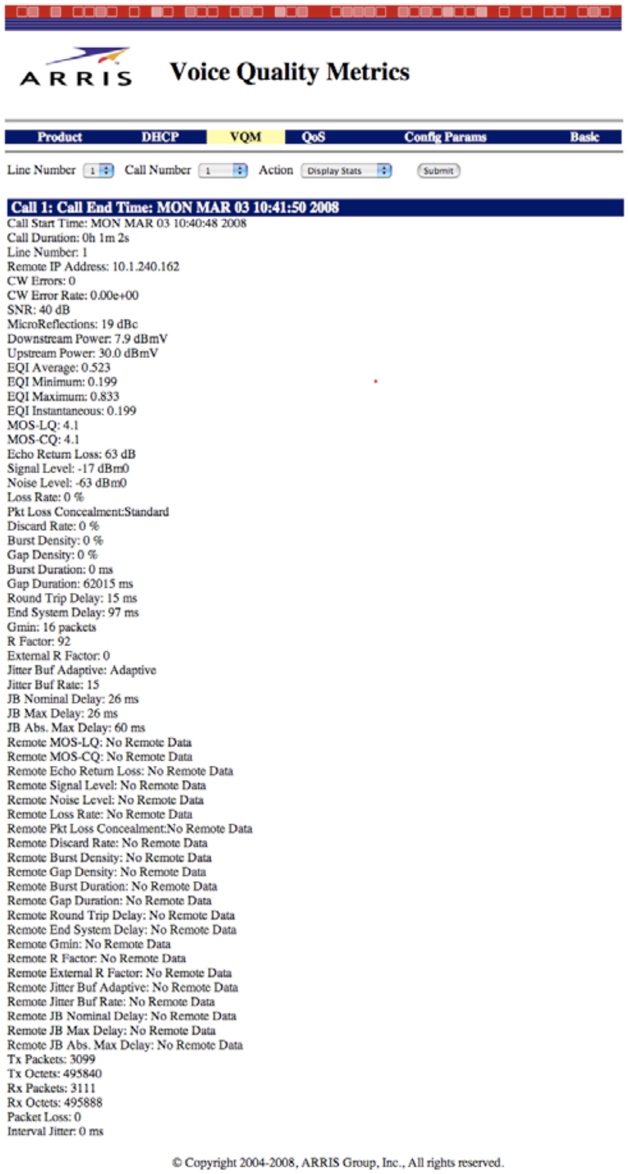
1. Access the Advanced web pages as described in see "*Accessing the Advanced Pages* (page 305).
2. Select the **VQM** link from the menu bar.

   The Voice Quality Metrics page displays. If there have been no calls since the last reboot, the page displayed is blank as shown below.



3. Select the line number from the Line Number drop-down. Valid range: `1` to the number of lines on the eDVA.
4. Select the call number from the Call Number drop-down. Valid range: `1` to `10`, or **All** to display all call data in a report format, or **Table** to display all call data in a table format.
5. Click the **Submit** button.

The eDVA displays the selected data. Examples:



Call Number	1 (Local)	1 (Remote)
Call End Time:	Mon Oct 19 09:55:56 2009	---
Call Start Time:	Mon Oct 19 09:52:46 2009	---
Call Duration:	0h 3m 10s	---
Line Number:	1	---
Remote IP Address:	10.1.38.139	---
CW Errors:	0	---
CW Error Rate:	0.00e+00	---
SNR:	0.0 dB	---
MicroReflections:	0 dBc	---
Downstream Power:	0.0 dBmV	---
Upstream Power:	0.0 dBmV	---
EQI Average:	Call too Short	---
EQI Minimum:	Call too Short	---
EQI Maximum:	Call too Short	---
EQI Instantaneous:	Call too Short	---
MOS-LQ:	4.1	No Remote Data
MOS-CQ:	4.1	No Remote Data
Echo Return Loss:	53 dB	No Remote Data
Signal Level:	Unavailable	No Remote Data
Noise Level:	-65 dBm0	No Remote Data
Loss Rate:	0 %	No Remote Data
Pkt Loss Concealment:	Standard	No Remote Data
Discard Rate:	0 %	No Remote Data
EQI Maximum:	Call too Short	---
EQI Instantaneous:	Call too Short	---
MOS-LQ:	4.1	No Remote Data
MOS-CQ:	4.1	No Remote Data
Echo Return Loss:	53 dB	No Remote Data
Signal Level:	Unavailable	No Remote Data
Noise Level:	-65 dBm0	No Remote Data
Loss Rate:	0 %	No Remote Data
Pkt Loss Concealment:	Standard	No Remote Data
Discard Rate:	0 %	No Remote Data
Burst Density:	0 %	No Remote Data
Gap Density:	0 %	No Remote Data
Burst Duration:	0 ms	No Remote Data
Gap Duration:	65+ seconds	No Remote Data
Round Trip Delay:	0 ms	No Remote Data
End System Delay:	67 ms	No Remote Data
Gmin:	16 packets	No Remote Data
R Factor:	92	No Remote Data
External R Factor:	0	No Remote Data
Jitter Buf Adaptive:	Adaptive	No Remote Data
Jitter Buf Rate:	15	No Remote Data
JB Nominal Delay:	30 ms	No Remote Data
JB Max Delay:	30 ms	No Remote Data
JB Abs. Max Delay:	120 ms	No Remote Data
Tx Packets:	0	---
Tx Octets:	0	---
Rx Packets:	0	---
Rx Octets:	0	---
Packet Loss:	0	---
Interval Jitter:	0 ms	---
Originator:	yes	---
Remote Interval Jitter:	No Remote Data	---

Setting VQM Thresholds

Follow these steps to set threshold monitoring. The eDVA sends a Syslog notification if a call does not meet the specified thresholds.

1. Configure event reporting for the “VQ Threshold” log as described in the "Operations" chapter. The Event ID for this log is **49**.
2. To enable threshold monitoring, set the **arrisMtaDevVqmThresholdEnable** object to the maximum number of “VQ Threshold” logs desired for reporting a threshold crossing.

   In practice, a single call may send up to three logs to report metrics for a threshold crossing. To minimize unnecessary network traffic, Touchstone firmware only reports metrics for which values were obtained (for example, if the eDVA does not receive remote metrics, it does not report any).
3. Set the instances in the **arrisMtaDevVqmThresholds** table to monitor VQM data as follows:

Index	Parameter
.16	MOS-LQ score (0.1 to 5.0)
.17	MOS-CQ score (0.1 to 5.0)

4. If you have added these objects to the eDVA configuration file, reset the eDVA to allow the changes to take effect.

Clearing VQM Data

Follow these steps to clear VQM data for either a single line or all lines.

1. To clear VQM data using SNMP:
   a. To clear a single line, set the **arrisMtaDevVqmLine** object to the desired line number.
      Note: You can skip this step if you want to clear all lines.
   b. Write one of the following values to the **arrisMtaDevVqmClear** object:
      `0` — clear a single line
      `1` — clear all lines
2. To clear VQM data using the web-based interface:
   a. Select the line to clear from the **Line Number** drop-down menu.
      Note: You can skip this step if you want to clear all lines.
   b. To clear the line selected in Line Number, select **Clear Line Stats** from the **Action** drop-down menu. To clear all lines, select **Clear All Lines**.
   c. Click the **Submit** button to perform the selected action.

Using the Speedtest Application

The Speedtest application allows remote testing of download and upload speeds between the modem and an external server, without using a CPE.

Measuring downstream and upstream speed is done by transferring a file between a network server and the FTP client embedded in the modem. Average throughput is calculated from the size of the file transferred and the total transfer time.

Speed tests can be conducted from the Web GUI or SNMP. Touchstone firmware stores the results of the last three tests. Syslog reporting is optional.

Tests may run whether or not phone lines are in use and tests do not impact calls in progress while the test is running. If phone calls or connected CPE devices are active, the speed test includes that traffic in its results.

Requirements and Limitations

TS11.1 supports Speedtest on TM3402 Telephony Modems.

Server Requirements

To conduct speed tests, an HTTP or FTP server must be available. Server location can be used to determine throughput in a variety of ways:

Server Location	Throughput Test
Local headend	HFC throughput
Master headend	Internal network throughput
External site	Peering throughput

An HTTP server used for Uplink testing must meet the following requirements:

- HTTP 1.1 support
- Support for an upload submission form, with an input element of type FILE
- The server must provide a script to process POST HTTP requests, in accordance with RFC 1867

The open-source Apache server supports all requirements and is widely available.

Testing Limitations

The following limitations apply to speed tests in general:

- Rate limiting at the CM is not disabled when running a speed test.
- Speed tests do not run when the modem is operating on battery power.
- Speed tests always run as Best Effort traffic.
- On Media Gateway products, transfer speeds are measured at the CM interface, not the Router interface.
- Regardless of the actual throughput, the speed test measures only to 80 Mbps downstream and 40 Mbps upstream.

The following items are not configurable:

- TCP Selective Acknowledgement (SACK) is always enabled.

The HTTP version is always 1.1.

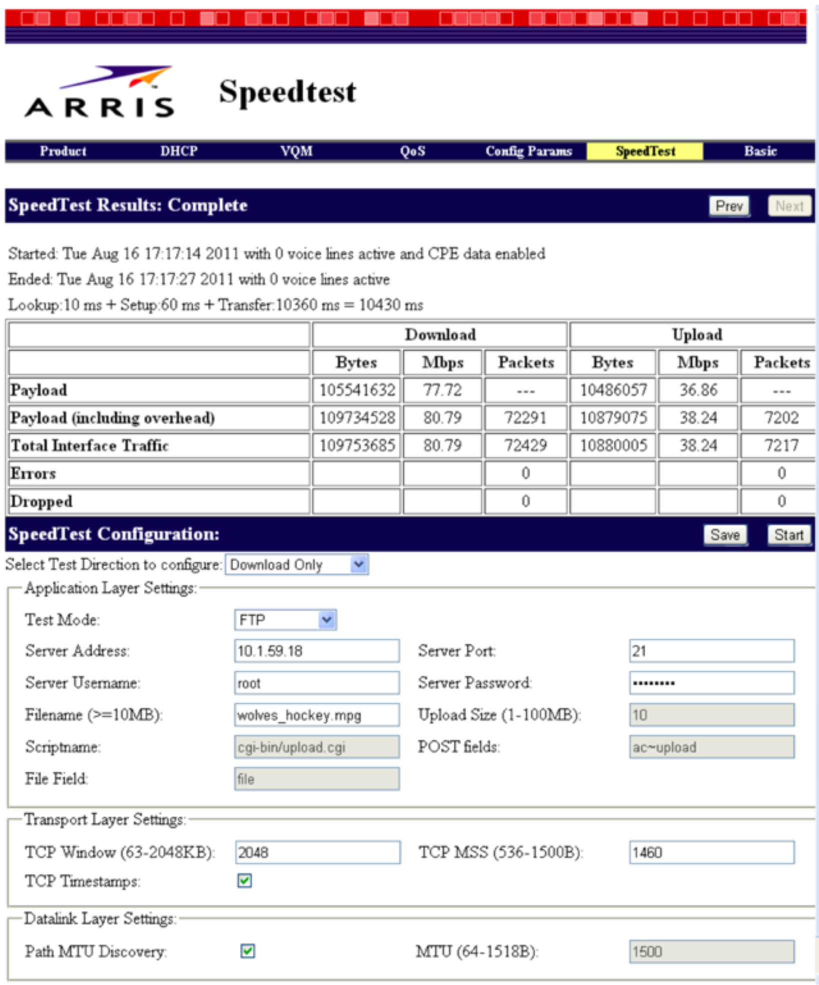
Action

Perform the following tasks as needed.

Running a Speed Test using the Web Pages

Follow these steps to use the Speedtest web page. Note that this requires access to the Advanced web pages.

1. Access the Advanced web pages as described in see "*Accessing the Advanced Pages* (page 305).
2. Click the SpeedTest link to display the Speedtest page:



3. Configure the test as follows:

Test direction

Select **Download Only** or **Upload Only** as desired.

Test mode

Select **FTP** or **HTTP** as desired.

Server address

Enter the IP address or FQDN of the server.

Port

Enter the port number for the server. The defaults are `21` for FTP, and `80` for HTTP.

Server Username

Enter the user ID required to access the server. Required for FTP, optional for HTTP unless authentication is required.

Server Password

Enter the password required to access the server. Required for FTP, optional for HTTP unless authentication is required.

Filename

Enter the name of the file to transfer. For Download tests, the modem discards the information received. For Upload tests, the modem sends random data to the server.

Note: The following parameters are optional. You can accept the defaults.

TCP Window

The size, in KB, of the receive buffer used in the transfer.

TCPMSS

The maximum TCP segment size, in bytes.

TCP Timestamps

Check this box to enable TCP timestamping.

Path MTU Discovery

Check this box to negotiate the MTU size for the transfer.

MTU

When **Path MTU Discovery** is not checked, specifies the size (in bytes) for the MTU.

4. Click **Save**.
5. Repeat steps 3 and 4 to configure test parameters for the other direction (download vs. upload), if desired.
6. To test both upload and download speeds, select **Download / Upload** from the **Test direction** drop down menu.
7. Click **Start**.

   The **Start** button changes to **Cancel**.
8. Allow the test to run, or click **Stop** to end the test before the transfer is complete.

   The test results display in the upper section of the Speedtest page.

Running a Speed Test using SNMP

Follow these steps to run a speed test using SNMP.

1. To configure the download test, set the **arrisSpeedTestConfigDownlinkURL** object to the URL specifying the server and file used for the transfer. The URL begins with either `ftp://` or `http://`, and is followed by:

   [*userID*:*password*@]*ipaddr*/*path*[?*param*=*value*[&*param*=*value*...]]

   The *userID* and *password* are required for FTP transfers, and optional for HTTP transfers. The *param* list may consist of any of the following:

tcpwindow

The size, in KB, of the receive buffer used in the transfer.

tcpmss

The maximum TCP segment size, in bytes.

tcptimestamps

Set to **`1`** to enable TCP timestamping.

pmtud

Set to **`1`** to negotiate the MTU size for the transfer.

mtu

When **pmtud** is set to **`0`**, specifies the size (in bytes) for the MTU.

cpeaccess

Set to **`1`** to allow CPE network access during the speed test, or **`0`** to disable CPE access.

Example:

```
ftp://root:xxxxxxxx@10.1.59.18/wolves_hockey.mpg?
tcpwindow=2048&tcpmss=1460&tcptimestamps=1
&pmtud=1&cpeaccess=1
```

2. To configure the upload test, set the **arrisSpeedTestConfigUplinkURL** object to the URL specifying the server and file used for the transfer. The format of the URL is the same as for the download object described above.
3. To start the test, write **`startDownlinkTest`**(1) or **`startUplinkTest`**(2) (or both) to the **arrisSpeedTestConfigStartStopTest** object as needed.
4. Allow the test to complete, or write **`stopTest`**(0) to the **arrisSpeedTestConfigStartStopTest** object to end the test early.
5. To display the results of the last test, read the **arrisSpeedTestResultsStatus**.1 object.

Network Performance Monitoring

Network Performance Monitoring uses the CM component of a Touchstone product to test the performance of individual network legs or end-to-end connectivity. All testing can be performed from the headend or NOC without subscriber intervention.

The tests provide results in JSON format, for easy database entry, parsing, and consolidation.

Note 1: In this release, Network Performance Monitoring supports only IPv4 networks.

Test Types

The following tests are supported:

- webpage download
- DNS latency
- network latency

Setup

Setup is SNMP-based. Add MIB objects from the **arrisNetPerfMonitorMib** as needed to configure Network Performance Monitoring.

The following sections describe how to set up Network Performance Monitoring.

Configuring Traffic Control

Set the following objects to control Network Performance Monitoring impact on the device and local network.

arrisNpmSetupBgTrafficRateEnable
Set to **`enable`**(1) to use background traffic rate checking. When enabled, the Touchstone device does not run performance monitoring tests if the upstream or downstream traffic exceed the rates set by the next two objects.

arrisNpmSetupBgTrafficMaxDownstreamRate
The maximum downstream traffic rate, in Kbps, allowed for Network Performance Monitoring tests to run.

arrisNpmSetupBgTrafficMaxUpstreamRate
The maximum upstream traffic rate, in Kbps, allowed for Network Performance Monitoring tests to run.

Traffic rate checking uses the **ifHCInOctets** and **ifHCOutOctets** objects associated with the DOCSIS interface (**ifIndex** 2, 3, and 4). If the traffic exceeds the current threshold, the modem:

1. Pauses rate checking for 10 seconds
2. Checks the traffic rate over the next 5 seconds
3. Repeats up to 5 times before abandoning the tests

Setting a Group Reference

Each device includes its CM MAC address in the results, for unique identification. By setting the **arrisNpmSetupGroupReference** object, you can associate devices for average and other aggregation.

The group reference is an arbitrary text string. Suggested grouping identifiers include:

- headend
- CMTS card slot
- fiber node

Using a hierarchical group identifier allows aggregation to be highly flexible.

Configuring Tests

Each test has a group of MIB objects used to configure and control the test. The following sections describe how to set up each test.

Configuring the Webpage Download Test

The **arrisNpmWebDlTest** objects configure and control this test. The Webpage Download test downloads up to 10 configured web pages, and measures the time taken.

To set the timeout for this test, set the **arrisNpmSetupWebPageDlTestTimeout** object to the desired time, in seconds. This timeout applies to each page downloaded, not the entire test.

To configure URLs to download for the test, set the following two objects in the **arrisNpmSetupWebPageDlTestTable**:

arrisNpmSetupWebPageDlTestConfigUrl
The URL of the webpage to download. Use an index value between `1` and `10`.

arrisNpmSetupWebPageDlTestConfigRowStatus
Set to `createAndGo`(4) to add the URL to the table.

Configuring the DNS Latency Test

The DNS Latency test uses the entries in the **arrisNpmSetupWebPageDlTestTable**, recording the time required to perform a DNS resolve of each URL in the table.

By default, the DNS Latency test uses the DNS servers assigned to the CM through DHCP. If you want to override the configured DNS servers, set the following objects to the IP address of the desired servers:

- **arrisNpmSetupDnsPrimaryServerIpAddress**
- **arrisNpmSetupDnsSecondaryServerIpAddress**

If you want to run the DNS Latency test alongside the Webpage Download test, set the **arrisNpmSetupDnsTestEnable** object to `enable`(1). When enabled, the DNS Latency test automatically runs when you initiate a Webpage Download test.

Configuring the Network Latency Test

The Network Latency test requires one or more dedicated UDP Ping servers to be deployed in the MSO network. The **arrisNpmSetupNetLatencyServerTable** designates up to five Ping servers for the Touchstone device to use for the test.

To configure the Ping servers, set the following objects:

arrisNpmSetupNetLatencyConfigServer
The IP address or FQDN of a Ping server.

arrisNpmSetupNetLatencyConfigServerPort
The UDP port where the Ping server listens for pings.

arrisNpmSetupNetLatencyConfigServerRowStatus
Set to **createAndGo**(4).

Configure the following optional objects, as needed:

arrisNpmSetupNetLatencyTestPingCount
The number of pings to send to each configured Ping server. Valid range: **1** to **10**. Default: **1**.

arrisNpmSetupNetLatencyTestPingInterval
The time, in milliseconds, between each ping. Valid range: **1** to **36000000**. Default: 5**0**.

arrisNpmSetupNetLatencyTestPingTimeout
The time, in milliseconds, to wait for a response from the Ping server before declaring the request failed. Valid range: **1000** to **6000**. Default: **3000**.

arrisNpmSetupNetLatencyTestRunUnderLoadEnable
Set this object to **enable**(1) to repeatedly run the Network Latency test during the Webpage Download test. This allows testing latency when the Touchstone device has a CM-originated network load applied to the default service flow. When this option is enabled, the Ping interval is 300ms.

Running Tests

Network Performance tests can be run separately or in a group. The Webpage Download test is the “anchor” test for running multiple tests.

Running the Webpage Download Test

To run the Webpage Download test, write a numeric value to the **arrisNpmSetupWebPageDlTestRunTime** object. The value is a countdown timer, in seconds, used to delay the start of the test. Write a value of **0** to start the test immediately.

Note: Wait at least five minutes for the test to complete before checking the results.

Running the DNS Latency Test

You can run the DNS Latency test by itself, or as part of the Webpage Download test:

- To run the test with the Webpage Download test, set the **arrisNpmSetupDnsTestEnable** object to **enable**(1).
- To run the DNS Latency test by itself, write to the **arrisNpmSetupDnsTestRunTime** object. The value is a countdown timer, in seconds, used to delay the start of the test. Write a value of **0** to start the test immediately.

Note 1: Wait at least five minutes for the test to complete before checking the results.

Note 2: If the **arrisNpmSetupDnsTestEnable** object is enabled, writing to the **arrisNpmSetupDnsTestRunTime** object has no effect.

Running the Network Latency Test

You can run the Network Latency test by itself, or as part of the Webpage Download test:

- To run the test with the Webpage Download test, set the **arrisNpmSetupNetLatencyTestRunUnderLoadEnable** object to `enable`(1). Note that when this option is enabled, the ping interval is always 300ms.
- To run the Network Latency test by itself, write to the **arrisNpmSetupNetLatencyTestRunTime** object. The value is a countdown timer, in seconds, used to delay the start of the test. Write a value of `0` to start the test immediately.

Note 1: Wait at least five minutes for the test to complete before checking the results.

Note 2: If the **arrisNpmSetupNetLatencyTestRunUnderLoadEnable** object is enabled, writing to the **arrisNpmSetupNetLatencyTestRunTime** object has no effect.

Results

The Network Performance tests output the results in JSON format. The following is an example of a Webpage Download test for one site:

```
{ 'results': { 'time':'2011-02-08 08:49:37', 'cm':'00ca.1231.3939',
     'group':'CMTS-MD-1-0-2', 'url':'www.cnn.com', 'bytes':'12023291',
     'duration':'8923', 'run':'1', 'fail':'0' } }
```

Webpage Download Results

The **arrisNpmResultWebPageDlTestTable** contains the results of the Webpage Download test. The table contains a **arrisNpmResultWebPageDlTestResult** entry for each URL configured in the **arrisNpmSetupWebPageDlTestTable**. The fields in each entry are:

time
The time the test was run, in the format **YYYY-MM-DD HH:MM:SS**.

cm
The CM MAC address.

group
The group string assigned to the device.

url
The URL downloaded.

bytes
The number of bytes transferred.

duration
The time, in milliseconds, elapsed during the test.

run

Returns **1** if the test was run. If **0**, the test was not allowed to run due to background traffic rates.

fail

Returns **0** if the test succeeded, or **1** if the test failed.

DNS Latency Results

The **arrisNpmResultDnsTestTable** contains the results of the DNS Latency test. The table contains an **arrisNpmResultDnsTestResult** entry for each URL configured in the **arrisNpmSetupWebPageDlTestTable**. The fields in each entry are:

time

The time the test was run, in the format **YYYY-MM-DD HH:MM:SS**.

cm

The CM MAC address.

group

The group string assigned to the device.

server

The IP address of the DNS server used.

url

The URL looked up or downloaded. The number of bytes transferred.

duration

The time, in milliseconds, elapsed during the test.

run

Returns **1** if the test was run. If **0**, the test was not allowed to run due to background traffic rates.

fail

Returns **0** if the test succeeded, or **1** if the test failed.

Network Latency Results

The **arrisNpmResultNetLatencyTestTable** contains the results of the Network Latency test. The table contains an **arrisNpmResultNetLatencyTestResult** entry for each Ping server configured in the **arrisNpmSetupWebPageDlTestTable**. The fields in each entry are:

time

The time the test was run, in the format **YYYY-MM-DD HH:MM:SS**.

cm

The CM MAC address.

group

The group string assigned to the device.

server
: The IP address of the DNS server used.

avg
: The average round-trip time (RTT), in milliseconds, measured during the test.

min
: The minimum RTT, in milliseconds, measured during the test.

max
: The maximum RTT, in milliseconds, measured during the test.

median
: The median value of all RTT measurements, in milliseconds.

range
: The difference, in milliseconds, between the maximum and minimum RTT values.

std
: The standard deviation, in milliseconds, of the sample.

run
: The number of Pings sent during the test.

fail
: The number of Ping timeouts or DNS failures during the test.

Maintenance

Maintenance involves upgrades, enabling new features, diagnostics, and troubleshooting.

Overview of Maintenance Interfaces

Touchstone firmware provides the following interfaces for maintenance and troubleshooting:

- Command-line Interface (CLI)
- Web-based interface (WebGUI)
- SNMP
- TR-069

The information below covers the CLI and WebGUI. SNMP is covered elsewhere in this document.

Note: SNMP access from LAN interfaces is disabled once the device has ranged and registered. To allow SNMP access to a test device after ranging and registering, set TLV-55 in the CM configuration file.

WebGUI Access Levels and Defaults

The WebGUI can be accessed from the WAN or LAN interfaces. It has two levels of access:

Subscriber level

Subscriber level access is restricted to user-controlled settings, and is only available at LAN interfaces.

User name: **admin**
Default password: **password** (can be changed by the subscriber)

Technician level

Technician level access allows making changes to all settings available in the WebGUI. It is available at the WAN interface.

User name: **technician**
Default password: MSO-dependent (may be changed using the **arrisCmDoc30AccessTechnicianPassword** object)

LED Patterns

The Touchstone Telephony Modem has eight indicator lights to assist in troubleshooting. Note that not all models have a Battery light.

Battery Mismatch Indication

If the Telephony Modem alternates flashing the Battery light and all other lights, the installed battery is incompatible with the Telephony Modem. Remove the battery and install a supported battery.

Wiring Problems Indication

If the Telephony Modem flashes all its lights for more than 10 seconds, this indicates a problem with the telephone wiring—the tip and ring (red and green) wires may be shorted (touching), or there may be undesired voltage on the lines. If this pattern persists for more than 10 seconds, disconnect the telephone lines from the Telephony Modem, then call a wiring technician for assistance.

TM3402 Normal Operation

The following table shows light patterns during normal operation.

Mode	Power	Ethernet (LEDs on rear panel connector)	US/DS	Online	Tel 1 / Tel 2
AC Power	**On**	**Green LED On** = Computer with 1Gbps port connected **Amber LED On** = Computer with 100 Mbps/10Mbps port connected **Amber/Green LED Flash** = Computer activity **Both LEDs Off** = Computer not connected	**On** = Connected to the Internet **Flash** = Not connected to the Internet	**On** = Internet available Off = Internet not available	**On** = On-hook **Flash** = Off-hook Off = disabled
No AC Power	Off	Off	Off	Off	Off
Firmware Upgrade	**On**	(normal operation)	**Flash**	**On**	(normal operation)

TM3402 Startup Sequence

The following tables show the TM3402 light patterns during each phase of the startup sequence. There are two phases of startup; the telephony phase and the cable modem phase. Both are outlined below.

Telephony Startup Sequence

Power, US/DS, Online	Telephone 1	Telephone 2	Description
Off	Off	Off	No power to modem
Flash	**Flash**	**Flash**	Power-on Self Test
See *Cable Modem Startup Sequence* (page 278)			
On	**Flash**	Off	Retrieving telephone network information
On	Off	**Flash**	Retrieving telephone line information
On	**Flash**	**Flash**	Activating telephone service
Normal Operation			

Cable Modem Startup Sequence

US/DS	Online	Description
Slow Flash (1/second)	Off	Downstream acquisition in progress
On (until Upstream acquisition starts)	Off	Downstream acquisition completed
Fast Flash (3/second)	Off	Upstream acquisition in progress
On	**Slow Flash** (during acquisition) **On** (when modem IP address obtained)	Upstream acquisition completed, ready for service

Running Loop Diagnostics

Use this procedure to run loop diagnostics on a subscriber's phone line.

About Loop Diagnostics

Loop diagnostics include the following tests:

Hazardous Potential Test

Tests for the presence of a foreign AC or DC voltage from tip to ground or ring to ground. Failure conditions are:

- Tip-ground or ring-ground AC voltage is greater than 50 V RMS
- Tip-ground or ring-ground DC voltage is greater than 135 V

The failure types are shown below, with possible causes and symptoms:

Test Failure Data	Possible Cause	Possible Symptoms	Normal Test Limits
-DC Tip-Grd	Tip shorted to other equipment power supply lines	No dial tone, can not dial out, no phones ring, LC in protection state	0 Vdc
-DC Ring-Grd	Ring shorted to other equipment power supply lines	No dial tone, can not dial out, no phones ring, LC in protection state	0 Vdc
+DC Tip-Grd	Tip shorted to other equipment power supply lines	No dial tone, can not dial out, no phones ring, LC in protection state	0 Vdc
+DC Ring-Grd	Ring shorted to other equipment power supply lines	No dial tone, can not dial out, no phones ring, LC in protection state	0 Vdc
AC Tip-Grd	Tip shorted to other equipment AC power supply lines or house wiring	No dial tone, can not dial out, no phones ring, loud 60Hz buzz when off hook on line, phone may ring at 60Hz, LC in protection state	0 Vac
AC Ring-Grd	Ring shorted to other equipment AC power supply lines or house wiring	No dial tone, can not dial out, no phones ring, loud 60Hz buzz when off hook on line, phone may ring at 60Hz, LC in protection state	0 Vac

Foreign Electromotive Test

Tests for the presence of an AC or DC voltage from tip to ground or ring to ground. Failure conditions are:

- Tip-ground or ring-ground AC voltage is greater than 10 V_{RMS}
- Tip-ground or ring-ground DC voltage is greater than 6 V

The failure types are shown below, with possible causes and symptoms:

Test Failure Data	Possible Cause	Possible Symptoms	Normal Test Limits
-DC Tip-Grd	Tip shorted to other equipment power supply lines	No dial tone, can not dial out, no phones ring, LC in protection state	0 Vdc
-DC Ring-Grd	Ring shorted to other equipment power supply lines	No dial tone, can not dial out, no phones ring, LC in protection state	0 Vdc
+DC Tip-Grd	Tip shorted to other equipment power supply lines	No dial tone, can not dial out, no phones ring, LC in protection state	0 Vdc
+DC Ring-Grd	Ring shorted to other equipment power supply lines	No dial tone, can not dial out, no phones ring, LC in protection state	0 Vdc
AC Tip-Grd	Tip shorted to other equipment AC power supply lines or house wiring	No dial tone, can not dial out, no phones ring, loud 60Hz buzz when off hook on phone, phone may ring at 60Hz, LC in protection state	0 Vac
AC Ring-Grd	Tip shorted to other equipment AC power supply lines or house wiring	No dial tone, can not dial out, no phones ring, loud 60Hz buzz when off hook on phone, phone may ring at 60Hz, LC in protection state	0 Vac

Resistive Faults Test

Tests the tip to ring, tip to ground, and ring to ground DC resistance.

Failure condition: tip-ring, tip-ground or ring-ground DC resistance is less than 150 kΩ. This test discriminates between a resistive fault in the loop and a receiver off-hook condition from properly working terminal equipment.

The failure types are shown below, with possible causes and symptoms:

Test Failure Data	Possible Cause	Possible Symptoms	Normal Test Limits
Tip-Grd	Tip shorted to cable shield, moisture in cable, house line protection device shorted Tip to Grd.	No dial tone, 60Hz buzz on line due to longitudinal imbalance, no incoming calls	150kΩ or greater
Ring-Grd	Ring shorted to cable shield, moisture in cable, house line protection device shorted Ring to Grd.	No dial tone, 60Hz buzz on line due to longitudinal imbalance, no incoming calls	150kΩ or greater
Tip-Ring	Staple or nail shorting Tip and Ring, moisture in cable, house line protection device shorted Tip to Ring	No dial tone, no incoming calls	150kΩ or greater

Receiver Off Hook Test

Failure condition: detection of a non-linear tip-ring DC resistance.

The failure types are shown below, with possible causes and symptoms:

Test Failure Data	Possible Cause	Possible Symptoms	Normal Test Limits
Off Hook	One of multiple phones left off hook	Can not receive calls, can not get dial tone on other phones in house	On Hook

Ringers Test

Measures the terminal equipment's AC ringer impedance at the frequency of the selected template.

Failure condition: the measured equivalent ringer count across tip-ring is less than 0.175 REN (no phones connected, low REN phone, or broken wire) or greater than 5 REN (too many phones connected).

Note 1: 1 REN = 7000Ω at 20Hz in North America

Note 2: Some CPE devices with electronic ringers present less than 0.175 REN and thus fail the Ringers test.

The failure types are shown below, with possible causes and symptoms:

Test Failure Data	Possible Cause	Possible Symptoms	Normal Test Limits
Less Than 0.175 REN	Ringer is open, no phone connected, open in copper line, phone may actually have REN value less than 0.175	Phone does not ring	Between 0.175 and 5.0 REN
More Than 5.0 REN	Total REN in multiple phones exceeds 5 REN, ringer capacitor shorted	Line ring trips after short ring burst, some phones may ring and others do not, low ring loudness on some phones with old bell type ringers.	Between 0.175 and 5.0 REN

Limitations

CAUTION
Service affecting
Performing loop diagnostics requires the line under test to be first taken out of service. This must be done manually when using SNMP. The web-based interface provide the option of forcing a line out of service. Note that forcing a line out of service while a call is in progress, drops a call.

Perform loop diagnostics only during installation or a scheduled maintenance window, unless the subscriber is unable to use the line.

Loop Diagnostics and Line Card Diagnostics always run together.

From any interface, you can start loop diagnostics on a second line while diagnostics are running on another line. However, you cannot start another loop diagnostics run on a line already running loop diagnostics until the tests have completed.

Test Results

You can examine the test results by walking the **arrisMtaDevDiagLoopTable** entry that corresponds to the line under test. Tests display the date and time of the last test run, and the following possible results for each test:

Result	Meaning	Action
Not Started	No tests have been run since the modem was last started.	Start (or re-start) the diagnostics.
In Progress	Diagnostics are still running	Wait 10–20 seconds and try again.
Invalid State	The line must be out of service (OOS) before starting loop diagnostics.	Take the line out of service and retry the diagnostics.
Aborted	The test could not be completed, possibly because a higher order test failed.	Check the other tests for failures. Correct any problems and re-try the diagnostics.
Fail	A test has failed.	The results show the values returned by the failing test. Correct any problems and re-try the diagnostics.
Pass	A test has passed.	None.
Unsupported	Loop diagnostics are not supported on single-line Telephony Modems.	None.

The **arrisMtaDevDiagLoopLastResult** MIB object provides a summary result:

`diagnostics-passed`(1),
`hazardous-potential-test-failure`(2),
`foreign-emf-test-failure`(3),
`resistive-faults-test-failure`(4),
`receiver-offhook-test-failure`(5),
`ringer-test-failure`(6),
`invalid-state-to-init-diags`(7),
`line-is-unprovisioned`(8),
`diagnostics-results-pending`(9),
`not-started`(10), or
`unsupported`(11).

If more than one a test failed, only the first failing test is shown.

The Telephony Modem stores the results of the last loop diagnostics run for each line since the modem was started or rebooted. If the **arrisMtaDevDiagLoopTime** object is set to midnight on January 1, 1970, no loop diagnostics have been run on that line since the modem was started or rebooted.

Action

Perform the following tasks as needed.

Running Loop Diagnostics from an SNMP Manager

Follow these steps to run loop diagnostics from an SNMP network management system.

1. Locate the MTA and line you wish to test.
2. Use the following table to find the interface index number for the line:

Line	Interface
1	9
2	10

3. Take the desired line out of service by setting the associated **ifAdminStatus** MIB object to **down**(2). This object is indexed by the interface number; use the interface number for the line found in step 2. For example, **ifAdminStatus**.9 corresponds to line 1.
4. Set the **arrisMtaDevDiagLoopRequest** object for this line to `true`(1). This object is indexed by the line number; that is, **arrisMtaDevDiagLoopRequest**.1 corresponds to line 1.

   The diagnostics begin running on the line. The tests require about 10 to 20 seconds to complete.
5. After 10 to 20 seconds, retrieve the diagnostics results from the **arrisMtaDevDiagLoopTable**.
6. See Test Results above to interpret the results.

Running Loop Diagnostics from the Web-based Interface

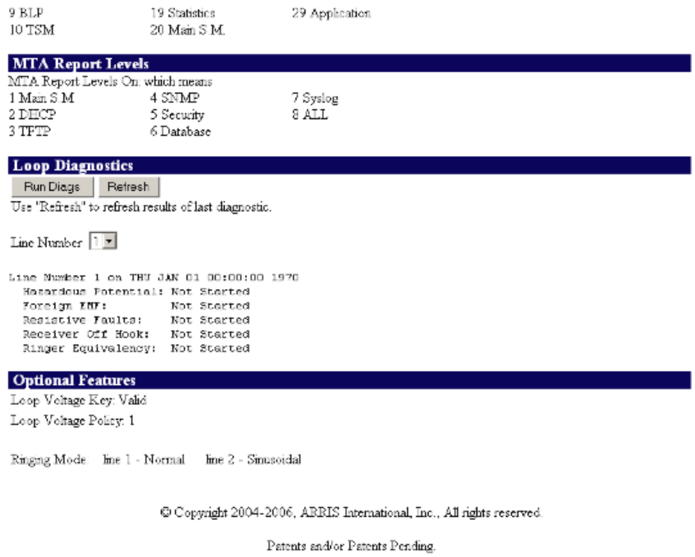
Follow these steps to run loop diagnostics from the web-based troubleshooting interface.

CAUTION
Service affecting
Performing loop diagnostics from the web-based interface give you the option of taking the line under test out of service after confirmation.

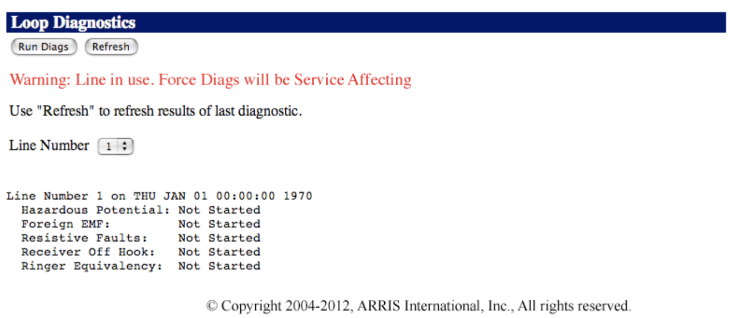
1. Access the Advanced web pages as described in see "*Accessing the Advanced Pages* (page 305).

   When accessing the Advanced pages for the first time, the Product Details page appears. If you are at another Advanced page, select the Product link to move to the Product

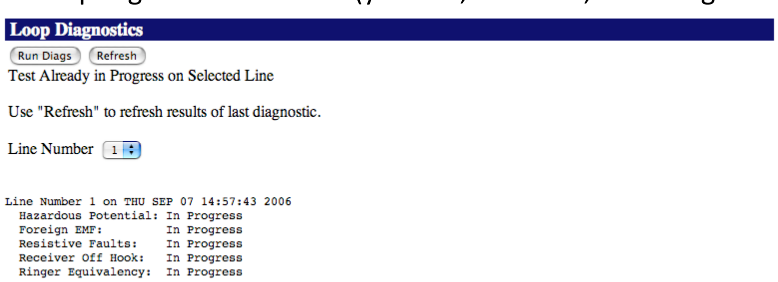
Details page. Note that the Loop Diagnostics is near the bottom of the Product Details page at first.



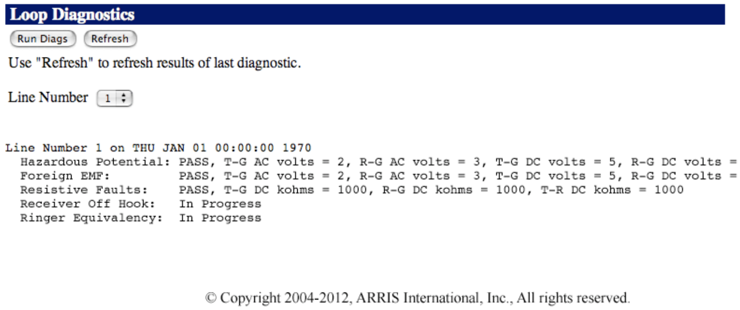
2. Select the line to test using the **Line Number** menu.
3. Click the **Run Diags** button to start the test.
4. If the chosen line is in service, the eMTA displays the following warning (note that the Loop Diagnostics section has moved to the top of the page). Click the **Force Diags** button to continue.



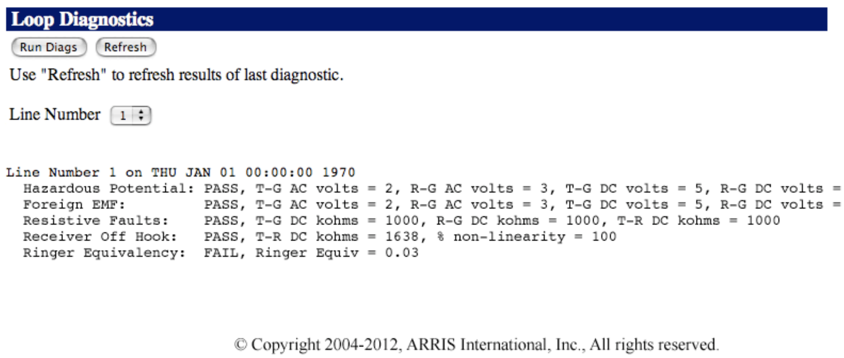
If the chosen line is already running loop diagnostics, the eMTA displays the following warning. Wait for 10 to 20 seconds for the current diagnostics run to complete before attempting to start another (you can, however, start diagnostics on another line).



5. To view the progress of the test, click the **Refresh** button.



6. The test typically takes about 10 to 20 seconds to complete. After 10 to 20 seconds, click the **Refresh** button to show the results. When the test completes, all five tests are marked either PASS or FAIL as shown in the example below.



Loopback Testing

Touchstone NCS firmware supports both NETWLOOP and NETWTEST loopback connection modes. You must activate loopback mode on a line from the call server.

Reset to Factory Defaults

Touchstone firmware provides the ability to reset a Telephony Modem to its factory-default configuration. Use one of the following methods to reset a Telephony Modem:

- Press and hold the **Reset** button on the back of the unit for 15 seconds. This resets the E-UE and (for Telephony Gateways) the router to factory defaults.
- Using an SNMP manager, set the **rdkbRgDeviceFactoryReset** object to **1**. This methods does not affect router settings.

Resetting the Router

All Touchstone Telephony Gateways and Data Gateways provide an additional method of resetting the router to factory defaults without resetting the E-UE. To reset only the router, use the “Restart Router” web page under the Utilities pages.

Using the Password of the Day Tool

This procedure describes the purpose and usage of the ARRIS Password of the Day (PWoD) tool.

Touchstone firmware provides a web-based troubleshooting interface consisting of two parts:

- Standard pages, available (by default) to both subscribers and operators.
- Advanced pages, available only by entering a password. These pages may contain sensitive information. The password changes daily for further protection.

The PWoD tool generates the appropriate password to access the advanced troubleshooting pages.

Note: the PWoD functionality is disabled until a seed is configured, as described in "see "*Changing the Seed* (page 288)." When setting up SNMP security, avoid using the default community strings ("Public" and "Private") or open access (no community string).

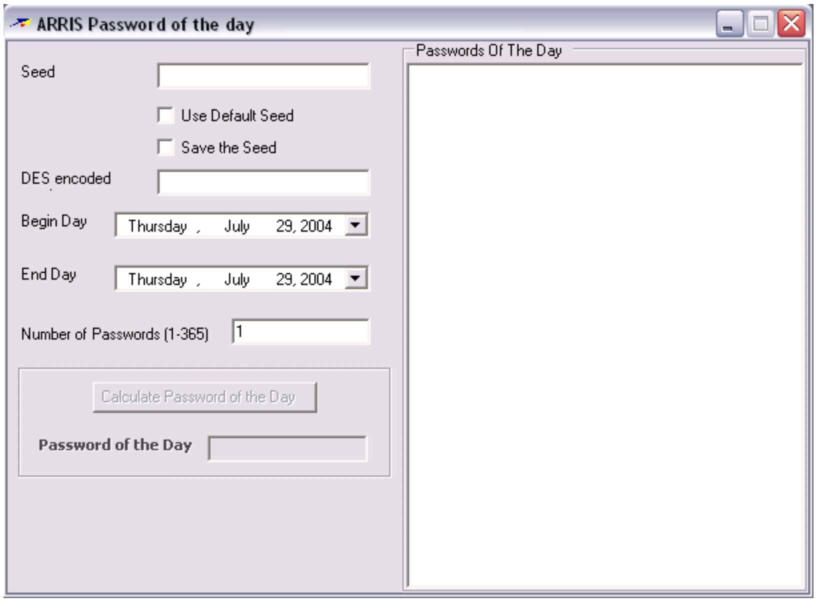
About the Password of the Day Tool

The PWoD tool, **ARRISpwod.exe**, is available through the Firmware Download Center. It is a Windows application, with the following requirements:

- Windows 98, Windows NT, Windows 2000, Windows XP or Windows 7
- Microsoft.Net plus Service Pack 2. The installer provides this package if required.
- Internet Explorer version 5.1 or newer. The newest version of Internet Explorer is available at *the Microsoft web site* (*http://www.microsoft.com/windows/ie/default.asp*).

The PWoD tool can create a single password, or a list of passwords for a range of days (up to 365). For added security, you can also define a seed value used to generate the passwords.

Before performing any of the tasks in this procedure, double-click the file to start the PWoD tool. The following diagram shows the tool.



Action

Perform the following tasks as needed.

Changing the Seed

Follow these steps to change the seed.

1. Start the PWoD tool, if you have not done so already.
2. Enter a new seed value (4 to 8 ASCII characters) in the **Seed** field at the top of the PWoD window. Make sure the **Use Default Seed** box is not checked.

   The encoded seed appears in the **DES encoded** field.

   Example seed values:

   ```
   abCd1234
   Abc#$^&
   *!() 1_?
   A1h53
   abcdEFGH
   ```
3. If you want to save the seed, make sure the **Save Seed** box is checked.

   The PWoD tool saves the new seed to a file called **password.dat**, in the directory where the PWoD is located. Make sure that any computer that can access a password file is reasonably secure.

   Touchstone E-UEs also need to have the changed seed so that their internal PWoD generators remain in sync with the external tool. Write the DES encoded seed to the **arrisCmDoc30AccessClientSeed** object in the eDVA configuration file.

Note: To disable Password of the Day functionality, set the **arrisCmDoc30AccessClientSeed** object to all zeroes.

Generating a Single Password

Follow these steps to generate a single password.

1. Start the PWoD tool, if you have not done so already.
2. Make sure the **Save Seed** box is checked, and the **Use Default Seed** checkbox is not checked.
3. Set the **Begin Day** and **End Day** dates to the same date (the default for both fields is the current date).
4. Click **Calculate Password of the Day**.

   The Password of the Day appears in the text box at the right of the PWoD tool window. You can select and copy the password as needed.

Generating a List of Passwords

Follow these steps to generate a list of passwords.

1. Start the PWoD tool, if you have not done so already.
2. Make sure the **Save Seed** box is checked, and the **Use Default Seed** checkbox is not checked.
3. Set the **Begin Day** and **End Day** dates to the range of days that you want to generate passwords for (the default for both fields is the current date).
4. Use **Browse** to specify a file name and location for the output file.
5. Click **Calculate Password of the Day**.

   The Password of the Day for the first day appears in the text box at the bottom of the PWoD tool window.

The PWoD tool writes the list of passwords to the specified file. The file contains a list of dates and the associated password for that day.

Using the Web-based Troubleshooting Interface

The firmware provides a web-based interface to a status monitoring and troubleshooting subsystem. Status information is available to anyone by using a standard web browser to access the cable modem IP address. A password-protected set of pages provides views of advanced network settings.

The following sections describe the screens available from the web interface. To access the screens, see “see "*Accessing the Standard Pages* (page 304)” below.

Access Options

You can access the troubleshooting screens through either the Touchstone Telephony product RF or Ethernet interfaces.

Requirements

You need the following equipment to access the troubleshooting pages:

- computer with an Ethernet interface and a web browser
- Ethernet cable (if using the E-UE Ethernet interface)
- (advanced pages only) the password of the day

Controlling Access to the Interface

The following MIB objects control access to the basic and advanced troubleshooting pages.

arrisCmDoc30AccessHttpLan

Values: **disable**(0), **enable**(1)

Determines which pages are available from the Ethernet port of the E-UE. The value is stored in non-volatile memory.

arrisCmDoc30AccessHttpWan

Values: **disable**(0), **enable**(1), **CMpage**(3)

Determines which pages are available from the network attached to the CMTS. The value is stored in non-volatile memory.

arrisCmDoc30AccessHttpPwCtrl

Values:

none(0): No password is required for any screen;
advanced(1): The PWoD is required for Advanced screens;
all(2): The PWoD is required for all screens.

Sets which groups of troubleshooting pages are password protected. The default is **advanced**(1).

arrisCmDoc30AccessClientSeed

Changing the seed changes the results of the Password of the Day (PWoD). See see "*Using the Password of the Day Tool* (page 287) for details about the PWoD. Store the seed value in the CM configuration file.

arrisCmDoc30AccessHttpTimeout

The time, in minutes, that the advanced pages are available before the Telephony Modem requires password re-entry. Valid range: **1** to **1440** minutes, or **0** to disable the timeout.

Standard Screens

Anyone can access the screens described in this section.

Status Screen

The status screen is the index page, and can also be selected by choosing the **Status** link at any standard screen. The following is an example:

ARRIS

Status

Status | HW/FW Versions | Event Log | CM State | Advanced

RF Parameters

Downstream QAM

Downstream #	DCID	Freq	Power	SNR	Modulation	Octets	Correcteds	Uncorrectables
1	201	303.00 MHz	-3.90 dBmV	43.38 dB	256QAM	1132	3336	12452
2	202	309.00 MHz	-4.00 dBmV	43.38 dB	256QAM	1132	3129	7258
3	203	315.00 MHz	-3.90 dBmV	43.38 dB	256QAM	1132	2925	10064
4	204	321.00 MHz	-3.80 dBmV	43.38 dB	256QAM	1132	2756	8478
5	205	327.00 MHz	-4.00 dBmV	43.38 dB	256QAM	1132	3732	10070
6	206	333.00 MHz	-4.10 dBmV	43.38 dB	256QAM	1132	3876	9503
7	207	339.00 MHz	-4.10 dBmV	43.38 dB	256QAM	1132	2996	8724
8	208	345.00 MHz	-4.20 dBmV	43.38 dB	256QAM	1132	2995	9154
9	209	351.00 MHz	-4.20 dBmV	43.38 dB	256QAM	1132	2701	9085
10	210	357.00 MHz	-4.10 dBmV	43.38 dB	256QAM	1132	2440	6070
11	211	363.00 MHz	-4.00 dBmV	43.38 dB	256QAM	1132	2589	8706
12	212	369.00 MHz	-3.90 dBmV	43.38 dB	256QAM	1132	2520	8607
13	213	375.00 MHz	-3.80 dBmV	43.38 dB	256QAM	1132	2783	10912
14	214	381.00 MHz	-3.70 dBmV	43.38 dB	256QAM	1132	2360	8577
15	215	387.00 MHz	-3.50 dBmV	43.38 dB	256QAM	1125	2290	7016
16	216	393.00 MHz	-3.40 dBmV	43.38 dB	256QAM	1132	2551	9995
17	218	405.00 MHz	-3.20 dBmV	43.38 dB	256QAM	1132	1799	4171
18	219	411.00 MHz	-3.40 dBmV	43.38 dB	256QAM	1132	2279	7170
19	220	417.00 MHz	-3.40 dBmV	43.38 dB	256QAM	1132	2231	7140
20	222	429.00 MHz	-3.70 dBmV	43.38 dB	256QAM	1132	2249	4935
21	223	435.00 MHz	-4.00 dBmV	44.63 dB	256QAM	1132	1961	7254
22	224	441.00 MHz	-4.10 dBmV	43.38 dB	256QAM	1132	2311	8646
23	225	447.00 MHz	-4.10 dBmV	43.38 dB	256QAM	1132	2386	10913
24	226	453.00 MHz	-4.10 dBmV	43.38 dB	256QAM	1132	2184	9084
25	227	459.00 MHz	-4.30 dBmV	43.38 dB	256QAM	1132	2075	9066
26	228	465.00 MHz	-4.40 dBmV	43.38 dB	256QAM	1796	1845	3656
27	229	471.00 MHz	-4.30 dBmV	43.38 dB	256QAM	1132	1881	6007
28	230	477.00 MHz	-4.40 dBmV	43.38 dB	256QAM	1132	1372	4229
29	231	483.00 MHz	-4.60 dBmV	43.38 dB	256QAM	37026	1650	4629
30	232	489.00 MHz	-4.70 dBmV	43.38 dB	256QAM	0	1567	6249

Reset FEC Counters

Downstream OFDM

Downstream #	FFT Type	Channel Width(MHz)	# of Active Subcarriers	First Active Subcarrier(MHz)	Last Active Subcarrier(MHz)	Average RxMER(dB)		
						Pilot	PLC	Data
0	4K	94	1880	501	594	45	43	45
1	4K	94	1880	601	694	42	41	42

Upstream QAM

Upstream #	UCID	Freq	Power	Channel Type	Symbol Rate	Modulation
1	28	53.00 MHz	31.25 dBmV	DOCSIS2.0 (ATDMA)	5120 kSym/s	64QAM
2	27	46.60 MHz	30.75 dBmV	DOCSIS2.0 (ATDMA)	5120 kSym/s	64QAM
3	26	40.20 MHz	30.75 dBmV	DOCSIS2.0 (ATDMA)	5120 kSym/s	64QAM
4	25	33.80 MHz	30.00 dBmV	DOCSIS2.0 (ATDMA)	5120 kSym/s	64QAM

Upstream OFDM

Upstream #	FFT Type	Channel Width(MHz)	# of Active Subcarriers	First Active Subcarrier(MHz)	Last Active Subcarrier(MHz)	Tx Power(dBmV)

Status

System Uptime:	12 d: 18 h: 20 m
Computers Detected:	staticCPE(1), dynamicCPE(0)
CM Status:	Telephony-Reg Complete
Time and Date:	Wed 2017-04-12 09:47:42

Interface Parameters

Interface Name	Provisioned	State	Speed (Mbps)	MAC address
LAN Port 1	Enabled	Up		5C:B0:66:BC:66:6D
LAN Port 2	Enabled	Up		5C:B0:66:BC:66:6D
LAN Port 3	Enabled	Up		5C:B0:66:BC:66:6D
LAN Port 4	Enabled	Up		5C:B0:66:BC:66:6D
CABLE	Enabled	Up	-----	5C:B0:66:BC:66:6E
MTA	PassWithWarnings	Up	-----	5C:B0:66:BC:66:6F

Diplexer

Band#	Upstream Range	Downstream Range	Current Band Setting
0	5 MHz-85 MHz	108 MHz-1002 MHz	
1	5 MHz-42 MHz	108 MHz-1002 MHz	X

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The RF Parameters section of this page provides downstream and upstream settings.

The Status section provides:

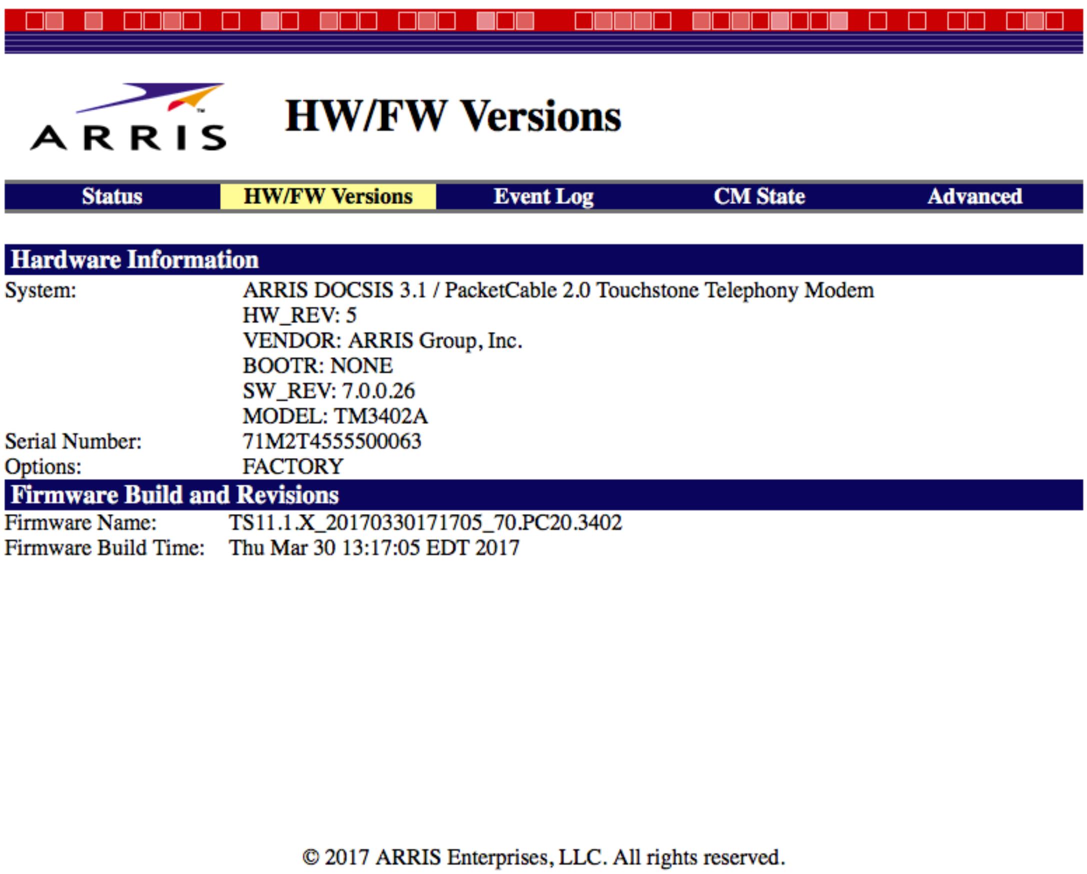
- System uptime
- Number of computers on the LAN that the modem detected
- Cable modem status

The Interface Parameters section provides the names, MAC addresses, and status of each interface on the E-UE. For subscriber-side data interfaces (Ethernet and WiFi), the page also reports interface speed in Mb/sec.

The Diplexer section shows the diplexers available on this device, and which one is in use.

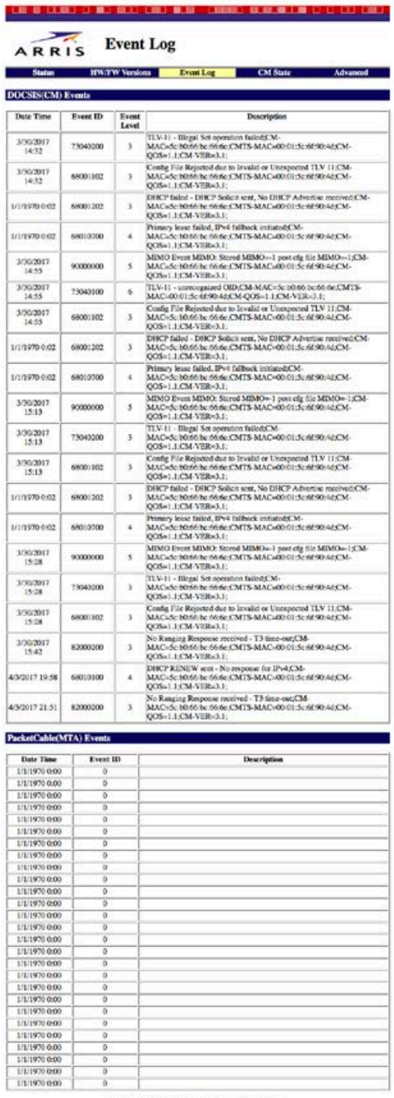
HW/FW Versions Screen

The Hardware/Firmware Versions screen provides information about the hardware and firmware revision levels. To display this screen, choose the **HW/FW Versions** link.



Event Log Screen

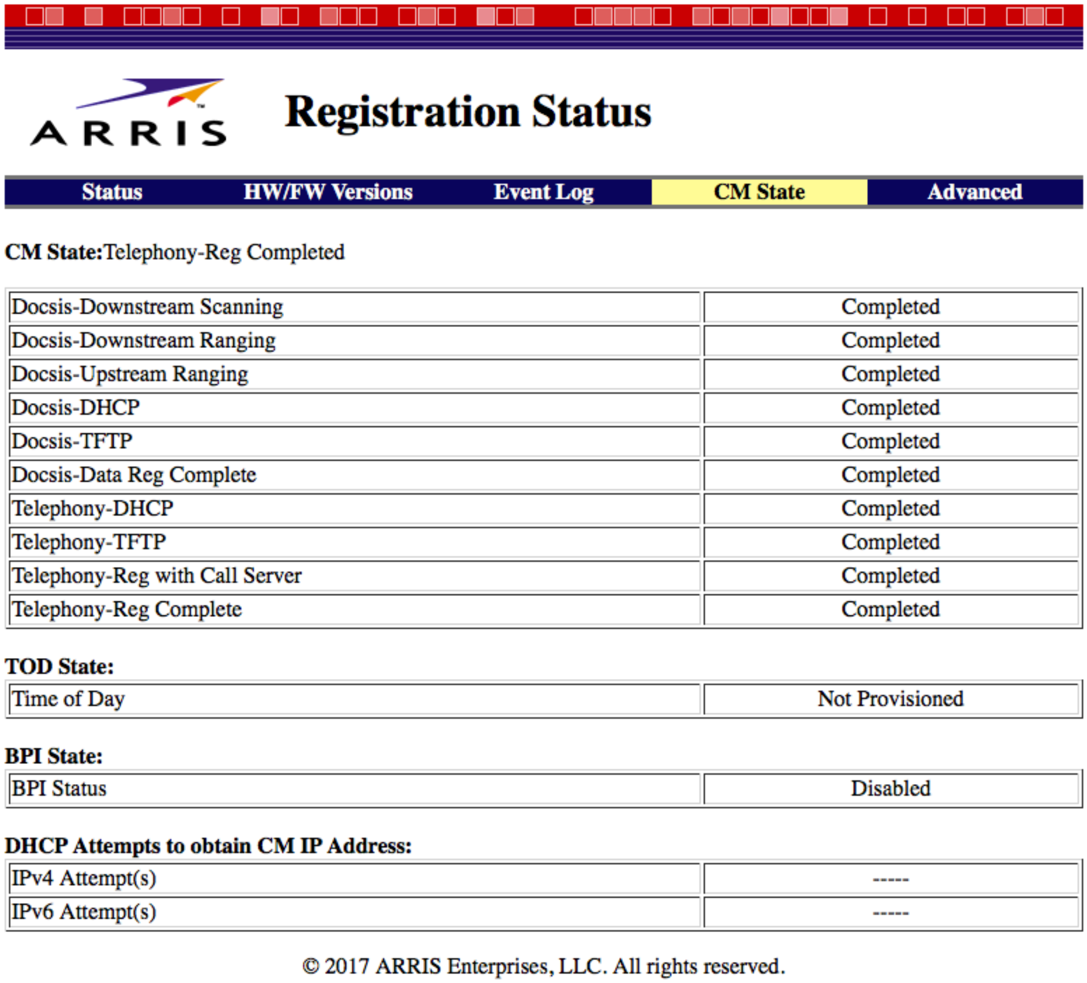
The Event Log screen displays a table of recent events. The E-UE stores the event log in non-volatile memory. To display this screen, choose the **Event Log** link.



Note: Setting the **arrisMtaDevEventHideMacFQDNandIPAddress** MIB object to `enable`(0) in the CM configuration file prevents display of the device FQDN and IP address on the basic Event Log screen. Users with access to the Advanced pages (using the Password of the Day) can always see FQDN and IP address information on the Advanced Event Log screen.

Registration Status Screen

The Registration Status screen shows the results of the E-UE registration process and current battery telemetry state and serial number. To display this screen, choose the **CM State** link.



Registration Status

Status | HW/FW Versions | Event Log | CM State | Advanced

CM State: Telephony-Reg Completed

Docsis-Downstream Scanning	Completed
Docsis-Downstream Ranging	Completed
Docsis-Upstream Ranging	Completed
Docsis-DHCP	Completed
Docsis-TFTP	Completed
Docsis-Data Reg Complete	Completed
Telephony-DHCP	Completed
Telephony-TFTP	Completed
Telephony-Reg with Call Server	Completed
Telephony-Reg Complete	Completed

TOD State:

Time of Day	Not Provisioned

BPI State:

BPI Status	Disabled

DHCP Attempts to obtain CM IP Address:

IPv4 Attempt(s)	-----
IPv6 Attempt(s)	-----

The following states may appear under the **CM State** listing.

Hardware Diags
: Status of hardware diagnostics. Displayed only if the diagnostics find a problem.

Docsis-Downstream Scanning
: Status of DOCSIS downstream scanning.

Docsis-Downstream Ranging
: Status of DOCSIS downstream ranging.

Docsis-Upstream Ranging
: Status of DOCSIS upstream ranging.

Docsis-DHCP
: Status of the CM DHCP phase. The Registration Status page also displays the number of attempts required to obtain a CM IP address.

Docsis-TFTP
Status of the CM configuration file download.

Docsis-Data Reg Complete
Shows whether the modem has completed DOCSIS data registration.

Battery Mismatch
Indicates that the wrong battery has been inserted in the Telephony Modem (see below).

The last two sections of this page provide information about the charging system, as described below.

Power Supply Telemetry Field

The Power Supply Telemetry field provides three fields:

Battery Operation Status
The operational status of the charging system. See

The Battery Operation Status is “No Status Reported” if the battery charger has been disabled. Check the Event Log page for a “Battery Charger Disabled” alarm to verify. Power-cycle or reset the E-UE to re-enable the charger.

Battery Serial Number
The battery serial number is available on most Telephony Modems. The following is a list of other messages that may appear in the Battery Serial Number field:

Battery Serial Number Not Supported
The Telephony Modem is an early Model 5 Telephony Modem, or another non-supported modem or battery pack.

Battery Missing
No battery is installed.

AC Failed. S/N Temporarily Unavailable
The Telephony Modem is running on battery power, and has disabled reading the battery serial number as part of the data services shutdown function to conserve power. Not all Telephony Modems disable reading the battery serial number upon data shutdown (for example, Model 7 Telephony Modems).

Unable to obtain Battery Serial Number(s)
Appears in cases where the serial number is missing or unreadable.

Battery Part Number
The ordering code of the installed battery. See Appendix A for details.

If the Battery Part Number is "Contact ARRIS," the battery is either defective or the Telephony Modem has an unsupported battery installed (for example, a model BPB088S battery installed in a TM502).

Battery Test State Field

The Battery Test State field indicates whether the charging system is testing the battery; and if so, the current progress of the test. This field provides one of the following indications:

Not Currently Under Battery Test
: The normal operating state.

Under Test – Initial Charge
: The battery is being charged to full capacity to begin the test.

Under Test – Initial Discharge
: The battery was fully charged, and the E-UE is now running on battery power.

Under Test – Final Charge
: The battery was fully charged, then discharged, and now is being charged to its operational capacity.

See Scheduling Tests for information about configuring battery tests.

Advanced Screens

The following screens require a Password of the Day (see see "*Using the Password of the Day Tool* (page 287) for details). Use the **Basic** link on any screen to return to the standard status pages.

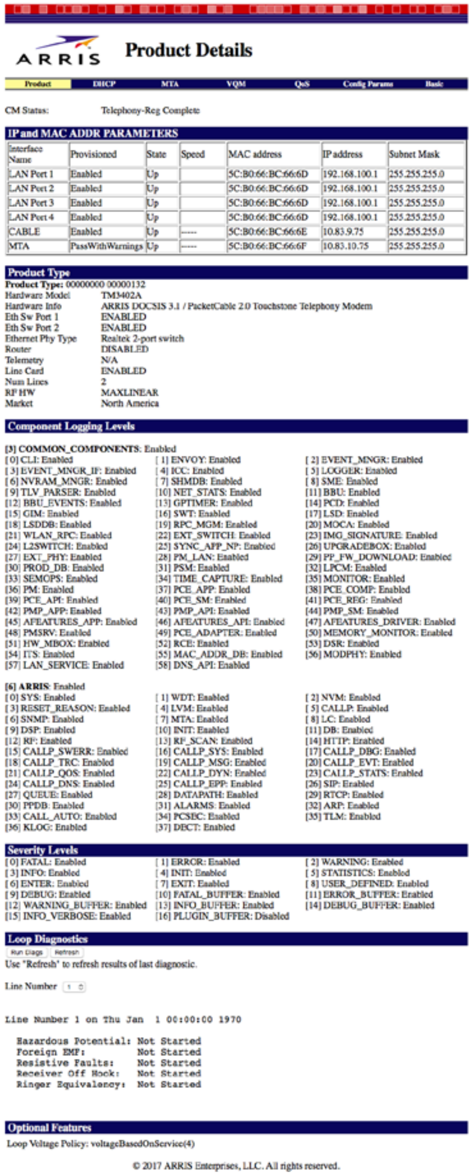
Note: Advanced screens contain potentially sensitive information about the network.

Product Details Screen

The Detailed Status screen provides the following information:

- E-UE status
- IP (IPv4 or IPv6, as appropriate) and MAC address parameters
- system information, including enabled features

To access this screen, choose the **Product** link from any advanced screen.



ARRIS **Product Details**

Product | DHCP | MTA | VQM | QoS | Config Params | Basic

CM Status: Telephony-Reg Complete

IP and MAC ADDR PARAMETERS

Interface Name	Provisioned	State	Speed	MAC address	IP address	Subnet Mask
LAN Port 1	Enabled	Up		5C:B0:66:BC:66:6D	192.168.100.1	255.255.255.0
LAN Port 2	Enabled	Up		5C:B0:66:BC:66:6D	192.168.100.1	255.255.255.0
LAN Port 3	Enabled	Up		5C:B0:66:BC:66:6D	192.168.100.1	255.255.255.0
LAN Port 4	Enabled	Up		5C:B0:66:BC:66:6D	192.168.100.1	255.255.255.0
CABLE	Enabled	Up	-----	5C:B0:66:BC:66:6E	10.83.9.75	255.255.255.0
MTA	PassWithWarnings	Up	-----	5C:B0:66:BC:66:6F	10.83.10.75	255.255.255.0

Product Type

Product Type: 00000000 00000132

Hardware Model	TM3402A
Hardware Info	ARRIS DOCSIS 3.1 / PacketCable 2.0 Touchstone Telephony Modem
Eth Sw Port 1	ENABLED
Eth Sw Port 2	ENABLED
Ethernet Phy Type	Realtek 2-port switch
Router	DISABLED
Telemetry	N/A
Line Card	ENABLED
Num Lines	2
RF HW	MAXLINEAR
Market	North America

Component Logging Levels

[3] COMMON_COMPONENTS: Enabled

[0] CLI: Enabled	[1] ENVOY: Enabled	[2] EVENT_MNGR: Enabled
[3] EVENT_MNGR_IF: Enabled	[4] ICC: Enabled	[5] LOGGER: Enabled
[6] NVRAM_MNGR: Enabled	[7] SHMDB: Enabled	[8] SME: Enabled
[9] TLV_PARSER: Enabled	[10] NET_STATS: Enabled	[11] BBU: Enabled
[12] BBU_EVENTS: Enabled	[13] GPTIMER: Enabled	[14] PCD: Enabled
[15] GIM: Enabled	[16] SWT: Enabled	[17] LSD: Enabled
[18] LSDDB: Enabled	[19] RPC_MGM: Enabled	[20] MOCA: Enabled
[21] WLAN_RPC: Enabled	[22] EXT_SWITCH: Enabled	[23] IMG_SIGNATURE: Enabled
[24] L2SWITCH: Enabled	[25] SYNC_APP_NP: Enabled	[26] UPGRADEBOX: Enabled
[27] EXT_PHY: Enabled	[28] PM_LAN: Enabled	[29] PP_FW_DOWNLOAD: Enabled
[30] PROD_DB: Enabled	[31] PSM: Enabled	[32] LPCM: Enabled
[33] SEMOPS: Enabled	[34] TIME_CAPTURE: Enabled	[35] MONITOR: Enabled
[36] PM: Enabled	[37] PCE_APP: Enabled	[38] PCE_COMP: Enabled
[39] PCE_API: Enabled	[40] PCE_SM: Enabled	[41] PCE_REG: Enabled
[42] PMP_APP: Enabled	[43] PMP_API: Enabled	[44] PMP_SM: Enabled
[45] AFEATURES_APP: Enabled	[46] AFEATURES_API: Enabled	[47] AFEATURES_DRIVER: Enabled
[48] PMSRV: Enabled	[49] PCE_ADAPTER: Enabled	[50] MEMORY_MONITOR: Enabled
[51] HW_MBOX: Enabled	[52] RCE: Enabled	[53] DSR: Enabled
[54] ITS: Enabled	[55] MAC_ADDR_DB: Enabled	[56] MODPHY: Enabled
[57] LAN_SERVICE: Enabled	[58] DNS_API: Enabled	

[6] ARRIS: Enabled

[0] SYS: Enabled	[1] WDT: Enabled	[2] NVM: Enabled
[3] RESET_REASON: Enabled	[4] LVM: Enabled	[5] CALLP: Enabled
[6] SNMP: Enabled	[7] MTA: Enabled	[8] LC: Enabled
[9] DSP: Enabled	[10] INIT: Enabled	[11] DB: Enabled
[12] RF: Enabled	[13] RF_SCAN: Enabled	[14] HTTP: Enabled
[15] CALLP_SWERR: Enabled	[16] CALLP_SYS: Enabled	[17] CALLP_DBG: Enabled
[18] CALLP_TRC: Enabled	[19] CALLP_MSG: Enabled	[20] CALLP_EVT: Enabled
[21] CALLP_QOS: Enabled	[22] CALLP_DYN: Enabled	[23] CALLP_STATS: Enabled
[24] CALLP_DNS: Enabled	[25] CALLP_EPP: Enabled	[26] SIP: Enabled
[27] QUEUE: Enabled	[28] DATAPATH: Enabled	[29] RTCP: Enabled
[30] PPDB: Enabled	[31] ALARMS: Enabled	[32] ARP: Enabled
[33] CALL_AUTO: Enabled	[34] PCSEC: Enabled	[35] TLM: Enabled
[36] KLOG: Enabled	[37] DECT: Enabled	

Severity Levels

[0] FATAL: Enabled	[1] ERROR: Enabled	[2] WARNING: Enabled
[3] INFO: Enabled	[4] INIT: Enabled	[5] STATISTICS: Enabled
[6] ENTER: Enabled	[7] EXIT: Enabled	[8] USER_DEFINED: Enabled
[9] DEBUG: Enabled	[10] FATAL_BUFFER: Enabled	[11] ERROR_BUFFER: Enabled
[12] WARNING_BUFFER: Enabled	[13] INFO_BUFFER: Enabled	[14] DEBUG_BUFFER: Enabled
[15] INFO_VERBOSE: Enabled	[16] PLUGIN_BUFFER: Disabled	

Loop Diagnostics

Run Diags | Refresh

Use "Refresh" to refresh results of last diagnostic.

Line Number 1

```
Line Number 1 on Thu Jan  1 00:00:00 1970

  Hazardous Potential: Not Started
  Foreign EMF:         Not Started
  Resistive Faults:    Not Started
  Receiver Off Hook:   Not Started
  Ringer Equivalency:  Not Started
```

Optional Features

Loop Voltage Policy: voltageBasedOnService(4)

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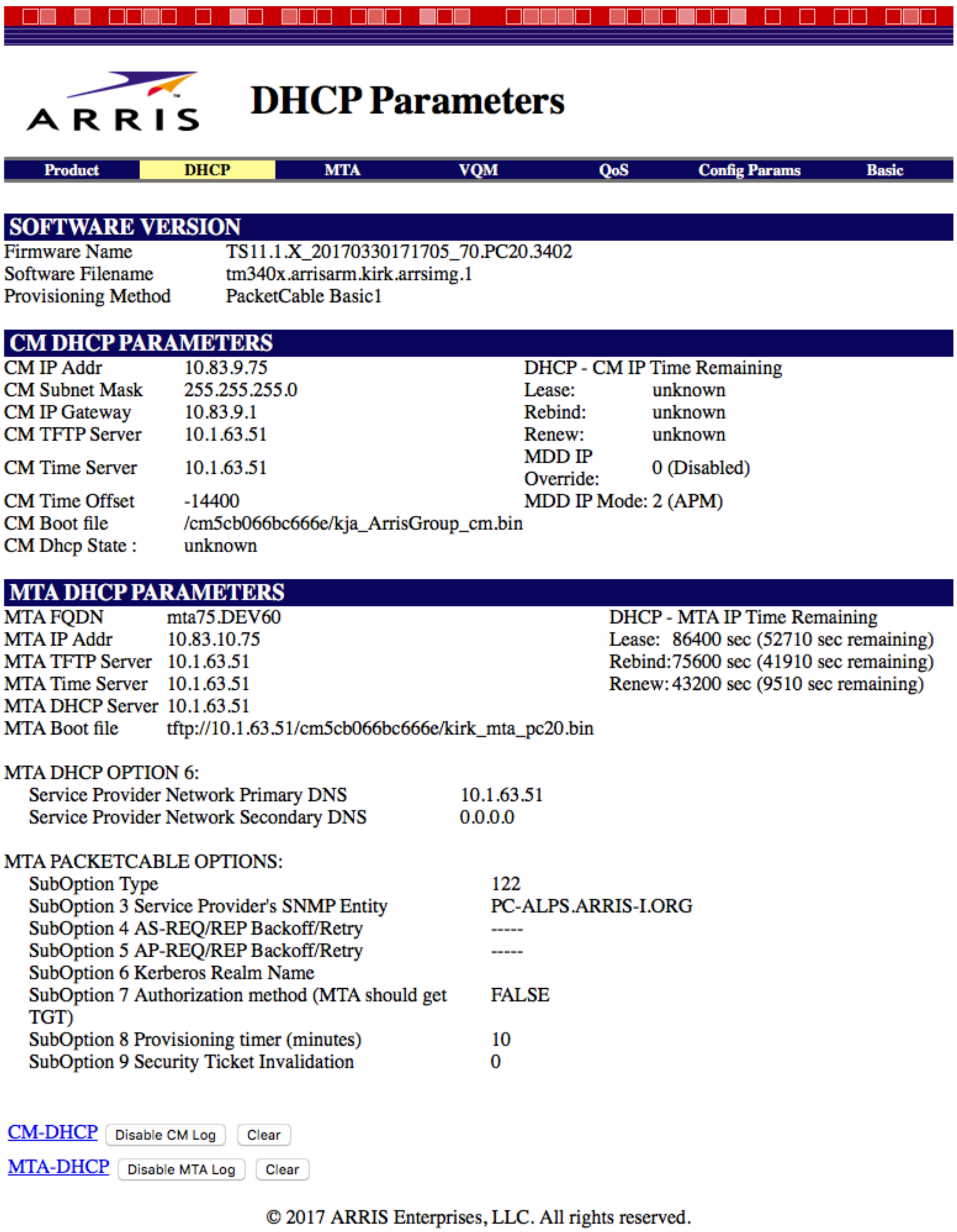
DHCP Parameters

The DHCP Parameters screen lists DHCP details for the cable modem and eDVA portions of the E-UE, including IPv4 or IPv6 addresses as appropriate. To access this screen, choose the **DHCP** link from any advanced screen.

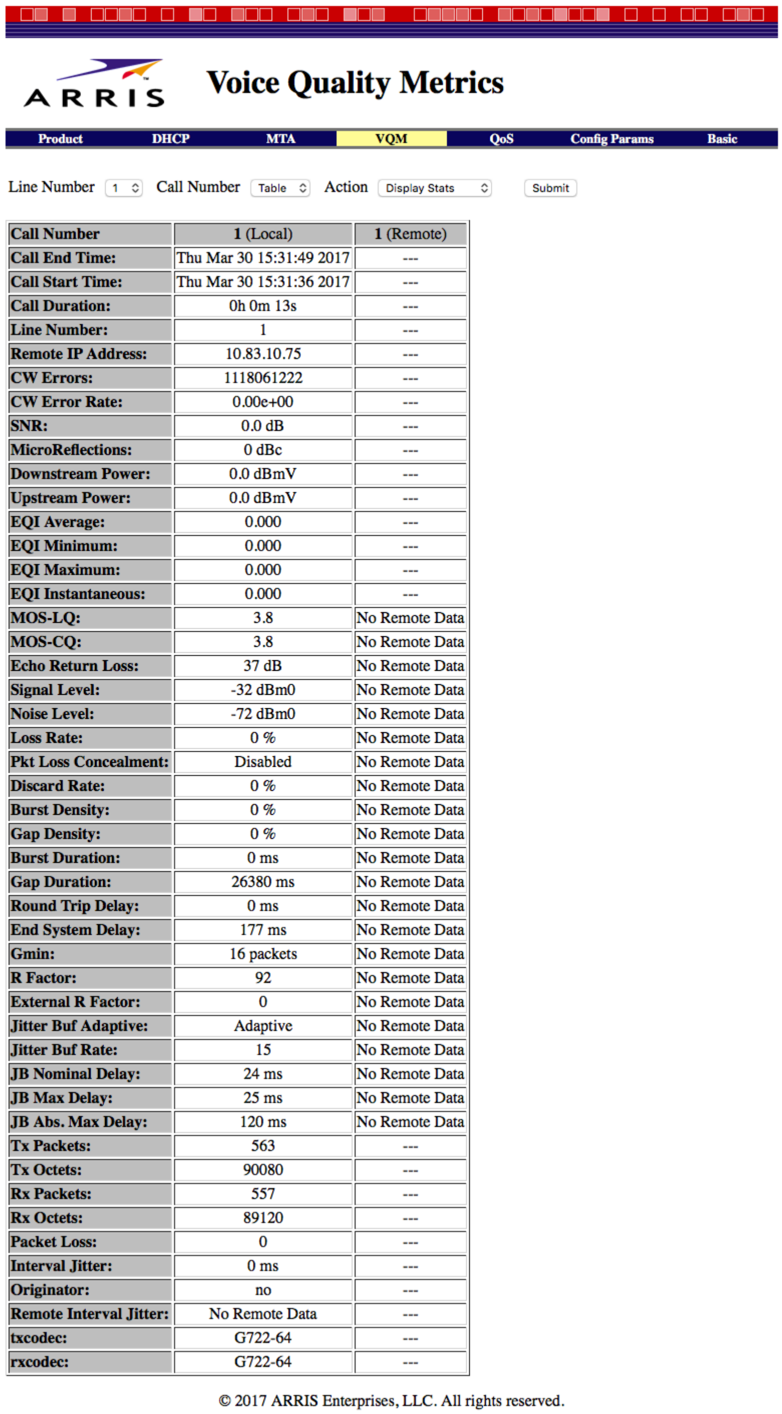


The DHCP Parameters screen provides links to CM DHCP and eDVA DHCP message trace logs.

Voice Quality Metrics Screen

The Voice Quality Metrics screen allows control and display of metrics for recent calls.

For more information, see see "*Managing Voice Quality Monitoring* (page 250).



Call Number	1 (Local)	1 (Remote)
Call End Time:	Thu Mar 30 15:31:49 2017	---
Call Start Time:	Thu Mar 30 15:31:36 2017	---
Call Duration:	0h 0m 13s	---
Line Number:	1	---
Remote IP Address:	10.83.10.75	---
CW Errors:	1118061222	---
CW Error Rate:	0.00e+00	---
SNR:	0.0 dB	---
MicroReflections:	0 dBc	---
Downstream Power:	0.0 dBmV	---
Upstream Power:	0.0 dBmV	---
EQI Average:	0.000	---
EQI Minimum:	0.000	---
EQI Maximum:	0.000	---
EQI Instantaneous:	0.000	---
MOS-LQ:	3.8	No Remote Data
MOS-CQ:	3.8	No Remote Data
Echo Return Loss:	37 dB	No Remote Data
Signal Level:	-32 dBm0	No Remote Data
Noise Level:	-72 dBm0	No Remote Data
Loss Rate:	0 %	No Remote Data
Pkt Loss Concealment:	Disabled	No Remote Data
Discard Rate:	0 %	No Remote Data
Burst Density:	0 %	No Remote Data
Gap Density:	0 %	No Remote Data
Burst Duration:	0 ms	No Remote Data
Gap Duration:	26380 ms	No Remote Data
Round Trip Delay:	0 ms	No Remote Data
End System Delay:	177 ms	No Remote Data
Gmin:	16 packets	No Remote Data
R Factor:	92	No Remote Data
External R Factor:	0	No Remote Data
Jitter Buf Adaptive:	Adaptive	No Remote Data
Jitter Buf Rate:	15	No Remote Data
JB Nominal Delay:	24 ms	No Remote Data
JB Max Delay:	25 ms	No Remote Data
JB Abs. Max Delay:	120 ms	No Remote Data
Tx Packets:	563	---
Tx Octets:	90080	---
Rx Packets:	557	---
Rx Octets:	89120	---
Packet Loss:	0	---
Interval Jitter:	0 ms	---
Originator:	no	---
Remote Interval Jitter:	No Remote Data	---
txcodec:	G722-64	---
rxcodec:	G722-64	---

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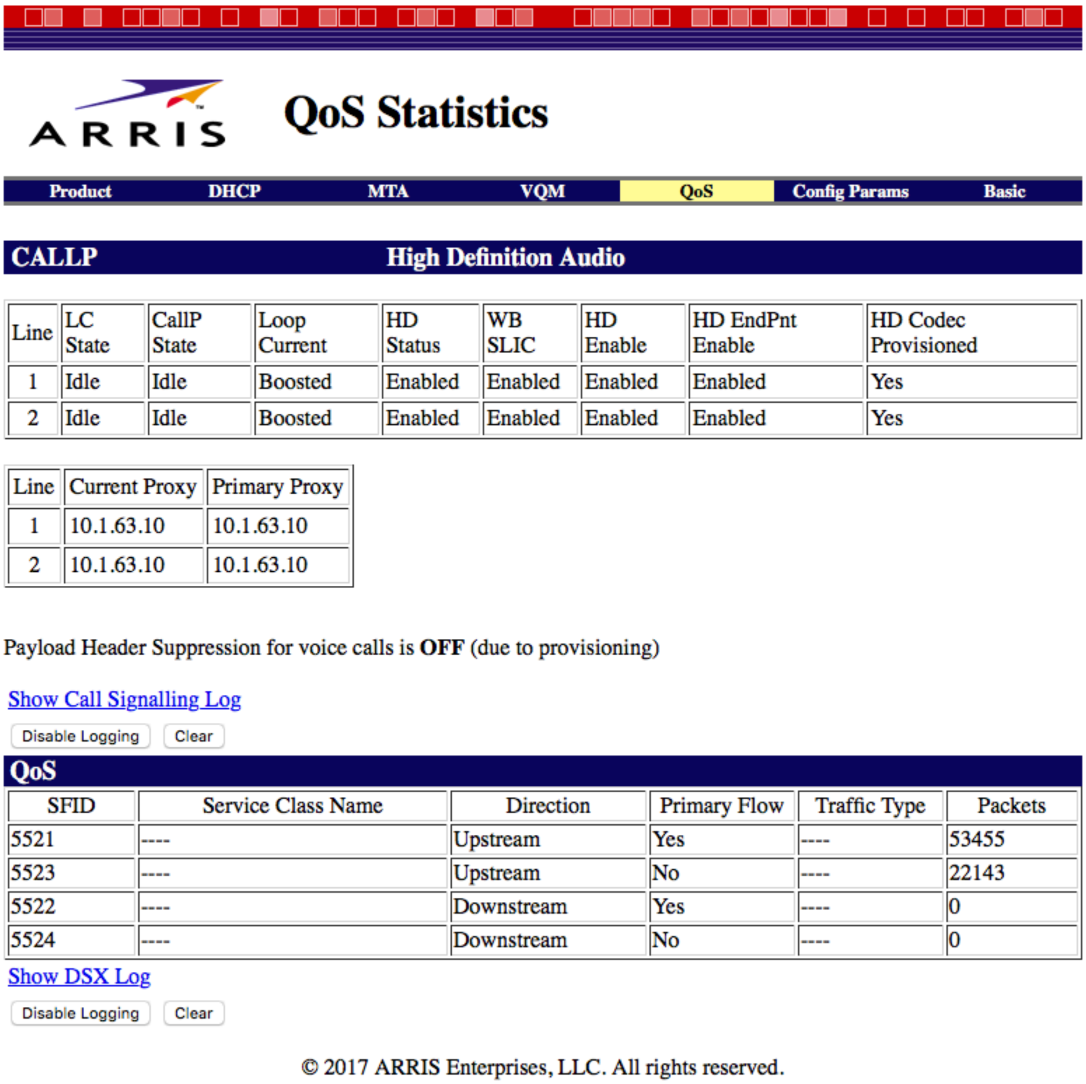
QoS Statistics Screen

The QoS Statistics screen shows the eDVA QoS statistics. To access this screen, choose the **QoS** link from any advanced screen.



Line	LC State	CallP State	Loop Current	HD Status	WB SLIC	HD Enable	HD EndPnt Enable	HD Codec Provisioned
1	Idle	Idle	Boosted	Enabled	Enabled	Enabled	Enabled	Yes
2	Idle	Idle	Boosted	Enabled	Enabled	Enabled	Enabled	Yes

Line	Current Proxy	Primary Proxy
1	10.1.63.10	10.1.63.10
2	10.1.63.10	10.1.63.10

Payload Header Suppression for voice calls is **OFF** (due to provisioning)

Show Call Signalling Log

Disable Logging Clear

QoS

SFID	Service Class Name	Direction	Primary Flow	Traffic Type	Packets
5521	----	Upstream	Yes	----	53455
5523	----	Upstream	No	----	22143
5522	----	Downstream	Yes	----	0
5524	----	Downstream	No	----	0

Show DSX Log

Disable Logging Clear

Note: Currently, the packet counts for the downstream service flows are not incremented and always show 0.

The following controls provide control and display of the CallP signalling log:

- The **Show Call Signalling Log** button displays the Call Signalling Log. The **Show DSX Log** button displays the DSX Log.
- The **Disable Logging** button disables logging for the associated signal type. When logging is disabled, the button name becomes **Enable Logging**.
- The **Clear** button clears the log for the associated signal type.

Call Signaling Log

The Call Signalling Log screen shows the call signaling log.

Call Signalling Log

Product | DHCP | MTA | VQM | QoS | Config Params | Basic

Call Signaling LOG 1
Wed, 12 Apr 2017 09:23:08 +0000
<-MSG- Message Sent
SIP/2.0 488 The event package is not supported
From: <sip:5555@arris-i.org:5060>;tag=45bbf0-a530a4b-13c4-50029-58edf1a9-41434472-58edf1a9
To: <sip:5555@arris-i.org:5060>;tag=45bdb8-a530a4b-13c4-50029-58edf1a9-1a750bc-58edf1a9
Call-ID: 45f550-a530a4b-13c4-50029-58edf1a9-14451a8c-58edf1a9
CSeq: 1 SUBSCRIBE
Via: SIP/2.0/UDP 10.1.63.10;branch=z9hG4bK9adf.6f005be.0
Via: SIP/2.0/UDP 10.83.10.75:5060;branch=z9hG4bK-58edf1a9-6177fce6-6c714713
Supported: timer,replaces,join,tdialog,100rel,histinfo,path
Content-Length: 0
Call Signaling LOG 2
Wed, 12 Apr 2017 09:23:08 +0000
-MSG-> Message Received
SIP/2.0 488 The event package is not supported
From: <sip:5555@arris-i.org:5060>;tag=45bbf0-a530a4b-13c4-50029-58edf1a9-41434472-58edf1a9
To: <sip:5555@arris-i.org:5060>;tag=45bdb8-a530a4b-13c4-50029-58edf1a9-1a750bc-58edf1a9
Call-ID: 45f550-a530a4b-13c4-50029-58edf1a9-14451a8c-58edf1a9
CSeq: 1 SUBSCRIBE
Via: SIP/2.0/UDP 10.83.10.75:5060;branch=z9hG4bK-58edf1a9-6177fce6-6c714713
Supported: timer,replaces,join,tdialog,100rel,histinfo,path
Content-Length: 0
Call Signaling LOG 3
Wed, 12 Apr 2017 09:23:42 +0000
<-MSG- Message Sent
REGISTER sip:arris-i.org SIP/2.0
From: TRAFFIC5556 Line2<sip:5556@arris-i.org:5060>;tag=45b308-a530a4b-13c4-50029-58dd2427-34e5bf30-58dd2427
To: TRAFFIC5556 Line2<sip:5556@arris-i.org:5060>
Call-ID: 462c18-a530a4b-13c4-50029-58dd2427-21f4c56a-58dd2427
CSeq: 7341 REGISTER
Via: SIP/2.0/UDP 10.83.10.75:5060;branch=z9hG4bK-58edf1cb-61788274-2bc76e1a
Max-Forwards: 70
Supported: timer,replaces,join,tdialog,100rel,histinfo,path
User-Agent: ARRIS-TM3402A release v.7.0.0.26 SN/5CB066BC666F
Contact: TRAFFIC5556 Line2<sip:5556@10.83.10.75:5060>;expires=300
Allow: INVITE,ACK,BYE,CANCEL,NOTIFY,PRACK,UPDATE,OPTIONS,REFER
Allow-Events: vq-rtcpxr
Authorization: Digest username="traff",realm="arris-i.org",nonce="",uri="sip:arris-i.org",response=""
Route: <sip:10.1.63.10;transport=udp;lr>
Content-Length: 0
Call Signaling LOG 4
Wed, 12 Apr 2017 09:23:42 +0000
-MSG-> Message Received
SIP/2.0 200 OK
From: TRAFFIC5556 Line2<sip:5556@arris-i.org:5060>;tag=45b308-a530a4b-13c4-50029-58dd2427-34e5bf30-58dd2427
To: TRAFFIC5556 Line2<sip:5556@arris-i.org:5060>;tag=780ff3a6fabbe46aa1cb7511ceb71262.459b
Call-ID: 462c18-a530a4b-13c4-50029-58dd2427-21f4c56a-58dd2427
CSeq: 7341 REGISTER
Via: SIP/2.0/UDP 10.83.10.75:5060;branch=z9hG4bK-58edf1cb-61788274-2bc76e1a
Contact: <sip:5556@10.83.10.75:5060>;expires=300
Server: Sip EXpress router (0.9.6 (i386/linux))
Warning: 392 10.1.63.10:5060 "Noisy feedback tells: pid=3528 req_src_ip=10.83.10.75 req_src_port=5060 in_uri=sip:arris-i.org out_uri=sip:arris-i.org via_cnt==1"
Content-Length: 0

Configuration Parameters Screen

The Configuration Parameters screen shows the values of common cable modem and eDVA configuration parameters. To access this screen, use the **Config Params** link from any advanced screen.

Configuration Parameters

Product | DHCP | MTA | VQM | QoS | Config Params | Basic

CM Config File: tftp://10.1.63.51/cm5cb066bc666e/kja_ArrisGroup_cm.bin
MTA Config File: tftp://10.1.63.51/cm5cb066bc666e/kirk_mta_pc20.bin

Warning: The modem does not store copies of its configuration files. Below you will find the currently used values of parameters commonly set using these files. The files may or may not set these values or contain additional information. This information is listed for convenience only.

Common CM Config Parameters

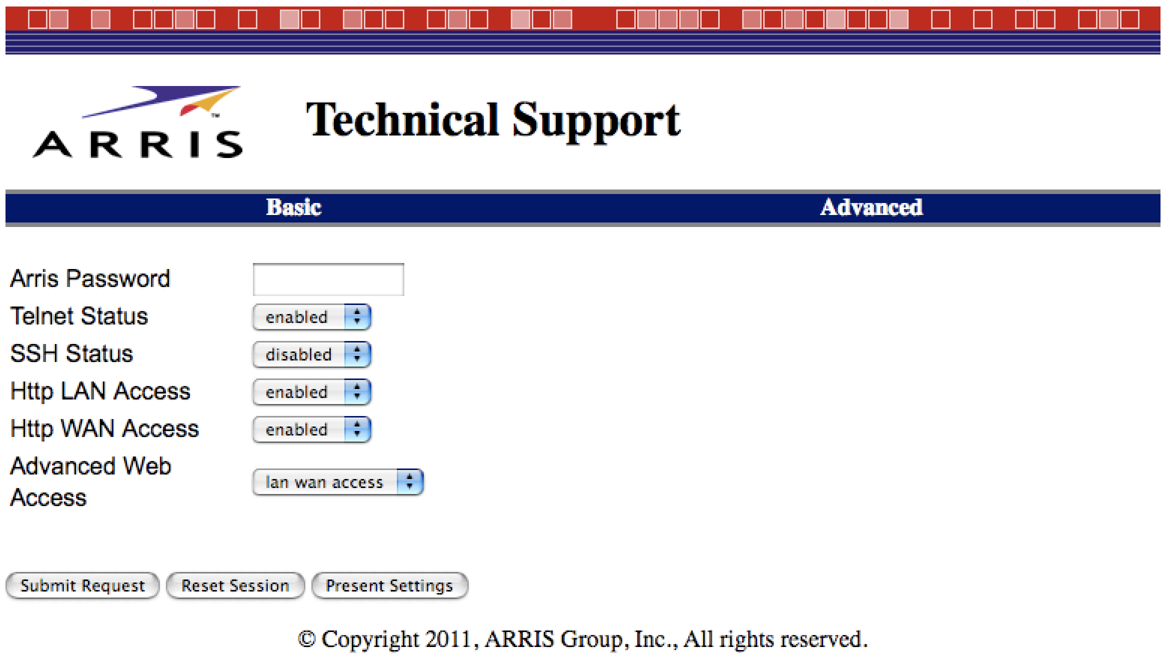
```
NetworkAccess = Yes
UpgradeServer = Not Set
MaxCpeAllowed = 10
PrivacyEnable = 0
SnmpMib = docsDevSwAdminStatus.0 2
SnmpMib = arrisCmDoc30AccessHttpTimeout.0 0
UpstreamServiceFlow =
    SfReference = 0
#   SfId = 5521  -- assigned by CMTS, correlates to PcSfId
    SfClassName = ""
    SfQosSetType = 7
    SfTrafficPriority = 1
    SfMaxTrafficRate = 50000000
    SfMaxTrafficBurst = 50000
    SfMinReservedRate = 0
    SfMinReservedRatePktsize = 64
    SfActiveQosTimeout = 0
    SfAdmittedQosTimeout = 0
    SfMaxConcatBurst = 5000
    SfSchedulingType  = 2
UpstreamServiceFlow =
    SfReference = 0
#   SfId = 5523  -- assigned by CMTS, correlates to PcSfId
    SfClassName = ""
    SfQosSetType = 7
    SfTrafficPriority = 1
    SfMaxTrafficRate = 384000
    SfMaxTrafficBurst = 3044
    SfMinReservedRate = 8000
    SfMinReservedRatePktsize = 0
    SfActiveQosTimeout = 0
    SfAdmittedQosTimeout = 200
    SfMaxConcatBurst = 3044
    SfSchedulingType  = 3
DownstreamServiceFlow =
    SfReference = 0
#   SfId = 5522  -- assigned by CMTS, correlates to PcSfId
    SfClassName = ""
    SfQosSetType = 7
    SfTrafficPriority = 1
    SfMaxTrafficRate = 160000000
    SfMaxTrafficBurst = 96000
    SfMinReservedRate = 0
    SfMinReservedRatePktsize = 64
    SfActiveQosTimeout = 0
    SfAdmittedQosTimeout = 0
DownstreamServiceFlow =
    SfReference = 0
#   SfId = 5524  -- assigned by CMTS, correlates to PcSfId
    SfClassName = ""
    SfQosSetType = 7
    SfTrafficPriority = 7
    SfMaxTrafficRate = 10000000
    SfMaxTrafficBurst = 1522
    SfMinReservedRate = 12000
    SfMinReservedRatePktsize = 64
    SfActiveQosTimeout = 0
    SfAdmittedQosTimeout = 200

UpstreamPacketClassification =
    PcServiceFlow = 5523
    PcIdentifier = 1
    PcRulePriority = 64
    PcActivationState = 1
        PcIpProtocol = 17
        PcIpSourcePortStart = 2427
        PcIpSourcePortEnd = 2427
UpstreamPacketClassification =
    PcServiceFlow = 5523
    PcIdentifier = 2
    PcRulePriority = 64
    PcActivationState = 1
        PcIpProtocol = 17
        PcIpSourcePortStart = 2727
        PcIpSourcePortEnd = 2727
UpstreamPacketClassification =
    PcServiceFlow = 5523
    PcIdentifier = 3
    PcRulePriority = 64
    PcActivationState = 1
        PcIpProtocol = 17
        PcIpSourcePortStart = 5060
        PcIpSourcePortEnd = 5060
DownstreamPacketClassification =
    PcServiceFlow = 5524
    PcIdentifier = 1
    PcRulePriority = 1
    PcActivationState = 1
        PcIpProtocol = 17
        PcIpDestPortStart = 2427
        PcIpDestPortEnd = 2427
DownstreamPacketClassification =
    PcServiceFlow = 5524
    PcIdentifier = 2
    PcRulePriority = 1
    PcActivationState = 1
        PcIpProtocol = 17
        PcIpDestPortStart = 2727
        PcIpDestPortEnd = 2727
DownstreamPacketClassification =
    PcServiceFlow = 5524
    PcIdentifier = 3
    PcRulePriority = 1
    PcActivationState = 1
        PcIpProtocol = 17
        PcIpDestPortStart = 5060
        PcIpDestPortEnd = 5060

#
# Note: Items assigned by CMTS are not included in the config file.
#       PcSfId maps to Instance of docsQosServiceFlowTable.
#
```

Technical Support Screen

The Technical Support screen allows an operator to set access parameters for the CLI and web pages. This screen does not have a link from the other screens. See Enabling CLI Access Using the Web Pages for details on accessing and using this screen.



The fields are:

Arris Password
: Enter the Password of the Day before submitting changes to the other fields.

Telnet Status
: Enables or disables CLI access for both Telnet and SSH.

Http LAN Access
: Enables or disables access to the troubleshooting pages from the Ethernet or WiFi interfaces.

Http WAN Access
: Enables or disables access to the troubleshooting pages from the cable network.

Action

Perform the following tasks as needed.

Accessing the Standard Pages

Follow these steps to access the standard troubleshooting pages.

1. Make sure the E-UE is configured to allow access to the pages (see “Controlling Access to the Interface).”
2. Obtain the cable modem IP address of the E-UE.

Note: If the E-UE has not registered, you can access the pages only from the Ethernet interface. After registration, you can use either the RF or Ethernet interfaces.

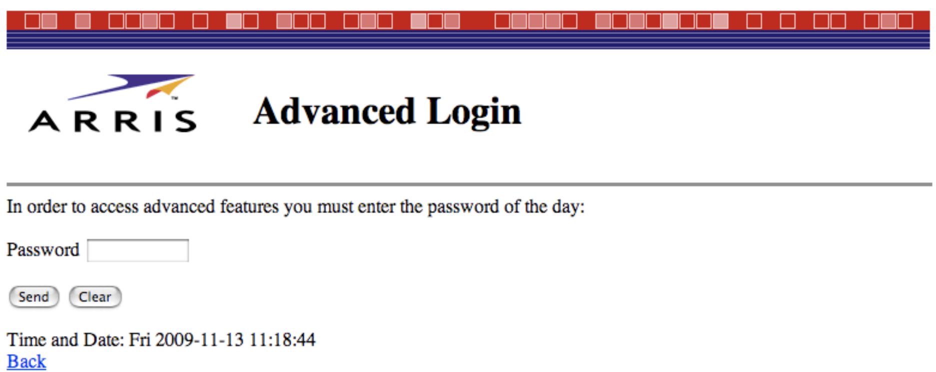
3. The Ethernet interface recognizes connections to the address **`192.168.100.1`**, whether or not the modem has registered. To use this address, set your computer's IP address to an address in the same subnet, such as **`192.168.100.2`**.
4. Start your web browser and access the E-UE using the address **http://192.168.100.1/**

Accessing the Advanced Pages

Follow these steps to access the advanced troubleshooting pages.

1. Perform steps 1 and 2 of "see "*Accessing the Standard Pages* (page 304)".
2. Obtain the Password of the Day from the PWoD tool. See "see "*Using the Password of the Day Tool* (page 287)" for details about the PWoD tool.
3. Start your web browser and access the E-UE, using the address **http://192.168.100.1/cgi-bin/product_cgi** (replace the IP address shown with the cable modem IP address if desired).

   The E-UE displays a web form that prompts you for the password.



4. Enter the Password of the Day in the web form and press Enter.

   The E-UE displays the Product Details Screen.

Note: If the Password of the Day does not allow you to access the advanced pages, verify that the date shown below the password entry field matches the date associated with the Password of the Day.

References

The following reference material is available:

Supported Calling Features

Touchstone firmware supports the following calling features:

- Automatic Number Assignment Confirmation (ANAC) via CID2
- Direct Distance Dialing (DDD)
- Critical Interdigital Timing for Dialing Plan
- International DDD (IDDD) Local Billing Control
- Residence Distinctive Alerting Service
- Free Terminating Service
- Code Restriction & Diversion
- Toll Restricted Service
- DTMF Dialing
- Pulse Dialing
- CLASS™: Calling Number Delivery
- CLASS: Customer Originated Trace
- CLASS: Anonymous Call Rejection
- CLASS: Calling Number Delivery Blocking
- CLASS: Calling Identity Delivery & Suppression
- CLASS: Calling Name Delivery Blocking
- CLASS: Calling Name Delivery
- CLASS: Calling Identity Delivery on Call Waiting
- Speed Calling 8
- Speed Calling 30
- Call Waiting
- Cancel Call Waiting (*70)
- Call Waiting Deluxe
- Access to Telecommunications Relay Service (TDD)
- Intercept Routing for blank/changed/etc. phone numbers
- Customer-Changeable Speed Calling
- Call Forwarding Variable
- Call Forwarding Busy Line
- Call Forwarding — Don’t Answer — All Calls
- VIP Alert (Distinctive Ringing)
- Visual Message Waiting Indicator (FSK)
- Message Waiting Tone (stutter dial tone)

- Conference Calling — Six-Way Station Controlled
- Call Hold, Call Pick-up, Toll Free Calling
- Emergency Calling Services (E911)
- Customer Call Back (Automatic Recall) (*69)
- Three-Way Calling
- Service Provider Originated Trace
- Courtesy Ring Generation
- Multiple Directory Numbers on a Line
- Customer Access Treatment (CAT) code restrictions
- Semi-Restricted Originating & Terminating (including 1010xxx blocking)
- Fully Restricted Originating & Terminating
- Single-Digit Dialing
- Manual Line Service
- Direct Connect
- Denied Terminating Service
- Denied Originating Service
- Local Number Portability
- Remote Activation of Call Forwarding (RACF)
- Outside Calling Area Alerting (OCAA)

Country Code Templates

Use the **ppCfgMtaCountryTemplate** object to set the country code template.

TS11.1 firmware supports the following country code templates:

Name	Gain settings (dB)		Flash time (ms) (NA load only)	
	Tx	Rx	Min	Max
North America 5/7 (1)	−5	−7	250	1200
Germany (7)	−3	−9.5	300	500
Poland (13)	−3	−9.5	50	520
North America 3/3 (17) (DOCSIS default)	−3	−3	250	1200
North America 0/9 (18)	0	−9	250	1200
Sweden (22)	−3	−9.5	250	1200
Switzerland (29)	−3	−9.5	88	600
Poland1010 (30)	−13	−19.5	70	250
Germany2 (31)	−3	−9.5	300	500
North America 6/6 (32) (.TW default)	−6	6	250	1200

North American Ring Cadences

The following ring cadences may be provisioned using the PacketCable NCS Signaling MIB (see PKT-SP-MIB-SIG1.5-I01-050128). The following table shows the default ring cadences for North America.

Name	Description	Default
L/RG	Standard Ringing	2 seconds on, 4 seconds off
L/R0	Distinctive Ringing #0	2 seconds on, 4 seconds off
L/R1	Distinctive Ringing #1	2 seconds on, 4 seconds off
L/R2	Distinctive Ringing #2	800 ms on, 400 ms off, 800 ms on, 4 seconds off
L/R3	Distinctive Ringing #3	400 ms on, 200 ms off, 400 ms on, 200 ms off, 800 ms on, 4 seconds off
L/R4	Distinctive Ringing #4	300 ms on, 200 ms off, 1 second on, 200 ms off, 300 ms on, 4 seconds off
L/R5	Distinctive Ringing #5	500 ms on, 5.5 seconds off (not repeated)
L/R6	Distinctive Ringing #6	2 seconds on, 4 seconds off
L/R7	Distinctive Ringing #7	2 seconds on, 4 seconds off
L/RS	Ring Splash	500 ms on, 5.5 seconds off (not repeated)
L/RT	Ringback Tone	2 seconds on, 4 seconds off

Touchstone firmware uses the default ring cadences shown above when the country template is provisioned to be one of the following:

- `northAmerica57`
- `northAmerica33`
- `northAmerica09`
- `northAmerica66`

Template (i.e. hard-coded country-specific) based ring cadences are used by default when the country template is provisioned. This default behavior may be overridden by setting the "Provisioned Ring Cadences" CallP Feature Switch setting, and updating the eDVA configuration file with the provisioning for the appropriate MIB objects to define ring cadences (for example, **pktcSigDevRgCadence**). To make this setting, add `0x02000000` to the current feature switch setting in the CM configuration file.

Germany/Germany2 Ring Cadences

The following table shows the default ring cadences for Germany and Germany2.

Name	Description	Default
RG	Standard Ringing	1000 ms on, 4000 ms off
R0	Distinctive Ringing #0	2 seconds on, 4 seconds off
R1	Distinctive Ringing #1	2 seconds on, 4 seconds off
R2	Distinctive Ringing #2	800 ms on, 400 ms off, 800 ms on, 4 seconds off
R3	Distinctive Ringing #3	400 ms on, 200 ms off, 400 ms on, 200 ms off, 800 ms on, 4 seconds off
R4	Distinctive Ringing #4	300 ms on, 200 ms off, 1 second on, 200 ms off, 300 ms on, 4 seconds off
R5	Distinctive Ringing #5	500 ms on, 5.5 seconds off (not repeated)
R6	Distinctive Ringing #6	2 seconds on, 4 seconds off
R7	Distinctive Ringing #7	2 seconds on, 4 seconds off
RS	Ring Splash	450 ms on (not repeated)
RT	Ringback Tone	1 second on, 4 seconds off

Poland/Poland1010/Slovakia Ring Cadences

The following table shows the default ring cadences for Poland and Slovakia.

Name	Description	Default
RG	Standard Ringing	1 second on, 4 seconds off
R0	Distinctive Ringing #0	2 seconds on, 4 seconds off
R1	Distinctive Ringing #1	2 seconds on, 4 seconds off
R2	Distinctive Ringing #2	800 ms on, 400 ms off, 800 ms on, 4 seconds off
R3	Distinctive Ringing #3	400 ms on, 200 ms off, 400 ms on, 200 ms off, 800 ms on, 4 seconds off
R4	Distinctive Ringing #4	300 ms on, 200 ms off, 1 second on, 200 ms off, 300 ms on, 4 seconds off
R5	Distinctive Ringing #5	500 ms on, 5.5 seconds off (not repeated)
R6	Distinctive Ringing #6	2 seconds on, 4 seconds off
R7	Distinctive Ringing #7	2 seconds on, 4 seconds off
RS	Ring Splash	450 ms on (not repeated)
RT	Ringback Tone	1 second on, 4 seconds off

Switzerland Ring Cadences

The following table shows the default ring cadences for Switzerland.

Name	Description	Default
L/RG	Standard Ringing	1 seconds on, 4 seconds off
L/R0	Distinctive Ringing #0	2 seconds on, 4 seconds off
L/R1	Distinctive Ringing #1	2 seconds on, 4 seconds off
L/R2	Distinctive Ringing #2	800 ms on, 400 ms off, 800 ms on, 4 seconds off
L/R3	Distinctive Ringing #3	400 ms on, 200 ms off, 400 ms on, 200 ms off, 800 ms on, 4 seconds off
L/R4	Distinctive Ringing #4	300 ms on, 200 ms off, 1 second on, 200 ms off, 300 ms on, 4 seconds off
L/R5	Distinctive Ringing #5	500 ms on, 5.5 seconds off (not repeated)
L/R6	Distinctive Ringing #6	2 seconds on, 4 seconds off
L/R7	Distinctive Ringing #7	2 seconds on, 4 seconds off
L/RS	Ring Splash	450 ms on

Customizing Default Ring Cadences

Any of the above ring cadences may be customized in the eDVA configuration file. All MIB objects are eDVA based; therefore, the first cadence is index 0.

The ring cadence is internally represented as a 64-bit string and provisioned in hex format. The ring cadence representation starts with the first **`1`** in the bit string pattern. Leading zeros are ignored, thus shortening the overall ring cadence duration. Each bit represents 100 ms of ringing (or tone in the case of L/RT); **`1`** is ring on, **`0`** is ring off.

All 64 bits must be provisioned. The least significant 4 bits are used for representing repeatable characteristics: **`0000`** indicates that the ring cadence repeats, and **`1000`** indicates a non-repeatable ring cadence. Therefore, only the first 60 bits are used to represent the actual ring cadence for a maximum duration of 6 seconds.

As mentioned earlier, shorter ring cadences may be provisioned by padding the ring cadence with leading zeros. For example, a ring cadence of 0.5 seconds on, 4 seconds off, repeatable, has a value of **`0x0001F00000000000`** and would be provisioned in the eDVA configuration file as **`00.01.F0.00.00.00.00.00`**.

Default Tone Settings

The following tables show default tones for supported country templates. The columns in each table are as follows:

type
The type of tone (busy, dial tone).

level
The tone level, in dB.

Freq. Type
Either **`1`** (first frequency modified by the second) or **`2`** (summation).

 # Freq.
The number of frequencies used to generate the tone (1–4).

Frequencies
The frequencies used to generate the tone.

 # on/off
The number of on/off cycles in the tone pattern (1–4).

1st tone – 4th tone
The duration, in milliseconds, of the on/off segments of each tone cycle.

rep. count
The number of times the tone pattern is repeated.

tone steady
Which of the four tones, if any, are held indefinitely after the pattern completes (used, for example, with stutter dial).

Note: Not all firmware loads support all country code templates and tones.

Germany (all)

pktcSigDevToneEntry			pktcSigDevMultiFreqToneEntry(s)										
Tone	Rep Count	Tone Steady	Tone Number	F1	F2	F3	F4	Mode	%	dB Level	On Time	Off Time	Freq. Repeat
Busy(1)	5000	2	1	425	0	0	0	2	0	-130	480	480	0
Confirmation(2)	2	2	1	425	0	0	0	2	0	-100	200	100	0
Dial(3)	0	1	1	425	0	0	0	2	0	-130	5000	0	0
Message Waiting(4)	0	1	1	400	425	0	0	2	0	-160	5000	0	0
Offhook Warning(5)	5000	2	1	425	0	0	0	2	0	-130	240	240	0
Ringback(6)	5000	2	1	425	0	0	0	2	0	-130	1000	4000	0
Reorder(7)	5000	2	1	425	0	0	0	2	0	-130	240	240	0
Stutterdial(8)	0	1	1	400	425	0	0	2	0	-160	5000	0	0
Call Waiting 1 (9)	1	2	1	425	0	0	0	2	0	-180	200	200	0
Call Waiting 2 (10)	1	2	1	440	0	0	0	2	0	-130	100	100	0
Call Waiting 3 (11)	2	2	1	440	0	0	0	2	0	-130	100	100	0
Call Waiting 4 (12)	0	2	1	440	0	0	0	2	0	-130	100	100	0
			2	440	0	0	0	2	0	-130	300	100	0
			3	440	0	0	0	2	0	-130	100	100	0

North America

type	level	Freq. Type	# Freq.	Frequencies 1st	2nd	3rd	4th	tone number	on time	off time	rep. count	tone steady
Busy(16)	-45.3	1	2	480	620	0	0	1	500	500	30	2
Confirmation(17)	-13	1	2	350	440	0	0	1	1000	1000	0	4
								2	1000	1000		
								3	1000	1000		
Dial(18)	-13	1	2	350	440	0	0	1	5000	0	0	1
Offhook Warning (20)	-6	1	4	1400	2060	2450	2600	1	100	100	5000	2
Ringback(21)	-19	1	2	440	480	0	0	1	2000	4000	5000	2
Reorder(22)	-24	1	2	480	620	0	0	1	250	250	60	2
Stutterdial(23)	-13	1	2	350	440	0	0	1	100	100	0	4
								2	100	100		
								3	100	0		
								4	5000	0		
Message Waiting (24)	-13	1	2	350	440	0	0	1	100	100	10	2
								2	5000	0		
Call Waiting 1 (25)	-13	1	1	440	0	0	0	1	300	0	1	2
Special Information Tone(30)	-17	1	3	950	1400	1800	0	1	330	1000	2	3

Norway/Sweden

pktcSigDevToneEntry			pktcSigDevMultiFreqToneEntry(s)										
Tone	Rep Count	Tone Steady	Tone Number	F1	F2	F3	F4	Mode	%	dB Level	On Time	Off Time	Freq. Repeat
Busy(1)	5000	2	1	425	0	0	0	2	0	-110	500	500	0
Confirmation(2)	5000	2	1	425	0	0	0	2	0	-110	400	0	0
			2	470	0	0	0	2	0	-110	400	0	0
Dial(3)	0	1	1	425	0	0	0	2	0	-60	5000	0	0
Message Waiting(4)	5000	5	1	425	0	0	0	2	0	-60	700	60	0
Offhook Warning(5)	5000	2	1	1400	0	0	0	2	0	-250	400	5000	0
			2	1400	0	0	0	2	0	-250	0	5000	0
			3	1400	0	0	0	2	0	-250	0	5000	0
Ringback(6)	5000	2	1	425	0	0	0	2	0	-110	1000	4000	0
Reorder(7)	5000	2	1	425	0	0	0	2	0	-110	200	200	0
Stutterdial(8)	5000	2	1	425	0	0	0	2	0	-60	700	60	0
Call Waiting 1 (9)	1	2	1	1400	0	0	0	2	0	-250	200	600	0
Call Waiting 2 (10)	1	2	1	1400	0	0	0	2	0	-220	100	100	0
Call Waiting 3 (11)	2	2	1	1400	0	0	0	2	0	-220	100	100	0
Call Waiting 4 (12)	0	2	1	1400	0	0	0	2	0	-220	100	100	0
			2	1400	0	0	0	2	0	-220	300	100	0
			3	1400	0	0	0	2	0	-220	100	100	0

Poland/Poland1010

pktcSigDevToneEntry			pktcSigDevMultiFreqToneEntry(s)										
Tone	Rep Count	Tone Steady	Tone Number	F1	F2	F3	F4	Mode	%	dB Level	On Time	Off Time	Freq. Repeat
Busy(1)	5000	2	1	425	0	0	0	2	0	-60	500	500	0
Confirmation(2)	2	2	1	425	0	0	0	2	0	-100	200	100	0
Dial(3)	0	1	1	425	0	0	0	2	0	-100	5000	0	0
Message Waiting(4)	0	1	1	350	425	0	0	2	0	-100	5000	0	0
Offhook Warning(5)	0	1	1	1440	0	0	0	2	0	-360	5000	0	0
Ringback(6)	5000	2	1	425	0	0	0	2	0	-60	1000	4000	0
Reorder(7)	5000	2	1	425	0	0	0	2	0	-60	500	500	0
Stutterdial(8)	0	1	1	350	425	0	0	2	0	-100	5000	0	0
Call Waiting 1 (9)	2	2	1	425	0	0	0	2	0	-100	150	150	0
Call Waiting 2 (10)	1	2	1	440	0	0	0	2	0	-130	100	100	0
Call Waiting 3 (11)	2	2	1	440	0	0	0	2	0	-130	100	100	0
Call Waiting 4 (12)	0	2	1	440	0	0	0	2	0	-130	100	100	0
			2	440	0	0	0	2	0	-130	300	100	0
			3	440	0	0	0	2	0	-130	100	100	0

Switzerland

pktcSigDevToneEntry			pktcSigDevMultiFreqToneEntry(s)										
Tone	Rep Count	Tone Steady	Tone Number	F1	F2	F3	F4	Mode	%	dB Level	On Time	Off Time	Freq. Repeat
Busy(1)	5000	2	1	425	0	0	0	2	0	-200	500	500	0
Confirmation(2)	2	2	1	425	0	0	0	2	0	-100	200	100	0
Dial(3)	0	1	1	425	0	0	0	2	0	-150	5000	0	
Message Waiting(4)	0	1	1	400	425	0	0	2	0	-160	5000	0	0
Offhook Warning(5)	5000	2	1	425	0	0	0	2	0	-130	240	240	0
Ringback(6)	5000	2	1	425	0	0	0	2	0	-200	1000	4000	0
Reorder(7)	5000	2	1	425	0	0	0	2	0	-200	200	200	0
Stutterdial(8)	5000	1	1	425	340	0	0	2	0	-170	1100	0	0
			2	425	425	0	0	2	0	-170	1100	0	0
Call Waiting 1 (9)	1	2	1	425	0	0	0	2	0	-290	200	200	0
Call Waiting 2 (10)	1	2	1	440	0	0	0	2	0	-130	100	100	0
Call Waiting 3 (11)	2	2	1	440	0	0	0	2	0	-130	100	100	0
Call Waiting 4 (12)	0	2	1	440	0	0	0	2	0	-130	100	100	0
			2	440	0	0	0	2	0	-130	300	100	0
			3	440	0	0	0	2	0	-130	100	100	0

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