SynCTI Upgrade Instruction

CTILinux Ver. 5.1.0.0

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New Boards

Series	Feature	**Board Model**	New Feature	Note
SHN	VoIP	**SHN-8B-CT/PCI+** **SHN-16B-CT/PCI+** **SHN-32B-CT/PCI+** **SHN-60B-CT/PCI+** **SHN-120B-CT/PCI+**	1) PCI 2.2 bus support 2) DMA read and write support 3) Integrated LAN 4) Processing of network protocols in hardware 5) Easy upgrade of firmware 6) Multiple programming modes support 7) Various VoIP CODECs support 8) Barge-in functionality 9) Highly efficient, real-time call control and voice processing 10) Unified SynCTI driver development platform	None

DST	Passive tapping digital station/PBX extension	**DST-24B/PCI** **DST-24B/PCI+**	1) High-impedance recording of digital phone lines through parallel connection 2) A variety of ways to start/stop recording 3) Supports simultaneous recording on 24 channels, each with a different format 4) Supports independent-recording of incoming, outgoing and mixed-recording modes 5) ANI and DNIS support 6) Synchronous acquisition of the information displayed on digital phones during recording 7) Detects all modes of keying supported by user phones 8) Activity/silence detection 9) Automatic Gain Control (AGC) support in recording operation 10) Call progress monitoring 11) Automatically checks board to see if modules are correctly inserted and to determine the number of modules on the board 12) Supports detection and alarming of line faults, including line break and errors in voltage level, signal-to-noise ratio and synchronization 13) Switches flexibly between voice channels B1 and B2 14) Besides, the DST-24B/PCI+ board supports MS-GSM, G.729A and MP3 (8K & 16K) encoding in hardware	None

		DST-24B/PCIe **DST-24B/PCIe+**	1) Developed with the design of PCIe X1, support PCIe X1, X2, X4 and X16 slots 2) High-impedance recording of digital phone lines through parallel connection 3) A variety of ways to start/stop recording 4) Supports simultaneous recording on 24 channels, each with a different format 5) Supports independent-recording of incoming, outgoing and mixed-recording modes 6) ANI and DNIS support 7) Synchronous acquisition of the information displayed on digital phones during recording 8) Detects all modes of keying supported by user phones 9) Activity/silence detection 10) Automatic Gain Control (AGC) support in recording operation 11) Call progress monitoring 12) Automatically checks board to see if modules are correctly inserted and to determine the number of modules on the board 13) Supports detection and alarming of line faults, including line break and errors in voltage level, signal-to-noise ratio and synchronization 14) Switches flexibly between voice channels B1 and B2 15) Besides, the DST-24B/PCIe+ board supports MP3 (8K & 16K) encoding in hardware	None
DTP	T1/E1/J1 passive tapping	**DTP-120C/PCIe** **DTP-120C/PCIe+** **DTP-60C/PCIe** **DTP-60C/PCIe+** **DTP-30C/PCIe** **DTP-30C/PCIe+**	1) Use of PCIe X1 interface 2) DMA transfer support 3) Supports DTMF detection 4) Supports tone detection 5) Supports Barge-in detection 6) Supports Automatic Gain Control (AGC) 7) Supports A-Law, μ-Law, ADPCM, GSM, G.729A, MP3 8K and MP3 16K encoding in hardware (note that only the boards herein with + support GSM, G.729A and MP3) 8) Supports E1-SS1, E1-ISDN, E1-SS7, T1-ISDN monitoring 9) H.100 bus unsupported 10) Echo cancellation, voice playback, voice mixing via bus unsupported	None

New Features & Fixed Bugs for Other Boards

Series	Feature	**Supported Board Model**	New Feature & Fixed Bug
SHT	Analog Voice Boards	All SHT series boards	The 16B boards optimize the A-Law, μ-Law encoding algorithms to prevent DSP exception.
			Add a new function RingEndDtrTim to set the maximum durations of the ringing current at on and off states.
			Fix the bug that analog voice boards cannot accurately receive FSK calls started from overseas.
			Solve the problem that one end of a station channel on 16B analog boards was probably blocked.
			Improve the 8B boards' capability of monitoring other boards.
		USB voice boxes	Eliminate the Blue Screen of Death caused by unprotected empty pointers within the USB driver.
			Fix the bug that USB voice boxes cannot accurately detect dual tones or low-energy tones.
			Fix the bug that USB voice boxes cannot accurately call back in double-buffer recording and playback.
			Improve the capability to receive DTMF.
SHD	Digital Trunk Boards	A-type and D-type	Support LoopBack testing.
	SS7	Digital trunk boards which support SS7 signaling	Fix the bug that the local end of an SS7 channel was unable to be idle while the remote end was sending CGB messages.
			Enable the function SsmSetTupParameter to set the calling party's category field in the IAI/IAM message.
			Fix the bug that the driver will return messages from those unused links while multiple signaling links are in use at a same time.

			Fix the bug that Test.exe reports an error: 'visit illegal address' in case the function SsmCheckIsdnMsu is invoked when the configuration item AutoHandleISDN is set to 1.
			Fix the bug that the number of PCM, if greater than 100, cannot be recognized by the driver based on the ISUP protocol. Now, the maximum number which can be recognized becomes 256.
			The driver newly supports the manual handling of MSU at SS7 MTP2.
			MSUDECODE.EXE supports the decoding of those messages made of 14-bit point codes.
			Solve the problem that SsmSendSs7MsuEx might send uncompleted messages in ISUP.
			Make an improvement that when an SS7 channel receives a GRA message, the corresponding channel will turn into the idle state, not affecting any other channels.
			Solve the problem that the driver could not recognize the GRA with the range of 31.
			Fix the bug that the signaling link number was incorrect when customizing IAM, ACM messages in SS7/ISUP with multiple link groups.
	E1-ISDN	Digital trunk boards which support ISDN signaling	Fix the bug that the CallerID and CalleeID buffers were not emptied when the ISDN channel went into the idle state.
			Fix the bug that when the network side disconnects an ISDN call first, the reason for why the user side shows suspended is incorrect.
			Fix the bug that the ISDN channel could not be left idle due to repeated calls of the function SsmHangup.
	T1-ISDN	Digital trunk boards which support ISDN signaling	Change the value of the fifth 8-bit block in the ISDN Bearer Capability Information Element from A1 to A2.
	SS1	Digital trunk boards which support SS1 signaling	The driver newly supports the LineSide (OPS-FX) protocol.

			Fix the bug that the channel would go into the pending state while it was disconnected from a call in LineSide.
	Channel Bank	Digital trunk boards which support channel bank	Fix the bug that sometimes rings could not be stopped after the large-capacity channel bank sent rings to 20 channels at the same time.
	FSK	Digital trunk boards	Improve the capability to receive FSK.
ATP	Analog Tap Passive Boards	ATP-24A/PCI+	Fix the bug that some data might be lost during the GSM recording in hardware.
		ATP-24A series boards	ATP-24A and DST-24B/PCI series boards support the new configuration item DspCoder to replace the old ldr531.
			ATP-24A series boards newly support the AGC feature in hardware.
DTP	Digital Trunk Passive Boards	All DTP series boards	Support the monitoring in CorNet, can acquire correct call status as well as calling and called party numbers.
			Add the feature to detect which party hangs up the call.
			Enable the function SpyGetCallInCh to output failure information.
			Fix the bug that 30B digital trunk passive boards recorded silence or noises in G.729A in T1.
		SHD-60B-CT/PCI/FJ SHD-30B-CT/PCI/FJ	Fix the bug that Blue Screen of Death occurs when using the SHD-60B-CT/PCI/FJ board in a multi-board system involving multiple progresses.
			Fix the bug that call states cannot be monitored by using ISDN TI.
			Fix the bug that the function SsmQueryRecFormat returns wrong values during the GSM recording in hardware.

			Fix the bug that the DTP board cannot acquire a complete called party number in ISDN monitoring.
DST	Digital Station Tap Boards	All DST series boards	Improve the parse of displayed messages on the Alcatel 4049 operator.
			Fix the bug that when a DST series board is working with the Nortel PBX, the connected phone goes wrongly into the idle state after it is picked up and the PBX turns to a ringing state from idle.
			Fix the bug that while working with the Alcatel 4400 PBX and the phone 4049 used in it, a DST series board fails to get the calling party number and the LCD information.
			Fix the bug which we find by using debugview to record remote signaling messages that the user channels always stay in a state of idle when they use a DST series board with the Sup60e Switchboard.
			Update the NORSTAR.BIN file to fix the bug that a DST series board fails to detect the initial RELEASE BUTTON event.
			Fix the bug that a DST series board fails to get the accurate LCD information while working with the Ericsson 4224 PBX.
			Fix the bug that the channel fails to be reset with the idle state when using a DST series board to monitor Toshiba CTX.
			Fix the bug that when a DST series board is monitoring the Tadiran PBX, what it records are all noises.
			Fix the bug that when a DST series board is working with the Siemens Hipath 4300 version 2 PBX, the on-board channel improperly stays in a ringing state as a new call comes after the previous one is forwarded by the monitored phone.
			For the use of ZXJ10 PBX, add a small tail to the end of FSK data.
			Optimize the support of the Panasonic PBX and the phone T7633.
			Shield the repeated events on the digital phone line.
			Fix the bug that E_RCV_DTR_DKEY had no output while the board was working with Ericsson MXONE, Dialog4222.

			Better support the state machine of the 302C operator of the Avaya 8400 PBX.
			Better support the state machine of Nortel Meridian1 61/3905.
			Fix the bug in message display of the Alcatel digital phone.
			Fix the bug that some events were lost while using the Alcatel digital phone.
			Add a D-channel state machine of the Harris PBX.
		SHR-16DA-CT/PCI	Support working with the 4059 PC operator of the OmniPCX 4400(OXO/OXE) PBX.
		DST-24B/PCI	Better support Nortel 61C and the phone 3905.
			Fix the bug that DST-24B/PCI was not supported by the function SsmStopListenTo.
			DST-24B/PCI and ATP-24A series boards support the new configuration item DspCoder to replace the old ldr531.
FAX	Faxing	All fax boards	Fix the bug that the fax board could sometimes not receive the training acknowledgement.
			Shorten the time to receive a fax page to increase the possibility of succeeding in training under the rate of 9600.
			Improve the compatibility of the fax board to the DIALOIC board.
			Fix the bug that sometimes the faxing system goes into an illegal state.
		SHD-30C-CT/PCI/FAX SHD-60C-CT/PCI/FAX	Enable these two board models to be configured with 32 fax channels.

Newly Added Function	1. Add new functions SsmSetVoiceEnergyMinValue and SsmGetVoiceEnergyMinValue to set and obtain the threshold value of Barge-in detector for noise judgment.
	2. Add a new function GetPciBoardSerialNo to replace the old GetPciBoardSerailNo.
	3. Add a new function SsmGetMaxSs7link to get the total number of configured SS7 links in the system.
	4. Add a new function SsmGetSs7Mtp2Msu to query if the buffer area for MTP2 MSU reception in the driver has received any MSU message; if has, take out the first received message.
	5. Add a new function SsmSendSs7Mtp2Msu to send MSU messages at MTP2 layer on a specified SS7 link.
	6. Add a new function fPcm_Pcm8ConvertGSM to convert the 8 bit PCM formatted file to the GSM formatted file.
	7. Add a new function SsmDstSetFlag to set control properties of channels on the DST series board.
	8. Add new functions SpyGetCallerType and SpyGetCalleeType to respectively obtain the calling and called party number types of the current call.
	9. Add the value 4 to nType in functions SsmSetIsupFlag and SsmGetIsupFlag to set and get the TMR (Transmission Medium Requirement) parameter in the IAM message.
	10. Add a new function SsmSetRingPeriod to set the on and off durations for the ringing current generator on station channels.
	11. Add a new function SsmSetTxRedirectingNum to set the redirecting number in the call setup message in ISUP.
	12. Add a new function SsmSipGetBoardRegStatus to get the status returned when the SHN board sends the Registration request for the last time.
	13. Add a new function SsmSipGetChRegStatus to get the status returned when the channel sends the Registration request for the last time.
	14. Add a new function SpyGetConId to obtain the connection number of the current call.
	15. Add a new function SsmGetUserInfo to obtain the content of the User-User message unit.
Newly Added Configuration Item	1. Add a new configuration item DefaultVoiceFormat under the section [BoardId=x] to set the voice data encoding format of the B-channel on digital lines.

	2. Add a new configuration item LocationNumber under the section [ISUP] to set the LocationNumber information field in the IAM message.
	3. Add a new configuration item Mp3IsOnlyRead under the section [SystemConfig] to set the property of the recorded MP3 files (read only or not).
	4. Add a new configuration item loopback under the section [BoardId=x] to set the loopback feature of trunks, used for diagnoses or debugging.
	5. Add a new configuration item ISDNProtocolType under the section [SpyPcm] to set the support of the CorNet protocol for ISDN monitoring.
	6. Add a new configuration item DefaultIAM_RedirectingNumber under the section [ISUP] to set the first two bytes of the redirecting number in the IAM message, including the nature of address indicator, numbering plan indicator and address presentation restricted indicator.
	7. Add a new configuration item PlayFilterFlag under the section [BoardId=x]. If voices are played in great volume, the DTMF detector probably fails to acquire the correct DTMF digits sent from the remote end. In such situation, the filter should be enabled to avoid the effect of the great volume on DTMF reception. This configuration item just determines whether to enable the filter or not.
	8. Add a new configuration item AppHandleMtp2Msu under the section [SS7] to set the way to handle SS7 MTP2 MSU.
Configuration Program	1. Fix the bug that the configuration of FilterFskCallerid made by the configuration program is incorrect.
	2. Fix the bug that Blue Screen of Death appears when the D-type digital boards are configured with monitoring modes.
	3. Fix the bug that the configuration program fails to find A3 upon the start when it is working with the 2003 terminal server.
	4. Support the national standard of tone configuration.
Board Startup	1. Fix the bug that the system shows illegal upon invoking SsmStartCti for more than one time within a single progress.
Play & Record	1. Fix the bug that the time for the last time callback of SsmRecFileA is incorrect.
	2. Fix the bug that that the wav header in G.729A is not standard.
	3. Fix the bug that there appear noises while playing back the first package of data in the double-buffer mode.

	4. Support of memory recording and playback as well as double-buffer recording and playback, in PCM8.
	5. Improve the quality of the alaw file converted by the function fPcm_MemAdpcmToALAW.
	6. Fix the bug that SsmPlayMemBlock cannot obtain the stop flag as it sets only one buffer.
	7. Add a new function fPcm_GSMToMp3 to convert the GSM formatted file to the MP3 formatted file.
	8. Add a new function fPcm_Pcm8ConvertGSM to convert the 8 bit PCM formatted file to the GSM formatted file.
Monitoring	1. Improve the capability of a single channel to monitor multiple channels.
FSK	1. Enhance the capability of all voice boards in FSK reception, eliminating misdetections.
Installation Package	1. Fix the bug that the system goes halted while installing the package in Windows 7 RTM.
	2. Fix the bug that the shortcut for the start menu stays invalid after the driver is properly installed.
	3. Fix the bug that the driver for digital station tap boards fails to be installed in VISTA 32 operating system.
	4. Add the software tool CasTool.exe and relative documents for digital station tap boards.
Log File	1. Fix the bug that the function call information is not printed into the log file when the function SsmWaitForEvent returns -1.
	2. Fix the bug that the API function SsmRectoFile is not output to the log file in case the configuration item EnableDebugAPI is enabled.
Demo	1. Fix the bug that there appear irrecognizable characters on the interface of Test.exe in English operating system.
	2. Fix the bug in loop playing of Test.exe.
	3. Enable the RecToFile demo program to newly support the C# environment.

About Synway

Synway specializes in designing hardware/software building blocks for use in Computer Telephony Integration (CTI) applications, such as IVR, Call Center, Recording, Unified Messaging and Value-Added Service (VAS) in both PSTN and IP environments. Our products feature rich media processing resources including Fax, conferencing, Codecs, echo cancellation and call control with an array of signaling capability for SIP, SS7 packets, ISDN and CAS in worldwide IP/T1/E1/Analog networks.

In the past two decades, Synway has attracted CTI solution providers through superior service, field-proven products and matchless pricing for the highly reliable Telco and enterprise communications platform. With internally coordinated Maximal Support Value (MSV) System, our service engineers provide pre-sales consulting, development support, and after-sales service, so you can monitor the service process effortlessly. Synway's competitive pricing will ensure your matchless competitiveness in any project bidding or distribution-channel expansion globally. Our products have been applied by hundreds of customers worldwide.

Synway marketing

Tel: +86-571-88860561

Email: market@synway.net

Sales Department

Tel: +86-571-88860561/88864579

Fax: +86-571-88850923

Email: sales@synway.net

Technical Support

Tel: +86-571-88864579

Mobile: +86-13735549651

Email: TechSupport@sanhuid.com or sll@sanhuid.com

MSN: Scycindy_Sh@hotmail.com