Mediant 2000 & TP-1610
Nortel H.248 Release Notes

Version 4.2

Document # LTRT-00728

Trademarks

AC, AudioCoded, AudioCodes, AudioCodes logo, IPmedia, Mediant, MediaPack, NetCoder, Stretto, TrunkPack and VoicePacketizer, are trademarks or registered trademarks of AudioCodes Limited.

All other products or trademarks are the property of their respective owners.

Customer Support

Customer technical support and service are provided by AudioCodes' Distributors, Partners, and Resellers from whom the product was purchased. For Customer support for products purchased directly from AudioCodes, contact support@audiocodes.com.

Abbreviations

Each abbreviation, unless widely used, is spelled out in full when first used. Only industry-standard terms are used throughout this manual. Hexadecimal notation is indicated by 0x preceding the number.

Applicable Products

This manual provides additional information for the following AudioCodes products:

TrunkPack Series: TP-1610

Mediant 2000/ MEGACO,

Notice

These Release Notes describe the functionality of the AudioCodes' TrunkPack Series Boards supported by Software Release 4.2. Information contained in this document is believed to be accurate and reliable at the time of printing. However, due to ongoing product improvements and revisions, AudioCodes cannot guarantee the accuracy of printed material after the Date Published nor can it accept responsibility for errors or omissions.

For **Technical Support** please contact:
E-mail: support@audiocodes.com
In the USA, fax 408-577-0492
In other countries, fax + 972-3-539 4040

This document is subject to change without notice.

Please refer to the current release notes that may be included with your documentation or hardware delivery.

Date Published: Feb-11-2004 Date Printed: Mar-16-2004

Table of Contents

1 What's New in Release 4.2

1.1 Hardware Supported in Release 4.2

- This version also supports the following hardware platforms:
  - TrunkPack TP-1610.
  - Mediant 2000.

1.2 Host API Related New Features

- VoPLib compilation with GCC 3.2 compiler is now supported for Linux/Solaris Operating Systems.
- VoPLib in TPCNP mode (not PCI) is supported for Solaris/Sparc 9.
- In many cases, new event names and corresponding event info structures were introduced. The old ones are still present for backward compatibility.
- The ability to get firmware download (via PCI) progress was implemented. Refer to last optional acInitLibrary() parameter.
- It is now possible to enter a User-defined board handle as an input parameter, i.e., the User can request the board handle that acOpenRemoteBoard() returns.
- When using acFullConfigurationModeWithIniFileCreation the *ini* file is no longer created as part of the board opening process, (it is now based on acFullConfigurationModeWithParametersSending).
- It is possible to request at any time after opening the board the version of the firmware that was loaded.
- Wave file support was added for acG723Low, acGSM610, acGSM610MS, acG726_16/24/32/40 coders.
- The following functions are now internal and should not be called by the User application: acGetBoardCPUGroup(), acAnalogIFReadRegister(), acAnalogIFWriteRegister(), acGetTFTPServerPath(), acGetTPNCPSocket().
- In all def*ini*tions, the prefix P9050 has been changed to PLX.
- Files p9050_lib.c and p9050_lib.h have been renamed to PLXlib.cpp and PLXlib.h.
- A new API function acGetChannelState() was added for access to the acOpenedChannels array. For backward compatibility, the internal lib structure acOpenedChannels is still opened for the User application, but this can change in future versions.

Note: PCI Users that upgrade to the latest VoPLib version are advised to upgrade the PCI drivers as well.

- The following API functions were added (refer to the VoPLib Reference Library User's Manual, Document #: LTRT-00744 for additional information about each API function):

```
acUpdateChannelInfo(),acGetFirmwareVersionString()
acSetDateAndTime()
acActivateT38Channel(), acDeActivateT38Channel()
acUpdateSyslogParams()
acChangeNWParameters()
acChangeDefaultGatewayAddress()
acQueryTemperature()
acGetChannelState()
acAddDestinationPoint()
acPstnApplyTrunkConfig(), acPstnConfigureTrunk(), acPstnDeleteTrunk()
acPstnStartTrunkTraffic(), acPstnStopTrunkTraffic()
acContinuePlay(), acPausePlay()
acSetCustomerKey(), acGetCustomerKey()
acPlayStream() ... (HTTP streaming related API functions)
acGetDefaultDigitMapParameters() ... (Digit-map related API functions)
acSS7...() (SS7 related API function)
```

- The following events were added (refer to the VoPLib Reference Library User's Manual, Document #: LTRT-00744 for additional information about each event):

```
acEV_CUSTOMER_KEY_RESPONSE
acEV_USER_DEFINED_TONE_DETECTED
acEV_CO1_TONE_DETECTED
acEV_CO2_TONE_DETECTED
acEV_TDM_QUERY_ALL_CONNECTIONS
acEV_DSP_CHANNELS_RESOURCES_BELOW_THRESHOLD
acEV_DSP_CHANNELS_RESOURCES_ABOVE_THRESHOLD
acEV_DSP_CHANNELS_RESOURCES_DECREASED
acEV_CLOSE_CHANNEL_DUE_TO_DEAD_DSP
acEV_OPEN_CHANNEL_FAILED_DUE_LACK_OF_RESOURCES
acEV_CAS_AKNOWLEDGE_WK
acEV_PLAY_STREAM_ACTIVATED
acEV_RECORD_STREAM_ACTIVATED
acEV_PLAY_STREAM_ENDED
acEV_RECORD_STREAM_ENDED
acEV_STREAM_SESSION_CLOSED
acEV_HTTP_ERROR
acEV_STREAM_ERROR
acEV_PLAY_TONE_ENDED
acEV_DIALED_STRING
acEV_CURRENT_DIALED_SUB_STRING
acEV_CURRENT_DIGIT_MAP
acEV_CALL_AGENT_STATUS_RESPONSE
acEV_HIGH_TEMPERATURE
acEV_FAR_END_CONNECTED
acEV_TEMPERATURE
acEV_TRUNK_TRAFFIC_ACKNOWLEDGE
acEV_MESSAGE_LOG
acEV_REPORT_ICMP_UNREACHABLE
acEV_REPORT_ICMP_REACHABLE
acEV_TTY_STARTED
acEV_TTY_ENDED.
```

  In addition, many event types were added to support SS7/MTP2-3.

- The version includes the following bug fixes:

  - A problem in acGetNumWaitingCmd() was fixed.
  - A problem in acQueryLanStatus() was fixed (memory operations on Solaris Sparc).
  - A problem in the sequence number mechanism in case of unsuccessful command sending was fixed.
  - The enumeration acTTrunkPackBoardType was fixed in order to set a constant value for each board type.
  - A Solaris mutex creation problem was solved (conflicting PTHREAD_PRIO_INHERIT and PTHREAD_MUTEX_RECURSIVE).

1.3 Ethernet-Interface Related New Features

- Improved NAT. Two new NAT board parameters were introduced into the NAT feature:

EnableIpAddrTranslation and EnableUdpPortTranslation. These parameters allow the User to specify the type of compare operation that takes place on the first incoming packet. The User can specify, for example, that when receiving a first incoming packet with different source UDP port, NAT does take place.

- Robust reception of RTP streams using a new filtering mechanism. This new mechanism filters out unwanted RTP streams that are sent to the same port on the board. These multiple RTP streams can result from traces of previous calls, call control errors and deliberate attacks.

- New API for online network configuration - acChangeNWParameters(). This new API allows the User to change the IP address, subnet mask and default GW address without resetting the board. This API doesn't affect currently opened channels - only channels that are opened after this API is invoked behave according to the new network parameters.

- ICMP unreachable packet reporting - the board can now report on reception of ICMP unreachable packets via acEV_REPORT_ICMP_UNREACHABLE (andacEV_REPORT_ICMP_REACHABLE). The User can configure the mechanism using the parameters EnableICMPUnReachableReport and ICMPUnReachableReportInterval.

- LAN watchdog - This new mechanism detects LAN failures on the board and restarts the board if required. A LAN failure can result from a software or hardware malfunction. A new board parameter EnableLanWatchDog enables this feature (default is disabled).

1.4 VoIP Engine New Features

- DSP Resource Management (DRM) - DSP channel resources are now dynamically allocated according to the application needs, enabling better utilization of the DSP resources. In addition, DSP resources can now be allocated to IP-only channels (e.g. for IP-to-IP mediation), without blocking the associated TDM resource (e.g., B-Channel). VoPLib API control over DRM is done using two new acTExtChannelParam fields – DspResourceAllocation and DisableLocalSwitchingConnection.

- Mediation enhancements – the mediation module now supports multiple connections on the same session. For example, an incoming RTP stream can be connected to multiple destinations – to the TDM, to other RTP channels and to PCI channels (for recording). Multiple IP destinations do not require additional DSP resources. Use the new VoPLib API command acAddDestinationPoint() to connect the output of a channel to an additional endpoint.

- HTTP streaming for playing and recording – it is now possible to access .wav or .au file stored on a remote Web server and to play it to the TDM or to the IP. It is also possible to record the voice from the TDM or from the IP to a remote file via HTTP.

- Activate/Deactivate T.38 API functions [acActivateT38Channel(), acDeActivateT38Channel()] were added to enable better control over T.38 sessions.

- Support for reordered RTP packets – a new algorithm has been implemented to handle RTP packets which are received reordered. This feature improves the voice quality on a network which suffers from packet reordering problems.

- Support for the EVRC TTY/TDD extension (3GPP2 C.S0014-0-3) and the CDMA TTY/TDD M*ini*mum Performance Specification (3GPP2 C.S0028-0 v2.0) ITU standards. Two additional TPNCP events were added: acEV_TTY_STARTED and acEV_TTY_ENDED.

- Support for the OKI-ADPCM coder (VOX ADPCM).

- A new G.168-2000 compliant Echo Canceller with support for up to 128 msec of echo tail has been added. Refer to new configuration parameter MaxEchoCancelerLength. When set to 64 msec or more, the number of available board channels is reduced.

- Support for ITU-T Q.724 COT (Continuity Test) signals detection and generation. To generate, use the new API function acPlayTone with acUserDefinedCo1Tone or acUserDefinedCo2Tone.

- CNG fax tone detection and T.38 Relay support. Enables fast fax session detection for CNG-generating fax machines.
- Support for a special Jitter Buffer optimization mode for Modem or Fax. User should set the jitter buffer Optimization Factor to 13 for Modem or Fax calls.

Note: Installation and use of vocoders is subject to obta*ini*ng the appropriate license in advance of use, and to royalty payments

The DSPVersionTemplateNumber *ini* file parameter has the following different meanings in different products:

- **TP-1610**
  - 0 = DSP firmware supports 240/480* channels of PCM/ADPCM, OKI ADPCM, G.723, G.729A, GSM FR and NetCoder.
  - 1 = reserved.
  - 2 = reserved.
  - 3 = reserved

Note: G.728 coder can be supported; for additional information, contact your AudioCodes representative.

1.5 PSTN Interface New Features

- The default trunk protocol was changed to "None" indicating that the trunk has not yet been configured by the User. In this mode the trunks can not run any traffic and the User does not get any alarms or events.
- Online trunk configuration is now supported. The User can now configure new trunks or change the configuration of existing trunks without restarting the system. The feature is currently available via the VoPLib API and via the embedded Web interface. The following changes are included:
  - Via Web or API the User can set the protocol and other trunk parameters and activate the trunk.
  - Reconfiguring an active trunk is possible after deactivating/deleting it.
  - Online loading of CAS tables is supported via the Web.
- Support for SS7/MTP-3 (in addition to the existing support for MTP2):
  - MTP-3 (SS7 network layer) protocol stack - Q.782 compliant.
  - Alarms and Indications for MTP-2 and MTP-3

- Support dynamic configuration and a rich set of maintenance actions required to manage MTP-3 at run time
- Initial provisioning using a new textual configuration file for MTP.
- Protocol inter-working between MTP-2/MTP-3 and M2UA/M3UA:
  - MTP-2 ⇔ M2UA
  - MTP-2 ⇔ MTP3
  - MTP-2 ⇔ MTP-3 ⇔ M3UA
  - MTP-3 ⇔ MTP-3 (SS7 STP/SSP)
- Several link-types can be used on same board.
- Support for two different signaling-nodes on same TrunkPack module.
- Support for all types of SS7 links (including F-links)
- MTP-2 56 kbps links supported (for T1 only).
- LSSU length: 1 and 2 supported.
- MTP-2: Variant type configurable via *ini* file.

Note:	The format of the MTP configuration file may change in future releases.

- ISDN Behavior bit - the default of ISDNIBEHAVIOR parameter is NS_RESTART_INDICATION
- Additional validation for ISDN and a new error indication: ISDN_INCOMPATIBLE_INCOMING_SETUP. This event notifies the User on an error in the incoming call message (SETUP). The error could be an unknown IE, missing mandatory IE, corrupted IE etc. The board automatically replies with RELEASE_COMPLETE to the far end.
- Additional information is included in the acEV_PSTN_INCOMING_CALL_DETECTED event: SourceNumberPlan.
- Additional parameters in the acPSTNPlaceCall() API command: SourceNumberPlan, DisplaySize and DisplayString.
- A new *ini* file parameters for T1 NFAS : NFASGROUPNUMBER. This parameter defines which NFAS group each trunk belongs to. This parameter is relevant only for T1 ISDN. When using NFAS this parameter is mandatory.
- The limitation regarding the order of the T1 NFAS trunk def*ini*tions was removed. The NFAS group members can be defined anywhere in the board trunks (including the primary trunk).
- Two PSTN performance monitoring limitations were removed - Path Coding Violation and UnavailableSeconds.
- In this version we introduce an improvement in the PSTN physical alarm processing: the LOS alarm (Loss Of Signal) and LOF alarm (Loss Of Frame) were separated and are presented in

the corresponding physical alarm.

1.6 SNMP Support

- New parameters were added to the proprietary acBoardMIB (more details can be found in the MIB):
  - Additional MEGACO parameters were added.
  - Added 'Set default' functionality.
  - Supplementary Parameters can added (for set actions only). The *ini* file syntax is used: "<Parameter>=<Value>".
  - Trunk LEDs – provides information by color on trunk status (similar to the front panel indications).
  - Channel Defaults – new parameters added.
  - ModCount parameter (TP-1610/ (IsMaster)
- Two new read-only parameters were added to the proprietary acBoardMIB:
  acLedStatusColor - Trunk status LED color (Off / Red / Green) and
  acLedStatusColor - Trunk status LED blinking indication (Steady / Blink).
- The functionality of the *ini* file parameter 'LineStatusChangeTrapEnableDefault' has been changed (refer to the User's Manual for more details).
- Note that SNMP is not supported on PCI controlled boards (TP-260, TP-260/UN)

1.7 MGCP New Features

- New package:
  - FMTP package - dynamic payload type
- New signals:
  - Call waiting2 (wt2)
  - Call waiting3 (wt3)
  - Call waiting4 (wt4)
  - Wink signal (prp)
  - Metering tone (mtr)
- New event:
  - RTP/RTCP Timeout (rto)
- Support for second endpoint ID parameter
- DNS support - MGCP notified entity supports conversion of URL
- Improved MGCP Quarantine mode
- New coder - GSM-EFR

- Increase max number of Digit Maps to 50.
- Gateway now sends new RSIP if call agent does not accept the endpoint with "200 OK" response
- Support for second endpoint ID according to RFC-2705

1.8 H.248 (MEGACO) Support

- The following MEGACO (H.248) packages were added in this version:
  - MF generation package (mfg)
  - MF detection package (mfd)
- Mediation support between ATM to IP, or IP to IP, or ATM to ATM, using 0-2 DSP channels (previously required 2 DSP channels in all scenarios).
- New *ini* file parameter - MGCPCommunicationLayerTimeout for setting the retransmission expiration in the retransmission module.
- New *ini* file parameter - KeepAliveInterval for setting the interval between two KeepAlive transactions.
- The SDP proprietary parameter 'Chantouse' is no longer used. The allocation of DSP is done internally according to the need.
- It is now possible to open a context with one RTP termination, and play Announcements/CallProgressTones to the network.

1.9 Web Browser Support

1.9.1 Web Browser Features

- User help - when delaying for more then 1 second on parameter name, a short description of this parameter and valid values is displayed.
- Invalid parameter value warning alert – invalid parameters are colored red and a short warning message is displayed.
- HTTP port can be configured via *ini* file.
- The User can now retrieve the board configuration in *ini* file format. The *ini* file is downloaded from the board via the Web browser and contains all parameters that are different from their default value. The *ini* file can then be uploaded to a second board in order to apply the same configuration.
- Restore default button – before uploading an *ini* file to the board, the User can choose to set all configuration parameters to their default values by pressing the "Restore Defaults" button. If an *ini* file is loaded without restoring the defaults first, only the parameters that appear in the *ini* file are modified.
- Support for Date and Time configuration.
- Online configuration of the PSTN Trunks (refer to PSTN section).
- New SS7 Web pages.
- Message log page – adds the option to watch the error logs directly without an external

SysLog server.

- Save Configuration button - burning the current configuration to the flash memory without resetting the board. This action should be done before activating traffic in the board.

- New adm*ini*stration features:

    - Generic *ini*-Parameters page for configuration of special parameters that are not supported by the regular web pages. Users can enter the parameter name or select the parameter from a list.

    - Logo images upload - the User has the option to change the logo that appears in the web pages.

    - Error control page - the User can define criteria for filtering error messages that are sent to different interfaces (e.g., by severity).

- Note that the Web server is not supported on PCI controlled boards (TP-260, TP-260/UN)

1.9.2 Web Browser Interoperability

The Web browser connection to AudioCodes' boards was tested using the following browsers:

- Microsoft™ Internet Explorer™ – Ver 5.0 or newer

- Netscape™ Navigator™ – Ver 7.0 or newer

1.10 SIGTRAN Support

In addition to M2UA and IUA that were introduced in previous versions, our SIGTRAN solution now includes support for M3UA according to RFC 3332. In addition, fail-over is now supported for SIGTRAN.

SIGTRAN is an IETF working group whose primary purpose is to address the transport of packet-based PSTN signaling over IP Networks. These protocols are used to transport SS7 as well as ATM 3GPP signaling (RANAP and Q.2630).

The following SIGTRAN standards are supported:

- SCTP support according to RFC 2960.
- M2UA support according to RFC 3331.
- M3UA support according to RFC 3332.
- IUA support according to RFC 3057.

Note:	Refer to the PSTN section for information regarding M2UA and M3UA inter-working with MTP-2 and MTP-3.

1.11 Miscellaneous

- Customer Key – By using the new acSetCustomerKey(), acGetCustomerKey() and acEV_CUSTOMER_KEY_RESPONSE APIs, the User can burn a certain key on the board's flash and read it later. This can be useful, for example, for customers who want to tag their boards with an encrypted key for commercial purposes.
- On TP-1610, ability to read the board temperature using acQueryTemperature() and acEV_TEMPERATURE event.
- Reset Reason information at board start up. This debug event is issued upon completion of board start-up. It informs the User of the type of reset that caused the board to restart.
- Time of day. This feature allows the board to keep the current date and time (day/month/year, hour/min/sec). The User can set the date and time either using acOpenboard or acRemoteOpenBoard - for VoPLib Users (in this case the time is taken from the PC), or via the Web interface. The time is included in board errors and exceptions and Users can also view it via the Web.

Reader's Notes

2 Known Constraints

Note: After loading and burning a CMP file to the boards FLASH memory, Users must reload all other downloadable data files (Call Progress Tones file, Voice Prompts file, etc.). This is due to the fact that in the CMP loading and burning process all the data files that were previously stored on FLASH are erased.

2.1 General Limitations

1. Modem relay is not supported.

2. Note that **MGCP** and **H.248** (MEGACO**)** control protocols cannot coexist on the device at the same time.

3. When the board is loaded through PCI, the MGCP or MEGACO can't be used.

4. (RTCP port) = (RTP port) + 1, must be kept for the incoming traffic, otherwise the RTCP traffic is ignored.

5. HTTP streaming limitations:

   - Only G.711 A-Law, µ-Law and Linear PCM coders are supported for the *.wav* files and only G.711 A-Law, µ-Law coders are supported for the *.au* files

   - Number of simultaneously streaming to the IP side is limited to 120.

   - Number of simultaneously recorded voice channels is limited by Web server's capability.

6. When using one of the PSTN CAS protocols (including raw CAS) it is not permitted to manipulate the cross connects between the PSTN signaling channels and the DSP channels. This Means that in this case the use of acMakeTDMConnection() or acBreakTDMConnection() with SwitchingOption set to acSIGNALING_ONLY or acVOICE_AND_SIGNALING is not permitted.

7. PSTNAUTOCLOCKENABLE board parameter and acPSTNSetClockSourceFromTrunkId() API are relevant only when the board is set to framers and is synchronized to to the network (TDMBUSTYPE=2, TDMBUSCLOCKSOURCE=4).

8. In some cPCI-controlled systems, it is not possible to power up the system without boards, and then add boards later (depending on the specific cPCI BIOS in the Host).

9. Note that even when using clear networks, UDP packets can still be lost due to the ARP mechanism (a UDP packet is lost each time the ARP table aging mechanism causes the entry to be dropped, as a result *ini*tiating a new ARP transaction). This is important when working with TPNCPoUDP. To bypass the problem, Users should modify the ARP behavior to be static instead of dynamic (NT Users should use the arp –s command).

10. Users should note that acOpenRemoteBoard() used with acTrivialOpenRemoteBoardMode operation mode does not configure the board in the same way as when using the acOpenBoard() function. When using acOpenRemoteBoard() in this mode, the only parameter used out of the acTBoardParam board configuration structure is the BoardIPAddr (def*ini*ng the IP address of the remote board). However, Users must provide the full board configuration, as the VoP library still uses these parameters.
    To configure other board parameters, different configuration processes should be used (for example, using BootP/DHCP with AudioCodes' proprietary *ini* file).

11. Note that acOpenRemoteBoard() can be invoked only when the board has completed its boot

sequence. Prior to this, the LAN interface of the board is not functional and the command is not processed. Refer to a source code example of how to use acOpenRemoteBoard() over UDP, ensuring that the board has completed its boot sequence and can be operational.

12. The NetCoder voice coder does not support rates 4.8 kbps and 5.6 kbps on all boards and modules.

13. H.110 clocks automatic Fallback limitations:

    - No fallback from or to Network (PSTN) clock is supported.
    - Fallback from or to NetRef clock is supported for NetRef 8 kHz only.

14. If PCM LoopBack is activated, there is no way to know if a new channel being opened is in LoopBack state or not (DRM-related).

15. DTMF Relay mode is not operational with channels set to PCI transport.

16. This version does not support RFC 2198 (DTMFoRTP is not mandatory in RFC 2833) redundancy mode (that is, if a complete DTMF digit was lost, it is not reconstructed). The current RFC 2833 implementation does support redundancy for inter-digit information lost.

17. It is impossible to test multiple fax relays simultaneously by having all channels hooked to the same input stream (coming from one fax). This is because the echo environment created in this case does not imitate the real echo environment.

18. On the channel parameters, when using FaxTransportType = TransparentWithEvents, the Fax events parameters regarding the side of the fax call (answering or calling) and the number of pages is invalid.

19. Mediation Limitation:

    - Mediation with PCM Coder Transcoding to/from an HBR coder using one DSP resource can be done only if the PCM coder is G.711 A-law.
    - The function acAddDestinationPoint() cannot connect between 2 TDM end points
    - The HBR channel of the Transcoding session can use only multiples of 20 msec RTP packet intervals.
      Hence, on the HBR channel, the SampleBasedCodersRTPPacketInterval parameter should not be changed from its default value.

20. When Continuity Test Tones (COT) and/or Special Intercept Tones (SIT) detectors are enabled, The number of Call Progress Tones that can be configured is reduced. For more info refer to the COT & SIT sections of the VoPLib User manual.

21. The M factor when using EVRC coder can be greater than “1” only when the EnableRFC2658Interleaving parameter is set to “2”.

22. A channel should NOT be reopened during IBS detection. Such an action could cause the detector to send multiple events on same IBS input.

23. The number of channels operating in internal IP loopback mode that can be supported by the board is usually less than the declared channel capacity.

24. Caller ID Detection limitations:

    - The maximal allowed size of the Caller ID message to be detected is according to the following formula:
      [152 – (Number_Of_Parameters * 4)]/2.

- The Check-sum field of the Caller ID detection event is not valid, therefore should be ignored.

25. The following fields are mandatory when generating ETSI Caller ID:

    - Date.
    - Time.
    - All the fields of the ETSI Network Operation Use section.

26. When Fax Transport Type is Transparent with Events, and the Fax session is V.34, the Answering side sends a Modem event instead of Fax event.

27. When CNG detector is not Transparent, a CNG tone received from the TDM can not be detected using the Call Progress Tone detector.

28. Voice Prompt buffer size is 10000 kB for all boards (for example: 20 minutes of voice in the G.711 coder).

29. Limitations regarding trunk configuration:

    - All trunks in one board should belong to the same Trunk Type (either E1 or T1).
    - E1_TRANS_62 cannot be mixed. If used, it must be selected for all trunks on board.

30. If the User configures the board to auto clock mode (using the parameter TDMBUSPSTNAUTOCLOCKENABLE), the trunk that is set as the local reference (using the parameter LOCALREFERENCE) must be connected at the board startup.

31. No more then 4 ISDN variants can be configured simultaneously.

32. This version supports T1 ISDN NFAS mode with 1 D-Channel for up to 8 trunks and up to 4 NFAS group per board.

33. On-the-fly trunk configuration is not applicable on trunks configured as SS7, V5 or NFAS.

34. SCTP limitations:

    - Up to 5 SCTP ports are supported
    - Up to 8 associations are supported.
    - Up to 200 streams per association are supported.
    - Maximal size of SCTP packet is 4096 octets.
    - Max number of packets stored for retransmission is 210 per association

35. M2UA limitations:

    - Textual Interface Identifier not supported.

36. When using MGCP, the color indication of an open or close channel in the channel status page in the Web is wrong; (the textual information is correct).

37. Only one SNMP manager can access the boards/modules at one time.

38. SNMP-Trunk MIB - The Trunk MIB is supported per RFC 2495. This version supports only the fields in dsx1ConfigTable Host API Related limitations.

39. Host API Related limitations:

- The VoPLib file structure was changed. Major efforts were made to preserve backward-compatibility. However, still some incompatibility with the previous versions of the library is possible, especially if some internal VoPLib definitions are used by the customer's application.
- The acFullConfigurationModeWithExternalIniFile is not supported.
- Red Hat 8.0 Linux is supported for PCI boards only (TP-260, TP-260/UN).
- All others Linux kernels that are compiled with GCC 3.2 are not supported in PCI mode (supported in TPNCP mode).
- PCI mode in Solaris 9 is not supported.
- The PTHREAD_MUTEX_RECURSIVE should be enabled also in Linux.
- Some compilation warnings might exist.
- 64-bit PCI data transfer on TP-260/UN is currently not supported (Software limitation).
- UNIXWARE specific changes that were in ver 4.0 are absent in the ver 4.2 beta.
- C_COMPAT_MODE shell is not presented in the package.
- When using the internal VoPLib TFTP server, ensure the ErrorHandler callback function that you supply is a reentrant function (as the TFTP server uses this function from multiple threads).

40. MEGACO Limitations:

- MEGACO support in relevant hardware is limited to 30 call set-ups per second.
- ModemDescriptor and MuxDescriptor are not supported in this version.
- The auditCapabilities reply does not return the parameter values.
- Version legality is not checked.
- Call Agent legality is not checked – every request is performed.
- Signal list can have up to 30 signals.
- When opening a context with one RTP termination to play Announcements / CallProgressTones to the network, it should start in send/receive mode. Modification of the stream state currently doesn't work.
- Size Limitations:
  - Maximal command length = 6000 bytes
  - Maximal number of transactions per message = 10
  - Maximal number of actions per transaction = 150
  - Maximal number of command requests per action = 10

- Maximal number of command replies per action = 100
- Maximal number of signals in a signal descriptor = 1
- Maximal number of signals in a signal list descriptor = 30
- Maximal number of reported signals = 50
- Maximal number of events in an event descriptor = 16
- Maximal number of reported events = 50
- Maximal number of observed events = 10
- Maximal number of event parameters = 5
- Maximal number of audit return parameters = 100
- Maximal number of DigitMap alternatives = 32
- Maximal length of DigitMap = 150
- Maximal dial string length = 30
- Maximal length of DigitMap name = 30

3 Version History

Details of previous releases can be found in the Release Notes of Version 4.0 Document # LTRT-00698, published by AudioCodes on Oct-21-2002.

US: AudioCodes Inc, 2890 Zanker Road, Suite # 200, San Jose, CA 95134.

Tel: 408-577-0488 - Fax: 408-577-0492

International: AudioCodes Ltd, 4 HaHoresh Street, PO Box 14, Yehud 56470, Israel.

Tel: +972-3-539 4000 - Fax: +972-3-539 4040

China: AudioCodes Ltd, Room 2811-12, JingGuang Center,

HuJiaLou, Chaoyang District, Beijing 100020, P.R.China

Tel: +86-10-6597-4134 - Fax: +86-10-6597-4260

Japan: S Building B1F, 2-30-2, Chuou, Nakano-ku, Tokyo 164-0011, Japan

Tel: Fax: +81-3-5348-1405 - Fax: +81-3-5348-1405

Original printed on recycled paper and available on CD or Web site

CERTIFIED QMS SI ISO 9001:2000 THE STANDARDS INSTITUTION OF ISRAEL