Driver (CTILinux 5.0.3.0) Upgrade Instruction

CTILinux5.0.3.0 is improved greatly to enhance echo cancellation and data processing capabilities under Linux OS

Synway, a leading player specializing in designing hardware and software building blocks for use in Computer Telephony Integration (CTI) and VoIP applications, announced add new value-added features in its latest driver version CTILinux 5.0.3.0.

CTILinux 5.0.3.0 has enabled ATP-24A/PCI, ATP-24A/PCI+, ATP-24A/PCIe, ATP-24A/PCIe+ series to support the high density recording and monitoring applications with full-size PCI form factor and PCI-X interface under Linux OS. Another advanced technical breakthrough is included in the ATP series products: Direct Memory Access (DMA) for high-speed data transmission/reception which minimizes the consumption of CPU resource. With this DMA technology, CTILinux 5.0.3.0 has perfectly improved all Synway products data processing capabilities.

Additionally, CTILinux 5.0.3.0 has released brand new SHD-120D-CT/PCI/CAS and SHD-240D-CT/PCI/CAS only for SS1(CAS) and channel banks application. To enhance echo cancellation, Synway's engineers adopt high adjustability, complimentary SuPerForm^{TM} for SHT-16C-CT/PCI/EC, prolonging 8ms to 32ms for time delay estimation. SuPerForm^{TM} is the most rugged, Synway patent-owned echo cancellation technology.

By adding some new features, CTILinux 5.0.3.0 is consistent with SynCTI 5.0.3.0 except for the operating system. For more information, please refer to the following table.

- **New Boards Supported by CTILinux 5.0.3.0**

Series	Feature	**Board Model**	New Feature	Note
ATP	Analog tap passive boards	**ATP-24A/PCI**	1) Support of full-size PCI slot and PCI-X slot; H.100 bus (CT-BUS) unsupported. 2) Support of DMA transfer of recording data. 3) A 50-pin RJ21 connector available on board for line connection. 4) Each board is equipped with 8 analog recording circuits and able to support another 2 recording modules (each module embraces 8 recording channels). 5) Support of a large selection of voice CODECs, including hardware-based A-Law, μ-law, IMA-ADPCM, software-based 16-bit linear PCM, MP3 and VOX.	None
		ATP-24A/PCI+	1) Support of full-size PCI slot and PCI-X slot; H.100 bus (CT-BUS) unsupported. 2) Support of DMA transfer of recording data. 3) A 50-pin RJ21 connector available on board for line connection. 4) Each board is equipped with 8 analog recording circuits and able to support another 2 recording modules (each module embraces 8 recording channels). 5) Support of a large selection of voice CODECs, including hardware-based A-Law (G.711), μ-law, IMA-ADPCM, software-based 16-bit linear PCM, MP3 and VOX; support of either GSM or G.729A for recording.	None

		ATP-24A/PCIe	1) Uses PCIe X1 design, PCIe X1, X2, X4 and X16 slots supported, H.100 bus (CT-BUS) unsupported. 2) Support of DMA transfer of recording data. 3) A 50-pin RJ21 connector available on board for line connection. 4) Each board is equipped with 8 analog recording circuits and able to support another 2 recording modules (each module embraces 8 recording channels). 5) Support of a large selection of voice CODECs, including hardware-based A-Law, μ-law, IMA-ADPCM, software-based 16-bit linear PCM, MP3 and VOX.	None
		ATP-24A/PCIe+	1) Uses PCIe X1 design, PCIe X1, X2, X4 and X16 slots supported, H.100 bus (CT-BUS) unsupported. 2) Support of DMA transfer of recording data. 3) A 50-pin RJ21 connector available on board for line connection. 4) Each board is equipped with 8 analog recording circuits and able to support another 2 recording modules (each module embraces 8 recording channels). 5) Support of a large selection of voice CODECs, including hardware-based A-Law (G.711), μ-Law, IMA-ADPCM, MP3, software-based 16-bit linear PCM, MP3 and VOX; support of either GSM or G.729A for recording.	None
SHD	Digital trunk boards	**SHD-120D-CT/PCI/CAS** **SHD-240D-CT/PCI/CAS**	1) Support of SS1 (CAS) and channel banks.	None
SHT	Analog voice boards	**SHT-16C-CT/PCI/EC**	1）Besides the features of the SHT-16B-CT/PCI board, it enhances the capability in echo cancellation, prolonging 8ms to 32ms for time delay estimation; 2）Supports FSK transmission and reception in the size of up to 1024 bytes, tone clamping over bus at the frequency of 450Hz, and energy threshold setting of FSK.	None

- **Fixed Bugs in CTILinux Ver.5.0.1.0**

Series	Feature	New Feature & Fixed Bug
All Boards	Bug fix	1. Fix the bug that event delay occurs while using the EVENT_POLLING mode for programming.
		2. Fix the bug that a great value is returned in initialization while invoking the API function ssmsetflag, with the parameter nType set to F_InVoiceToBus=8.
		3. Correct the error that the system may report 'SEGMENT FAULT' while invoking the functions in the library ShInitPci.
		4. Fix the bug that the system fails to play voice files specified in shindex.ini by invoking the function SsmPlayIndexString().

- **Newly Added Features from SynCTI 5.0.2.0**

Series	Feature	**Supported Board Model**	New Feature & Fixed Bug
SHT	Analog voice boards	All SHT series boards	Fix the bug that the program will exit abnormally when 2 fax channels are connected over bus.
			Fix the bug that the API function call of SsmSearchIdleCallOutCh on an SHT series board returns the on-board physical channel number.
			Fix the bug in invoking SsmGetPolarState based on either SynCTI 5.0.0.0 or SynCTI 5.0.1.0.
		C-type analog voice boards	Support FSK transmission and reception in the size of up to 1024, tone clamping over bus at the frequency of 450Hz, and energy threshold setting of FSK.
			Fix the bug that the SHT-16C-CT/PCI/EC board will wrongly receive CallerID in FSK mode when it stays in an idle state.

			Improve the faxing capability by upgrading voice bin files for C-type analog boards and fax bin files for C-type and D-type fax boards.
		USB voice boxes	Fix the bug that a channel on the SHT-4B/USB box, when it exits the conference, can still hear voices from other channels in the conference.
			Support off-line detection of USB, adding a returned event S_CALL_DISCONECT with the value of 135 to the function SsmGetChState(int ch).
			Fix the bug that the FSK reception is improper on the SHT-4B/USB box by using the function SsmStartRcvFSK_II.
		SHT-16B-CT/PCI SHT-16B-CT/PCI/FAX	Fix the bug that the program will exit abnormally when the function SsmTalkWith is invoked on a fax channel which is added by configuration.
		SHT-8C/FAX SHT-8B-CT/PCI SHT-16B-CT/PCI SHT-2B/4B/USB	Support the enhanced tone detector, newly adding the following configuration items ToneDetectorMode=1 VoiceOffDetermineTime=5000 MaxToneDetectorItem=5 ToneDetectorItem[0-4]=0
SHD	Digital trunk boards	All SHD series boards	Fix the bug that the event E_CHG_PcmLinkStatus appears again and again when all on-board physical links are in good state.
	SS7 support	Digital trunk boards which support SS7 signaling	Add an event interface to the function SsmGetSs7SpyMsu, with the event code of E_RCV_Ss7SpyMsu.
			Fix the bug that the driver performs an illegal operation when signaling messages are delivered through TS1 based on TUP and the PBX sends the signaling message of Single Circuit Reset.
			Fix the bug that illegal operation occurs upon the API function call of SsmIsupSendUsr.

			Support the call of SsmGetIsupParameter to acquire the Reason field and its content in the ISUP ACM message.
			Fix the bug that illegal operation of Ss7Monitor occurs in very few PBX environment.
			Fix the bug that Ss7Monitor shows irrecognizable characters in English operating system.
			Fix the bug that the program hangs when the first voice path for TUP is deleted in SS7Cfg.exe.
		Digital trunk boards which support SS7 and SS1 signaling	Fix the bug that the setting is invalid by the function of SsmSetWaitAutoDialAnswerTime Only support SS1, TUP and ISUP (ISDN unsupported)
		A-type digital trunk boards	Fix such bugs as the improper silence and the missing of DTMF digits which result from the lack of on-board DST resources.
		D-type digital trunk boards	Fix the bug that errors occur when a D-type board is acquiring the board authorization code.
			Fix the bug that some PCM fails to transmit and receive DTMF in the stress testing using six 240D/EC boards
			Fix the bug that the SS7 server fails to be started after the position of a 120D board is changed with other boards which are used with it in the configuration program.
		cPCI boards	Fix the bug that there are noises in an outgoing call from a channel on the 16E1 board.
	E1-ISDN	Digital trunk boards which support ISDN signaling	Fix the bug that at the ISDN user side the first digit of the called party number will be ignored during log output of decoded messages, which results in a lack of the first digit in the sent out called party number.
			Fix the bug that the remote pickup event is triggered by the driver twice when the remote end picks up the auto-dialed ISDN call.

			Modify the reference values for ISDN call, eliminating any repeat in values for PCM.
			Enable the support of the configuration item MaxWaitAutoDailAnswer for ISDN applications.
			Fix the bug that a wrong type of called party number is sent during a dialed-out ISDN call.
	SS1	D-type digital trunk boards	Enable the SHD-120D-CT/PCI board to support SS1
		C-type digital trunk boards	Support the use of TS1 to process signaling messages based on SS7.
		SHD-120A-CT/PCI	Fix the bug that the SHD-120A-CT/PCI board fails to detect BargeIn when it serves as a large-capacity station board.
		SHD-60A-CT/PCI	Fix the bug that the system which uses SHD-60A-CT/PCI as a large-capacity board will hang when the operating system is restarted after the driver is closed.
	Faxing	C-/D-type fax boards	Enable the driver to support the PageNumber tag for tif files.
			Fix the bug in fax transmission by upgrading related bin files for both C-type and D-type fax boards.
ATP	Analog tap passive boards	SHT-8A/PCI	Fix the bug that the voltage event for the recording modules on the SHT-8A/PCI board is output abnormally.
DTP	Digital trunk passive boards	SHD-60B-CT/PCI/FJ	Fix the bug that the recorded voice file is only in the size of 1K when a 60B/FJ board and a 240D board are using the hardware-based GSM format for recording at a same time.
		SHD-30B-CT/PCI/FJ	Fix the bug that when the SHD-30B-CT/PCI/FJ board is used for ISDN monitoring and FIFO_Push_Isdn_Setup is first selected to identify the channel which delivers the SETUP message, the board will fail to monitor the call if the channel specified by the SETUP message is inconsistent with that specified by its response message.

		SHD-30A-CT/PCI/FJ SHD-60A-CT/PCI/FJ SHD-30B-CT/PCI/FJ	Fix the bug that there is no voice on the line when the function SsmTalkWith is invoked on the 30A/FJ, 60A/FJ or 30B/FJ board.
		All DTP series boards	Add an event E_RCV_Ss7SpyMsu to correspond to the function SsmGetSs7SpyMsu.
DST	Digital station tap boards	All DST series boards	Support D-channel events for ToshibaCTX PBX.
			Support D-channel events for View 32KD (SA) PBX and the phone 32KD-06G used in it.
			Mend the bin files for IWASTU, MERIDIAN, SUMSANG, TECOM PBXes to prevent the overflow of signaling buffer.
			Fix the bug that the channel state doesn't have any change when the phone NorstarBCM T7100 is pushed into or pulled out from a DST series board.
			Fix the bug that wrong information is displayed when a DST series board is working with Notel Meridian 61C PBX and the phone 2616 used in it.
			Fix the bug that when a DST series board is working with NOTEL MISL100 PBX and the phone 3903 used in it, the channel state still stays at 'Active' in the driver after a call is hung up.
			Fix the bug in parsing LED events when a DST series board is working with Nortel Norstar PBX and the phone 7316 used in it.
			Fix the bug in the output of D-channel events when a DST series board is working with Alcatel OmniPCX Enterprise PBX and the phone 4019 used in it.
			Fix the bug that there are errors in LCD display when a DST series board is working with the phone Alcatel4049.
			Fix the bug that the use of Toshiba CTX670 PBX and the phone DKT3020C-SD for monitoring gives a bad voice effect.
			Fix the bug that the program will be illegally shut down when the occupied memory becomes greater and greater after running TEST or when the function SsmCloseCti is invoked.

			Fix the bug that while using the AVAYA 2-line PBX based on the driver SynCTI 5.0.1.0, the reconfiguration after writing the PBX type into module fails to be passed.
			Fix the bug that there appear noises when the DST series board is working with BP250 PBX as well as the phones 3211 and 3212 used in it.
			Fix the bug that there appear noises when the DST series board is working with Avaya8720 PBX and the phone 2410D used in it.
			Fix the bug that when a DST series board is working with G3SI PBX and the phone 6416 used in it, after calling the monitored phone, the calling party number can only be seen in LCD when the on-board channel stays in an idle state, and the received CallerID under both the 'ringing' and 'active' states is wrong.
			Fix the bug that there are errors in the displayed messages when NEC2400 PBX is supported by the DST series board.
			Fix the bug that the driver changes the channel state from 'ringing' to 'idle' when a DST series board is working with Nortel OBT 61C PC console and the monitored console picks up the console-to-console call.
SHN	VoIP boards	All SHN series boards	Fix the bug that the function call of SsmSearchIdleCallOutCh fails when an SHN series board and an SHT series board are running in a same system.
			Fix the bug that RTP numbers list in disorder when a VoIP board connects to an soft terminal which has the capabilty of silence detection.
			Fix the bug that only one DTMF can be sent at a time based on SIP, i.e. if more than one DTMF are sent at a same time, the Test program will hang after a long time of sleeping.
			Fix the bug that the SHN-32A-CT/PCI board, when it connects to Huawei gateway, fails to receive DTMF and get through incoming calls.

			Fix the bug that a bad voice effect is given by using SHN-32A-CT/PCI to talk with some soft terminals.
			Fix the bug that a voice with the default value of volumn is decreased rapidly when it goes onto bus after two channels on a SHN-32A-CT/PCI board are linked to each other by invoking the function SsmTalkWith.
Newly Added Function	1. Add a new function SsmFaxGetDcnTag to judge if the remote fax machine has ever been compelled to stop when the fax reception is successfully completed by using a fax board. If the application program uses the event mode for programming, we suggest you use the event E_PROC_FaxDcnTag instead.		
	2. Add a new function SsmStartRcvFSK_III to support FSK transmission and reception in the size of 1024 bytes.		
	3. Add new functions SsmGetNumType and SsmSetNumType to get and set the type of number based on ISDN protocol.		
	4. Add a new function SsmGetRingFlag to get the ring flag, with the return value -- 0: not ringing; 1: ringing.		
Modified Function	1. Add new features to SsmSetFlag and SsmGetFlag: nType = 17 Sets whether to put tones onto bus. The corresponding configuration item: ClearInVoiceOnRx450Hz nType = 18 Sets an energy threshold value for FSK receiving. The corresponding configuration item: FSKMinGate		
	2. Enable the function SsmRecToFileA to support hardware-based GSM for recording		
	3. Fix the bug that the program crashes upon the function call of SsmSetUnimoduleState.		
Newly Added Configuration Item	1. Add a new configuration item ClearInVoiceOnRx450Hz =n,n,n,n,n,n,n,n,n,n,n,n,n,n,n,n under the section [BoardId=x] to set whether to put tones onto bus. n=0: Not onto bus (default); n=1: Onto bus.		
	2. Add a new configuration item FSKMinGate=b,b,b,b,b,b,b,b,b,b,b,b,b,b,b,b under the section [BoardId=x] to set an energy threshold value for FSK receiving. b=0: No threshold value (default); b=n(n>0): The set threshold value.		
	3. Add a new configuration item DeventUpdates=n under the section [SystemConfig] to control the event filtration as a switch, helping the board to filtrate some repeated events. n=0: Filtrate repeated events (default); n=1: Not to filtrate repeated events. This configuration item is only applicable to the DST Series boards.		

	4. Add a new configuration item AllowTimeoutInSpyISDN=n under the section [BoardId=x] to determine whether to reset the channel state back to S_SPY_STANDBY if time's out after the channel keeps in the S_SPY_RCVPHONUM state for a long time. b=1: Reset the channel state back to S_SPY_STANDBY if time's out after the channel keeps in the S_SPY_RCVPHONUM state for a long time (default); b=0: Not to reset the channel state back to S_SPY_STANDBY if time's out after the channel keeps in the S_SPY_RCVPHONUM state for a long time. This configuration item is only applicable to the Synway DTP series boards.
	5. Add a new configuration item ProgressExt=n under the section [ISDN]. In an incoming call, if the ALERT message is sent by the local end when the value of ProgressExt is not 0, it will contain the progress indicator. For the meaning of each bit, refert to Synway Programmer's Manual (Version 5.0.2.0). It is an advanced configuration item, only applicable to ISDN incoming calls.
	6. Add a new configuration item UserWaitAfterCallProceeding=n under the section [ISDN] to set the maximum time that the local end waits for the remote end to send back the acknowledgement message in an outgoing call. If the local end doesn't receive the acknowledgement message from the remote end before timeout, it will actively disconnect the call. n: The waiting time, with the default value of 10s. It is an advanced configuration item only used for ISDN user-side outgoing calls.
	7. Add a new configuration item ToneDetectorMode=b under the section [SystemConfig] to select a kind of tone detector for use. b=0: Use the common tone detector (default); b=1: Use the enhanced tone detector.
	8. Add a new configuration item VoiceOffDetermineTime=n under the section [SystemConfig] to set the time of silence detection. n: Usually being 1.2 times the period of a ringback tone, with the default value of 5000.
	9. Add a new configuration item MaxToneDetectorItem=N under the section [SystemConfig] to set there are how many kinds of tones to be detected. 1≦N≦20, with the default value of 5.
	10. Add a new configuration item ToneDetectorItem[n] = M,F1,F2,F3,T1,T2,T3,Con,Coff,Ferr,Rbw,Terr,Pend,Ccnt,Tclr,Ppop,Eevent,Sstate,Sstop,We under the section [SystemConfig] to set each parameter for tone detection so as to detect all specified tones.

	11. Add a new configuration item ldr531=s under the section [BoardId=x] to set the ldr file which is loaded to the chip bf531 to enable the hardware-based encoding for the ATP-24A/PCI+ board. s: Indicates the name of an ldr file which is loaded to the chip bf531 for hardware-based encoding. For GSM recording: Load 'bf531_gsm.ldr'; For G.729A recording: Load 'bf531_729.ldr'.
Modified Configuration Item	Change the maximum value of WaitAfterDialTime from 20000 to 60000(ms).
DTMF	Fix the bug that the event triggered by the driver to receive DTMF keeps in a wrong size.
Log File	Enable the function SsmWaitForEvent to support log output to debugview.
Header File	1. Upgrade of VB header file: Solve the problem that the definition of MESSAGE_INFO.wEventCode in shpa3api.h is inconsistent with that in Shpa3api.bas.
	2. Upgrade of VC header file: Add SsmFaxGetFailReason, SsmSetContactInConf, SsmSetInterEventType to the header file of shpa3api.h.
Configuration Program	1. Fix the bug that the configuration item FileterInvalidCID generated by the configuration tool ShCtiConfig is written in a wrong format.
	2. Fix the bug in FJ board configuration by ShCtiConfig.
MP3 Engine	Update the MP3 engine in DEMO package to Version 3.98 (lameACM)
Demo	1. Enable the ATP_Polling demo to support ringing detection.
	2. Disable the ATP_Polling, ATP_Event demo to set voltage and BI sensitivity.
	3. Update the header file for Vbdemo in the DEMO package.
	4. Fix the bug that FAXDEOM fails to establish calls under the platform of VB.NET.
	5. Fix the bug in flashing for PBX Demo.
Software Tool	1. Add a software tool ShTA.exe, a tone analyzer, to the installaiton package. It is used to enable the start of detection and analysis on parameters of the call progress tones that go through analog trunks.

- **Newly Added Features from SynCTI 5.0.3.0**

Series	Feature	**Supported Board Model**	New Feature & Fixed Bug
SHT	Analog voice boards	All SHT series boards	Fix the bug that the function call of SsmSetUnimoduleState returns -1 and prompts failure in operation whether it is actually successful or failed
			Fix the bug that some tones cannot be detected by using the new tone algorithm.
		large-capacity station boards	Add a new event E_CHG_CBLinkStatus and an API function SsmGetCBLinkStatus for identifying the disconnection between the large-capacity board and the channel bank.
		C-type analog voice boards	Enable the newly developed C-type analog voice boards to support the enhanced tone detector.
			Fix the bug that the trunk and the station in a teleconference cannot hear each other when using the SHT-16C-CT/PCI/EC board.
		SHT-16C-CT/PCI/EC SHT-8C-CT/PCI/EC	Fix the bug that these two boards perform echo cancellation only allowing 32ms for time delay estimation although they are actually designed with 64ms.
		USB voice boxes	Fix the bug that the polarity reversal is sometimes detected wrong when using the USB voice box.
		SHT-8C/PCI/FAX SHT-16C-CT/PCI/FAX	Fix the bug that files in FAX format are unsupported after the driver is enabled to support ECM.
		SHT-4B/USB	Fix the bug that the voice output from audio jack is heard with large noise when using Channel 0 to monitor incoming calls on Channel 0.
SHD	Digital trunk boards	All SHD series boards	Fix the bug that when you set the length of calling party number to 16 or more than 16 digits for a TUP call outside, the calling party number sent out is null, with the length of 0.
			Add a new function SsmSearchIdleCallOutCh to enable a specified PCM in SS7 to search for an idle channel to make outgoing calls.
			Add a new event E_RCV_Ss7IsupCpg which is returned upon the receipt of a CPG message.

			Fix the bug that the function SsmGetRemoteChBlockStatus returns wrong values.
	SS7	Digital trunk boards which support SS7 signaling	Support the acquisition of original SS7 messages by setting the configuration item GetMsuOnAutoHandle under Section [ss7]. For auto call connections, original SS7 messages can be obtained via the E_RCV_Ss7Msu event or the SsmGetSs7Msu function.
			Fix the bug that channels on the SHD-120A-CT/PCI/SS7 board fail to enter the idle state in some ISUP environment, by setting the configuration item CircuitReset under Section [ISUP].
		A-type digital trunk boards	Fix the bug that our boards fail to connect with Ericsson MD110-CAS PBX based on LineSide E1.
		D-type digital trunk boards	Enable the SS7 E1 for D-type digital boards to support 31 voice paths.
			Fix the bug that channels go into the idle state upon the start of Test.exe while 16 240D boards are running together.
			Fix the bug that Blue Screen of Death occurs when a 240D board used in a large-capacity system is started for more than one time.
			Improve the message queue for SHD-240D-CT/PCI, adding the flag of FFt_Done, enhancing the capability to receive DTMF.
	E1-ISDN	Digital trunk boards which support ISDN signaling	Fix the bug that the channel number acquired by parsing an ISDN message is incorrect.
	SS1	Digital trunk boards which support SS1 signaling	Enable SS1 channels to support flash transmission and add two new states ‘wait for the end of flash’ and ‘flash ends’ for SS1 channels.
ATP	Analog tap passive boards	ATP-24A/PCI+	Support 8kbps and 16kbps MP3 recording in hardware. You may choose whichever format for recording and sampling through the configuration item ldr531 under the section [BoardId=x]. ldr531="bf531_mp3_8k.ldr": 8kBit/s, 8.000Hz, Mono MP3 encoding in hardware; ldr531="bf531_mp3_16k.ldr": 16kBit/s, 8.000Hz, Mono MP3 encoding in hardware.
		ATP-24A/PCI+ ATP-24A/PCI	Fix the bug that there is something wrong with the voice played back when Channel 0 on the ATP-24A/PCI board is used for double-buffer recording

			Add the message queue for ATP-24A/PCI, adding the flag of FFt_Done, enhancing the capability to receive DTMF.
		All ATP series boards	Add a new event E_CHG_FlashCount to be thrown out once the recording channel detects a flash. Meanwhile, enable the functions SsmGetFlashCount and SsmClearFlashCount to support recording channels.
DTP	Digital trunk passive boards	SHD-60B-CT/PCI/FJ SHD-30B-CT/PCI/FJ	Fix the bug that when the SHD-240D-CT/PCI and SHD-60B-CT/PCI/FJ boards are used together, SHD-240D-CT/PCI is placed in front and SHD-60B-CT/PCI/FJ is configured with ISDN monitoring, Blue Screen of Death appears when you click 'Apply'.
			Enable the function SsmRecToFile to support GSM recording in hardware.
			Fix the bug that a time of silence appears while using the SHD-60B-CT/PCI/FJ board to do DTMF recording.
		SHD-30A-CT/PCI/FJ SHD-60A-CT/PCI/FJ SHD-30B-CT/PCI/FJ	Fix the bug that only the last digit can be acquired when using the function SpyGetCalleeId to get the monitored called party number.
			Fix the bug that when using the SHD-30A-CT/PCI/FJ board to perform ISDN monitoring, incoming calls are under a good control but outgoing calls becomes out of control.
DST	Digital station tap boards	All DST series boards	Support the phones DTR3210c-SD and DTR2010D used in the Toshiba CTX 670 PBX.
			Support D-channel events for NEC Aspire PBX and the phone 24TXH used in it.
			Enable the state machine to support the Siemens AC5.0 console.
			Support D-channel events for NEC_DX2U_288M PBX and the phone DX2E-12BTH-A used in it.
			Enable the state machine to support D-channel events for harris and harrismap PBXs, but not to support ringing events.
			Support D-channel events for Ericsson MD110 PBX and the ELU5 - Console Card used in it.
			Fix the bug that a wrong calling party number is acquired when the DST series board is used to monitor two phones DKT3020c-SD and DKT2020 in the Toshiba CTX PBX.
			Fix the bug that there appear noises when the DST series board is used with the EON PBX.

			Fix the bug that when a DST series board is working with the Alcatel OXE PBX and the phone 4039 used in it, the calling party number acquired by invoking SsmGetCallerid changes to be irrecognizable characters after you do some operations such as software upgrade.
			Fix the bug that when a DST series board is working with the Alcatel OXE4400 4059 console, program exception occurs after test.exe is running for some time.
			Fix the bug that the status fails to be changed correct if you press the Speaker button before picking up the PANOSONIC digital phone, and that rings fail to be detected for incoming calls.
			Fix the bug that when a DST series board is working with the Alcatel PBX, the board will burst with large memory and return a great number of E_PROC_Recongnize events.
			Fix the bug that only silence is recorded by using the Nortel BCM400 PBX and the digital phones 7316 and 7100 used in it.
			Fix the bug that when a DST series board is working with the Siemens Hicom 118 2 IM PBX, the phone optiset e memory used in it cannot display the information that should be displayed.
			Fix the bug that the Siemens 3000 series phone AC 4.0 fails to monitor the ring status on a channel.
			Fix the bug that when a DST series board is working with Nortel 1000M and 3904 phones, the LCD information returned by the board becomes inconsistent with that displayed on the phone.
			Fix the bug that when a DST series board is working with the Meridian 61C PBX and the phone 2216 used in it, the system will show some channel states always stay idle after running for some time.
			Fix the bug that channel states cannot be monitored correctly when a DST series board is working with the Meridian PBX and the 3902, 3903, 3904 phones used in it.
			Fix the bug that no parameter values are output upon throwing the 1007 (release the button) event for an NEC phone.

			Fix the bug that when a DST series board is working with the Alcatel PBX and the 4200, 4400 and 4068 phones used in it, you can invoke the function DTRGetLCDStr() to get the incoming call number but not the LCD display for an outgoing call.
SHN	VoIP boards	All SHN series boards	Fix the bug that several channels fail to call out when you make outgoing calls at a same time from multiple (9) channels.
SHV	Voice-alteration boards	SHV-240A-CT/PCI	Fix the bug that the SHV-240A-CT/PCI board cannot alter voices when it is used with the D-type digital board and the SHT-16C-CT/PCI/EC board.
Record & Playback	All boards		Enable the boards which support MP3 recording to support MP3 prerecording.
			Fix the bug that the function call of SsmGetRecOffset returns an invalid result in GSM recording.
			Fix the bug that LoadIndexIniFile is likely to assign 0 byte memory and do not release the function call of Ssmclosecti.
			Fix the bug that the WAV header information recorded by using 16kbps MP3 format is incorrect.
			Support color ring detection, which can be enabled via the new event E_CHG_AMD.
Newly Added Function	1. Add a new function fPcm_MemGSMToPcm8 to convert the GSM formatted data stored in the buffer to the unsigned 8 Bit PCM formatted data.		
	2. Add a new function SsmGetCBLinkStatus to retrieve the connection state of a line between a large-capacity station board and a channel bank.		
Modified Function	1. Add new control values for relative channels to the function SsmSetFlag to solve the problem of returning wrong values.		
	2. Fix the bug that program exception occurs when the parameter pV of the function SsmRecToFileB is set to NULL.		
	3. Fix the bug that the type of calling or called party number does not change even though the function call of SsmSetNumType is successful.		
	4. Fix the bug that the function call of SsmISDNGetDisplayMsg fails to get the Display field.		
Newly Added Configuration Item	1. Add a new configuration item JudgeLineVoltage under the section [SystemConfig] to adjust the threshold voltage for ringing detection. Value range: integral number between 0 and 233.		
	2. Add a new configuration item PolarIgnore under the section [BoardId=x] to set a threshold voltage for ignoring interfering signals on a line lest they are mistaken by the driver for polarity reversal signals.		

	3. Add a new configuration item GetMsuOnAutoHandle under Section [SS7] to determine the acquisition of original SS7 call messages. GetMsuOnAutoHandle=1: Allowed to obtain the original SS7 call messages in auto call connection; GetMsuOnAutoHandle=0: Not to output the original SS7 call messages in auto call connection.
	4. Add a new configuration item ProgressExt=n under Section [ISDN] to specify the progress indicator for altering, with the default value of 0.
	5. Add a new configuration item CircuitReset under Section [ISUP]. CircuitReset=0: After the SS7 service is enabled, the circuit goes into an idle state without sending a circuit reset message; CircuitReset=others: The circuit is reset after the SS7 service is enabled. The default value is 1.
Log File	1. Enable the functions SsmSetIsupParameter() and SsmSetNumType to support log output to debugview.
	2. Enable the function SsmISDNGetStatus to support the log output feature.
	3. Support the log output of SS7 monitoring to the file SpyIFrame.log.
	4. Modify the way to output ISDN log. LogMode=0: Resave a log file once the number of messages reaches 100,000 (default); LogMode=1: Save a log file each day.
Configuration Program	1. Fix the bug that when the SHD-240D-CT/PCI and SHD-60B-CT/PCI/FJ boards are used together, SHD-240D-CT/PCI is placed in front and SHD-60B-CT/PCI/FJ is configured with ISDN monitoring, Blue Screen of Death appears when you click 'Apply'.
	2. Fix the bug that if you use ShConfig.exe to configure items about monitoring, then close and reopen ShConfig.exe, the relative dialog shows nothing (what you previously wrote in is completely missing).
	3. Fix the bug that the number of channels goes erroneous when you use ShConfig.exe to configure multiple boards and delete one of the boards.
	4. Enable ShConfig.exe to support the configuration of CircuitReset.
	5. Fix the bug that the system prompts error when you use ShConfig.exe to configure the SHD-120A-CT/PCI/ISDN board.
	6. Fix the bug that the system reports 'MaxPhoNumRule=0' when you use the English version of ShCtiConfig.exe to configure digital boards with default settings.
	7. Fix the bug that the system prompts error when you use ShConfig.exe to configure 30-channel ISDN boards with default settings and click on 'Apply'.
	8. Fix the bug that the system restarts or halts when you use ShConfig.exe to configure two E1s numbered 0 and 1 for SS7 and the other two numbered 2 and 3 for ISDN, and then click 'Apply'.
	9. Correct the default value and the value range of the configuration item RecordIFrame in ShCtiConfig.exe.
	10. Increase the maximum voice paths in SS7Cfg.exe from 256 to 512.

MP3 Engine	1. Fix the bug that voices recorded in MP3 format can not be played back through the local windows media.
Demo	1. Add a new demo with the function SSMRECTOFILEA to support VB.NET.
	2. Provide a new demo to enable the recording on the board and the playback on the computer at the same time.
	3. Add a new demo Dial_Event_Messagev.exe for color ring detection.
	4. Fix the bug that TEST.exe crashes when it recalls the memory recording after modifying the buffer size.
	5. Fix the bug that the demo of SHR_StateEvent reports error.
	6. Fix the bug that CasTool.exe fails to record the information about the second DST board.
	7. Fix the bug that the system reports 'Fail to pick up' when you use the demo of Fax_Event_Message to send fax.
	8. Add a new demo for digital phones to support C+ Builder6.
	9. Add vb.net (8.0) (programming mode: windows message) to the Recorder-DST demo.
Software Tool	1. Fix the bug that when Ss7Monitor.exe is used in dual-gateway mode, the system prompts error 'can not read the 0x00000000 memory quoted by the 0x00000000 instruction' once you switch over the gateway.
	2. Fix the bug that Ss7Monitor.exe fails to read configurations after 47 when you configure more than 47 CIC PCM under Section [ISUPRouter] in ss7server.ini.
	3. Add mtp3.dll to the driver installation package for quasi-associated configuration, temporarily named 16dpc_stp_mtp3.dll.
	4. Enable SS7Cfg.exe to support the configuration of multiple UP_DPC, enhancing its capability in quasi-associated configuration.

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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.

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