

CTI32 .Net Version Release Notes  
V4.6  
8 December 2008

\*Note: If you install a new version of the CTI32Engine, you may have to re-compile your application since the structures may have changed that are passed in to your class library.

#### **Version 4.6 (8 December 2008)**

Please go to this link for information in this release;  
[http://support.inventivelabs.com/index.php?title=Release\\_Notes\\_For\\_V4.6](http://support.inventivelabs.com/index.php?title=Release_Notes_For_V4.6)

#### **Version 4.51 (26 October 2007)**

- Improved channel recycling in the .NET Engine.
- When playing Vox8Mulaw file the system will ensure any wave headers are skipped if a wav file is supplied for input to the method.
- Improved handling of the drop call and release call mechanism.
- Enhancements to global tone detection.
- Changed the order of the SDP attributes in SIP calls that use 3PCC.
- Corrected a problem where DNI and DM3 boards utilizing global call would fail to transmit the dialed number to the switch.

#### **Version 4.50 (15 September 2007)**

- Added SetStdCSPCallback method to allow the user to stream the CSP data back to the application during speech recognition.
- CTI32Config now has a controlled stop option. This allows the system to complete existing calls before completing the shutdown process.
- CTI32Engine users can now subscribe to an event to be alerted that the system is shutting down. Use the event EngineStateChange.
- The Engine has a new flag to determine if the system is running as a service. (IsRunningAsService)
- The array FaxPorts is now obsolete at the engine level. Fax resource pooling is now managed by CTI32.dll.
- The GetFaxResource and FreeFaxResource commands are now overloaded. The non-parameterized versions are still functional, so long as you are not using HandleReqModifyCall in the INI file. The new overloads provide for non routable fax boards and for T.38 SIP functionality.
- You can now shut down the system programmatically. Use the method StopService to cause the service to shutdown.
- Voice Resource Pooling – You can now specify UseResourcePooling without using AutoConfiguration. This allows you to pool voice resources in systems that are mixed SIP and TDM.

- AnalogRingCount is now configurable in CTI32Config. See the default section. When using analog boards, you can set this value to delay the system from answering the call until the rings count is met.
- The system no longer requires voice resources for fax only applications. To do this, remove the OpenName in the Configuration for each channel. The system will not attempt to attach voice resources to the calls. (This will not work on analog boards.)
- CustomBoardSetupCode. The system will attempt to locate and call a method called BoardSetup in the DefaultDLL. If it exists, the engine will call the method during startup. This allows you to set specific board parameters at open time.
- Tmonitor has been modified to allow the user to specify the outbound caller id during call testing (SIP & PRI Only). In a SIP setting, you can also set the entire From header by specifying the From header exactly as it should appear on the INVITE.
- MSI Station Routing – The library now takes into account the board the MSI station is on when it is routed or added to a conference.
- Contact headers are corrected to ensure the BYE messages are received properly in a SIP setting.
- Corrected SIP authorization when using custom from headers and the gateway asks for credentials.
- Added code for Persistent Header SIP Support (SU 150 and up). The application should still use SIPHeadersAdd function to add the headers to the call.
- Corrected problems in routing IP media channels when the system is in 3PCC.
- Corrected CTIDial to trim trailing blanks when placing SIP calls.
- PlayTTSAndRecognizeWord now has wide string overloads to allow the user to supply wide string text to the TTS engine.
- PlayTTS now provides a voice parameter to allow the caller to switch the voice during the TTS call.
- CTI32 now will handle RE-INVITES offered by endpoints in 1PCC mode. To allow this, set the HandleReqModifyCall=1 in the INI file. If you use this for faxing and T.38, you must use the new overloaded versions of GetFaxResource() in the engine.
- The default fax mode in CTI32.INI is now R4Fax.
- Added a CTI32.INI setting to allow the PROCEEDING event to start call progress. Set AllowProceedingToStartCPA=1 in the INI file.
- Added SetEosDelay/GetEosDelay to set/get the end-of-speech timing for voice recognition modules.
- The SipSignalingPort can now be specified for systems with multiple NIC cards. This tells HMP which IP address to use for SIP signaling.
- Added functionality for custom SIP headers. 512 Max Length. For example, the following header can be set using the SipHeadersAdd method:

```
Remote-Party-ID: "John Doe" <sip:jdoo@foo.com>
```

- Added the ability to specify user IO routines for playing and recording files. The user must first set the UIO callback addresses. The user can then call CTIPlay or CTIRecord and the system will call back to the user-defined routines to get the data to play.
- Added support for LumenVox Lite. In the CTI32 ini file, you can set the license type by LicenseType=SpeechPort or VoxLite
- Added ForceCoInitializeEx to the [TTS] section of the .ini file. Set the value to 1 to force the use of CoInitalizeEx instead of CoInitialize. Set the value to 0 (the default) to continue using CoInitialize.
- Modified CTI32Config to allow the user to specify larger HMP port quantities (1024).
- Added CCDropCallWaitTime to the .ini file to allow the user to override how long to wait for a cc\_DropCall to complete. The default is 60 seconds. The minimum is 5 seconds. This new parameter is in addition to the GCDropCallWaitTime (global call version) added in release 4.10.
- Added DnisExtDLL, DnisExtType and DnisExtMethod to the .config file to support a DNIS DLL that is independent of the DefaultDLL. If this is specified the DnisMethod parameter is overridden, and the DnisExtMethod is called for DNIS lookup.
- Fixed problem in PlayNumber and PlayNumberSpanish and PlayOrdinal if length was larger than 10 digits.

#### **Version 4.10 (4 December 2006)**

- New support for third-party-call control while initiating and receiving SIP base calls. See section 13 in cti32.dll.doc and the HMP section in the CTI32Guide.
- Corrected a 30 second limitation on recording DM3 channels while using the TMonitor recording feature.
- When reporting errors via email (this must be configured in the .config file) the system now includes the server name of the failing machine in the body of the email.
- Added GlobalCallProtocol tags to each individual channel. This allows the system configuration to allow DM3 and SIP calls simultaneously. If the individual channel does not specify a GlobalCallProtocol tag then it automatically defaults to the GlobalCallProtocol value in the default section of the .config file.
- Added UseAutoConfiguration to the cti32engine.config file. Valid values are 'true' and 'false'. True specifies that you want the system to ignore the current boards section of the .config file and scan for boards and devices. False (the default) indicates to use the current boards section of the .config file. This only works if you are using Dialogic drivers SR6 or HMP. If you are using autoconfiguration with HMP, you must explicitly set the number of SIP ports you have licenses for in the .ini file. See the SipMaxCalls parameter.
- Added UseResourcePooling to the cti32engine.config file. Valid values are 'true' and 'false'. True indicates that you wish to pool your voice resources. False indicates that you want to keep a 1-to-1 assignment of voice resources to channels. See the CTI32Guide for more information.

- The Type of Service (TOS) byte in outgoing SIP packets is now set to an RTP payload. Some routers give priority to RTP streams, if they are marked as such.
- Dial strings passed on a CTIDial command are now trimmed for leading and trailing whitespace.
- Added CTISipReinvite. This function allows the caller to redirect the RTP stream of an established call to a different location. See CTI32.dll.doc for more information.
- Added CTISipRefer. This function allows the caller to refer an existing SIP call to another SIP device. See CTI32.dll.doc for more information.
- Added CTIGetSipStats. For HMP installations, this allows the caller to get information concerning a currently connected SIP call. The function will tell the caller the current codec in use, the IP address and port the call is connected to, and the SDP (session description protocol) that was sent from the remote station at the time the call was established. See CTI32.dll.doc for more information.
- Added CTISetFaxMode. This allows the caller to specify the fax mode to place the fax device in. The allowable mode values are: 0 (FAX\_MODE\_V17) and 1 (FAX\_MODE\_T38.) The default mode is 0. Note that FAX\_API in the .ini file must be set to 3 (Manual mode) to use this function. See CTI32.dll.doc for more information.
- Added TrueSpeech Format for playing and recording of files.
- Added GetOutboundHandle overload. The method now accepts a string to tell the engine what protocol type to search for. Valid values are "SIP" and "DM3".
- Corrected two-port and four-port licensing issues when opening boards in addition to the individual channels.
- Enhanced the recovery for errant SIP calls that do not ACK the BYE message sent from the server.
- Enhanced the codec support for SIP calls.
- Added IP Media Pooling for SIP calls in 3PCC mode. (See cti32.dll.doc).
- Added SipMaxCalls to the .ini file for systems with more than 120 ports. The default is 0, telling HMP to autoconfigure up to 120 ports. If you have more than 120 ports, you must specify the correct number of licensed ports here. Also, if you are using autoconfiguration in the CTI32Engine, you must set this value to the number of ports you have even if it is less than 120.
- Added Contact to the .ini file to allow the SIP messages to contain a Contact header. If left blank, a default contact header is manufactured by the SIP stack.
- Added SipSignalingIp to the .ini file to allow configuration of the IP address to use for the RTP streams. (For use by systems with multiple NIC cards.) If left blank the system picks the NIC card for RTP traffic.
- Added LogEventsStats to the .ini file to allow the user to see periodic reports of how the system is handling events signaled by the Dialogic drivers. The default is 0, for no logging. 1 turns on the logging.
- Added EventsThreadPriority to allow the user to set the priority of the events thread. The default is 0 for normal priority. 1 give the event thread a slightly higher priority. 2 gives the highest priority.

- Added GCDropCallWaitTime to the .ini file to allow the user to override how long to wait for a gc\_DropCall to complete. The default is 60 seconds. The minimum is 5 seconds.
- Various changes as requested by customers

#### **Version 4.09 (9 September 2006)**

- Corrected startup error when the dialogic libgc.dll is 5.1.1 SP 1 or older.
- Corrected the mapping of cause codes returned by the Dial method that occur during a GCEV\_DISCONNECT in a DM3 setting.
- Call progress template values are now specified in the cti32.ini file. The templates are automatically applied when a Voice Resource Board is opened. Voice Resource Boards are automatically opened on analog cards. However, on ISDN or T1 lines you must now specify a 1 on the UseQualificationTemplates value in the config file to force the Voice Resource Boards to open.
- GetSilence now has an overloaded method that allows the caller to specify how long the call waits for the silence period. GetSilence without this additional parameter waits for 30 seconds by default. GetSilence now returns a TermCode instead of an integer. If the maximum wait time expires, TM\_MAXTIME is returned.
- Added new method MonitorSilence. Monitor silence allows the caller to determine how much silence is found during the interval.
- Corrected line lockout when channel was being recorded.
- Enhanced the Engine to check for failures opening channels, resources and boards. Engine Stops if errors are found.
- Set cause values and cause messages in CTIDial for non-gc calls.
- Increased the CTI32Engine GlobalStatus Name/Value pairs to 200. Also, added error logging to trap attempts to reference Name/Value pairs out side of array boundaries.
- Added overloads to Play, Record, PlayNumber, PlayNumberSpanish, PlayOrdinal, PlayDate, PlayDateSpanish, PlayTime, PlayTimeSpanish, PlayChars, RecordConversation, PlayandRecognizeWord, and SetComfortCSP to allow for playing and recording directly from memory buffers.
- Added 'Exec' method to the Db class in the Engine. 'Exec' calls stored procedures that take parameters. The first argument is the name of the stored procedure, and the second argument is an object that MUST be a SqlParameter array, an OdbcParameter array, or NULL if no parameters are needed for the stored procedure.
- Overloaded the GetFirst method to the Db class in the Engine. This method allows the user to pass a stored procedure with or without parameters to fetch a rowset. The first argument is the name of the stored procedure, and the second argument is an object that MUST be a SqlParameter array, an OdbcParameter array, or NULL if no parameters are needed for the stored procedure.
- Fixed parsing of DNIS/ANI on T1 digits. If there are separation digits such as \* or # and the ANI/DNIS is missing, it will align on the separation digits and properly get the other value.

- Added SkipDriverLoadedCheck to cti32.ini file. If set to 1, on startup, the dialogic driver will not be tested to see if it was running. (call to NCM\_GetDlgSrvStateEx is skipped)
- Added the MapNetworkDrive class to the CTI32Engine that simplifies the mapping of network drives from code. See Playing Files from a Network Drive or Mapped Drive in the CTI32Guide.

#### **Version 4.08 (26 July 2006)**

- Enhanced TMonitor Dial-Out Function to allow user to set call progress value for the call, remember the last number entered, and to allow sending a fax sample
- Enhanced TMonitor to allow user to busy out a line
- TMonitor status now shows:
  - Last Call Result
  - Last Cause Codes (ISDN)
  - Last Fax Result
  - Last Answer Length
  - If the line is being recorded
  - If the line has been manually busied out
- Enhanced TMonitor to record ports and stream recordings to TMonitor in real-time
- The CtiConnectionStart method in CTI32StatusClient is now overloaded. The new overload accepts an additional delegate that provides status messages from the engine and/or user code.
- Text To Speech now accepts embedded tags to control various aspects of the TTS engine. See: [http://msdn.microsoft.com/library/default.asp?url=/library/en-us/SAPI51sr/Whitepapers/WP\\_SimpleTTS.asp](http://msdn.microsoft.com/library/default.asp?url=/library/en-us/SAPI51sr/Whitepapers/WP_SimpleTTS.asp) for more information
- Added FAX\_USEEXPLICITOPEN and FAX\_EXPLICITBOARDS to the CTI32.ini [FAX] section. These settings instruct the application to search only the specified fax boards for fax resources. For more information, see cti32guide.doc under the fax ini settings section.
- Added GetDigitsLcoeffOption to the CTI32.ini [GENERAL] section. Option gives greater control of when and if GetDigits returns LCOFF in the return value of the call. For more information, see cti32guide.doc under the general ini settings section.
- Minor bug fixes

#### **Version 4.06 & V4.07 (9 June 2006)**

- Changes to make compatible with VS2005
- Minor bug fixes as requested by customers

#### **Version 4.05 (1 Feb 2006)**

- Fixed a problem with the daily stats. Sometimes the call record was not being written on a T1 line (usually single wink system) because the call record was not being initialized properly on the OFFHOOK because the status was WINK rather than RING.

- Small change to the stats so that dnis/ani is saved when on ONHOOK rather than OFFHOOK
- Added all ISDN dialing call block options to CTI32.INI file
- Added ISDN\_AcceptCall method
- Worked on Glare conditions on high density dialing application
- Cached DLL's so you don't have to start / stop service during development. Added a section to CTI32Guide.
- Worked on Speech Reco (LumenVox) with HMP / DM3. There was a compatibility issue running under DM3.
- Added comfort tone to speech reco.
- Tested with HMP 2.0
- Added new dialing call templates and new section in CTI32 Guide
- Fixed HMP call progress return codes for BUSY or NO ANSWER
- Changed Outbound sample to read the XML file differently
- Added PlayTTSAndRecognizeWord which is a TTS version of the voice reco function
- Added native support for SQL Server in addition to ODBC support
- Added section in CTI32 Guide talking about the DNIS logic which allows you to load multiple concurrent modules based on the dialed phone number. Share your box across multiple application.

#### **Version 4.04 (9/26/2005)**

- Added PlayTTS function. You must have the Microsoft SAPI engine installed for this to work. (See the CTI32 Guide for information on how to do this)
- CTI32config not refreshing right after "Reconfigure board"
- Added PbxCallInfo and PbxDisplay methods for the Dialogic PBX boards
- Added PBX LineType to the engine.
- Resetting of line on ISDN-PRI if Dialogic times out on Line Dropped
- Additional options for T1 lines. If mask = CUSTOM then engine does not collect digits nor does it answer the call. <SW> in mask means Single Wink. <IS> is immediate start. Using <SW> or <IS> you can optionally set D=Dnis digit or A=Ani digit. (i.e.) <SW>DDDD would wink, collect 4 digits then answer call.

#### **Version 4.03 (August 2005)**

- Cleaned up old code for HMP prior to 1.3
- Code for T.38 Fax
- Added AppendToWave()
- Added rawDigits to ChannelData structure. This is used to store the raw digits coming in on a T1 call (contains DNIS/ANI)
- Made DNIS/ANI buffers larger for VoIP strings.
- Changed DnisLogic to pass ChannelData structure
- Added algorithm overload to cfg.GetOutboundHandle(). 0=Top down 1=Bottom up 2=Most Recently Used (as long as idle at least 2 seconds)

#### **Version 4.02 (7/13/2005)**

- Added SetChannelState to configuration. If set to true, it will set the ISDN channel state to INSERVICE when the system comes up. With certain ISDN protocols, this locks the system.
- Updated HTML help in CTIConfig utility.
- Added callerIDName to ChannelData structure. Modified engine to load this on PRI or GlobalCall.
- New CallingCard sample in VB
- New IncomingFax sample in C#
- Big item: Support for R4 Fax API – set cti32.ini file FAX\_API=1

#### **Version 4.01 (7/6/2005)**

- Many Documentation changes
- Missing voice file in Dial Test in TMonitor
- New MakePrompts sample in C#

#### ***New Features in Version 4.0 (7/1/2005)***

- Support for Intel HMP 1.3 including SIP registration
- New pricing and support plans
- New Software License
- Separate Source code version. Engine source code included in regular release.
- VoIP support via HMP
- New support web site
- New .NET documentation for CTI32NetLib and CTI32Engine classes
- New Getting Started document
- Additional examples
- Support for GlobalTone
- Additional support for Voice Recognition using CSP and LumenVox
- Support for Amtelco XDS conference board
- CTI32 configuration utility
- Improvements to Dial and call progress (support for all CAP parameters)
- Spanish numbers and dates
- New client support so that applications can get information from the CTI32 service.
- Support for DM/V dialogic boards