

A horizontal row of seven green squares with varying shades and internal patterns, including diagonal lines and gradients.

Asterisk Open-Source Enabling Technologies



About us

Synway specializes in providing superior Media Processing & Signaling Technologies, Telephony Hardware and Integrated Multimedia Switch in use for convergence (voice/data/video) communications. Since 1995, over 1000 software developers and system integrators have integrated Synway's offerings to deliver a broad range of TDM and VoIP-based applications, including unified communications, call center, mobile VAS, media gateway, fax, conferencing, call recording, Asterisk-based open source applications for operators and enterprises worldwide.

Having continued to optimize and expand its product portfolios to cater to various needs, Synway has consolidated its position as a leading vendor in international market for its widest range of portfolios: IP&TDM board & new-generation integrated multimedia switch platform for SP developers, Asterisk-based hardware & appliances, and the most diverse hardware options for passive call recording (logging) applications in IP and TDM network.

With 200 teammates, Synway makes all efforts to deliver quality support and service and help clients offer a variety of customizable, high-performance and cost effective solutions.



FXM Series

- Up to 32 ports FXO/FXS configurable analog telephony cards(per PCI/PCle slot)
- DSP-enabled echo cancellation of carrier-grade(32ms or 128ms optional)
- Six-year warranty, refundable two-month return, lifetime maintenance

Introduction

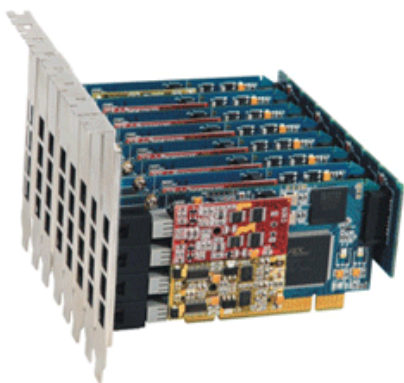
The FXM is the state-of-art open source architecture which supports configurable FXO and FXS lines for any high-scalability open source applications. Widely used for small/medium/large sized systems such as IP-PBX, voicemail, IVR, call center and unified messaging, the FXM series provides software developers with robust enabling hardware to create open source solutions with high flexibility and matchless price/performance advantages.

Modular architecture enables clients to design countless solution portfolios by convenient configuration and high expandability. A single card supports up to 32 channels when installing mixed FXO/FXS modules

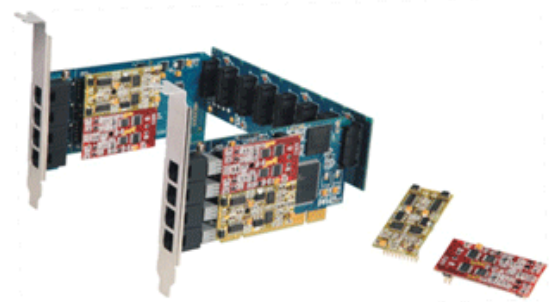
on the baseboard. With rich DSP resources and advanced data processing DMA technology, the FXM minimizes the utilizations of host CPU to establish versatile, high-scalability platforms in highly demanding environments.

The FXM series leverages SuPerForm (up to 128ms echo tail for each channel) to reassure superior voice quality and echo cancellation in complex networking environments. High-adjustability, original and complimentary SuPerForm is based on Synway-owned, certified DSP algorithm and carrier-grade applications, and can be "automatic Adaptability" optimized by site environments.

FXM Architecutre



To configure any density (2~32 channels), the FXM baseboard can be mounted on a backplane, which has up to eight bus connectors to mount additional daughterboard for higher density. For that, a FXM baseboard and up to seven daughterboards (with no PCI/PCle form factor) can be combined together by this backplane, and share the synchronous clocking of the FXM320X baseboard's PCI interface. This add-ons architecture(FXM series) is powered internally.



Each FXM baseboard or daughterboard has two sockets to mount two FXO/FXS/FXC modules. Each FXO and FXS module supports two FXO lines and FXS lines respectively. The FXC module supports one FXO line and one FXS line, and works properly in power-off conditions for its power-off protection design. Up to 16 FXO/FXS modules can be installed in FXM's baseboard/daughterboards(connected by a backplane) for 32 FXO/FXS lines

FXM Series

Key features & Benefits

- **SuPerForm superior voice quality and echo cancellation**

High-adjustability, original and complimentary SuPerForm, can be "automatic adaptability" optimized by site environments for the unmatched voice enhancements (up to 128ms echo tail), accurate DTMF/tone detection.

- **Largest-scalability powered by most flexible architecture**

Four interdependent components, baseboard, daughterboard, FXO/FXS modules and backplane Bus connector, reassure the most cost effective and flexible IP PBX system.

- **Compact size for saving power consumption and space**

The most compact-sized when compared with all comparable products, and allow highest-density (up to 32 FXO/FXS channels) installation in any 2U server; Apart from space efficiency, the FXM series adopt lowest-power-consumption design to relieve the ever-expandable systems off high power consumption.

- **Carrier-grade reliability and performance in any situation**

32-bit bus master DMA data exchange at 132 MB/s across PCI(e) interface for data reading and writing helps minimize intervention over host CPUs, optimizing per channel DMA streams and maximizing system's reliability in the most difficulty situation.

Technical Specifications

- **Product models**

FXM3200P: PCI, half length, 2 to 32ports configurable, 32ms EC
FXM3201P, PCI, half length, 2 to 32ports configurable, 128ms EC
FXM3200E, PCIe, half length, 2 to 32ports configurable, 32ms EC
FXM3201E, PCIe, half length, 2 to 32ports configurable, 128ms EC
FXM3210P: PCI, full length, 2 to 32ports configurable, 32ms EC
FXM3211P, PCI, full length, 2 to 32ports configurable, 128ms EC
FXM3210E, PCIe, full length, 2 to 32ports configurable, 32ms EC
FXM3211E, PCIe, full length, 2 to 32ports configurable, 128ms EC
FXD400: 4 ports daughter board
FXO200: 2 ports FXO module
FXS200: 2 ports FXS module
FXC200: Combo module with 1 FXO & 1 FXS ports

- **Physical characteristics**

Dimensions: Half length: 132.5x64mm² (excluding L-bracket);
Full length: 270x64mm² (excluding L-bracket);
Weight: Half length: approx. 70g (Excluding modules);
Full length: approx. 120g (Excluding modules);
Full length: approx. 120g (Excluding modules);
Module: approx. 10g;
Includes compatible mounting clips for installation in 2U rack-mount servers;
The on-board RJ11 jack can connect directly or via a proper 2-way hub with telephone lines, making connection easy and malfunctions are.

- **Audio Specifications**

CODEC: CCITT A/μ-Law 64kpbs
Distortion: ≤3%
Frequency response: 300-3400Hz (±3dB)
Signal-to-noise ratio: ≥38dB
Echo suppression: ≥40dB

- **Operating Systems**

Linux, FreeBSD, Solaris, Unix.

- **Self-diagnostics and alerting for efficient maintenance**

Self-diagnostic indicator light positioned behind RJ11 port; technician can detect internal installation (FXO or FXS modules) and diagnose malfunction outside of server.

- **Fax capability optimized by Synway hardware**

Resource saving: With Synway T1/E1/J1 card, Fax synchronization cable and FXM analog card mounting a FXS module, can connect the FXS module (FXS line) to your fax machine, and so leave a separate analog line out;

Error Correction Mode (ECM): The FXM series analog cards can synchronize with Synway T1/E1/J1 cards and PSTN clocking for error-free fax and modem pass through, and optimize your Fax capability and avoid missed lines, blank sheets or page missing;

- **Field-upgradeable firmware**

On-board DSP algorithm can be loaded through driver, and other firmware is conveniently upgradeable through server.

- **OS and open sources IP-PBX supported**

Support Unix, Linux and Solaris; compatible with Zaptel, and support a broad range of open source PBX systems, including Asterisk, Trixbox, Yate, Freeswitch, CallWeaver, Elastix and more.

- **Power Consumption**

FXM series baseboard with 2 FXO/FXS modules
+3.3V DC: 1500mA (4.95W)
+12V DC: 500mA (6W)
FXM series daughterboard with 2 FXO/FXS module
+5V DC: 600mA (3W)
+ 12V DC : 500mA (6W)

- **Input/output Interface**

Ringing current & battery feed power supply jack: One 4-pin MPC-4;
Telephone line jack: Four 4-pin RJ11/RJ21;

- **Environment**

Operating temperature: 0℃—55℃
Storage temperature: -20℃—85℃
Humidity: 8%—90% non-condensing
Storage humidity: 8%—90% non-condensing

- **Maximum System Capacity**

Depends on the system consumption of Asterisk and the processing capability of computer.

- **Certification, Environment standard & Production Quality**

CE, FCC Part 15 Class A and Class B, EN 55022, EN 55044, CISPR 22, CISPR 24;
Safety: Lightning resistance: Level 4
ISO 9001:2000
RoHS

- **Warranty**

6-year warranty, refundable 2-month return, lifetime maintenance, free support.

TEJ Series

- 1/2/4/8 ports of T1/E1/J1 Digital telephony cards
- DSP-enabled echo cancellation of carrier-grade(32ms or 128ms optional)
- Six-year warranty, refundable two-month return, lifetime maintenance

Introduction

Compatible with Asterisk and other open source platform, the TEJ series is the robust hardware architecture to help software developers and service providers deliver cost effective open source solutions, including IP-PBX, gateway, IVR or other telephony applications. With 1, 2, 4 or 8 trunks (T1/E1/J1) per card, the telephony cards of the TEJ series support up to 720 voice calls in a single system in the most difficulty situations.

The TEJ series leverages Synway's patent-pending echo cancellation technology and decades of voice processing expertise to guarantee high audio quality and unmatched pricing points. Up to 128 ms, the Synway's

echo canceller is on-board DSP-enabled and eliminates the need for expensive echo cancellation module. Supporting each channel on the full eight E1, T1 and J1 lines, the echo processing capability has been field-proven by some leading carrier labs and made Synway's product offerings the most cost effective and accountable worldwide.

Compatible with PCI or PCI-express interface, the TEJ series also introduces the latest DMA data processing technology to minimize the consumption of CPU capability, enabling developers to design the high-performance solutions featuring high robustness, scalability and interoperability in the demanding telecommunication environments.

Key features & Benefits

● Configurable T1/E1/J1

A kind of highly flexible hardware platform, TEJ series supports software configurable T1/E1/J1 interfaces. Selectable interfaces, T1/E1/J1 lines, are conveniently available in a single box to fit into complex and hybrid environments.

● SuPerForm superior voice quality and echo cancellation

High-adjustability, original and complimentary SuPerForm can be "automatic adaptability" optimized by site environments for the unmatched voice enhancements(over 128ms echo tail), accurate DTMF/tone detection.

● Carrier-grade reliability in any situation

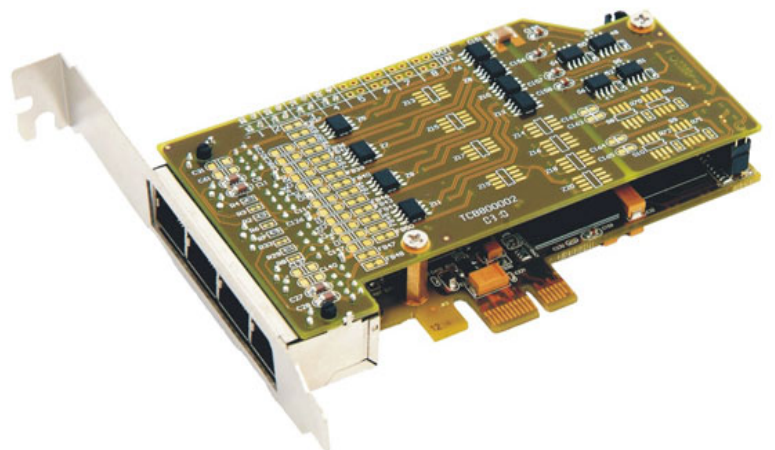
TEJ series (digital telephony card) utilizes DMA technology, and the optimized DMA streams minimize the hitting to host CPU, enabling software developers and service providers to deliver highly flexible and scalable systems (up to 720ports).

● Easy of use and intelligence

To design an array of open source applications conveniently, the TEJ series is optimized with rich features like friendly interface, automatic configuration and troubleshooting.

● OS and open sources IP-PBX supported

Support Unix, Linux and Solaris; compatible with Zaptel, and support a range of open source PBX systems, including Asterisk, Trixbox, Yate, Freeswitch, CallWeaver, Elastix and more.



TEJ Series

Technical Specifications

● Product models

TEJ100P/101P, PCI, 1E1/T1/J1.
TEJ100E/101E, PCIe, 1E1/T1/J1.
TEJ200P/201P, PCI, 2E1/T1/J1.
TEJ200E/201E, PCIe, 2E1/T1/J1.
TEJ400P/401P, PCI, 4E1/T1/J1.
TEJ400E/401E, PCIe, 4E1/T1/J1.
TEJ800P/801P, PCI, 8E1/T1/J1.
TEJ800E/801E, PCIe, 8E1/T1/J1.

Notes: TEJ*01 means 128ms echo cancellation.

TEJ*00 means 32ms echo cancellation.

● T1/E1/J1 Interface

1, 2, 4 or 8 E1/T1/J1 ports with a single PCI or PCI-Express interface optimized for high performance voice and data applications; Only software configurable for any kind of physical interfaces. Both T1 or E1 or J1 are mixed simultaneously in a single system.

● A complete range of open sources supported

Support for Asterisk, Trixbox and FreeSwitch, as well as other open source IP-PBX, Switch, IVR, or VoIP gateway applications.

● Audio Specifications

CODEC: CCITT A/μ-Law 64kbps

Distortion: ≤3%

Frequency response: 300-3400Hz (±3dB)

Signal-to-noise ratio: ≥38dB

Echo suppression: ≥40dB

● Form Factor

Automatically compatible with 5 V and 3.3 V PCI busses, and fully PCI 2.2 compliant PCI-X.

● SuPerForm echo cancellation

The self-adaptive echo cancellation technology taps 128 ms tails (1024 taps/128 ms tail per channel) and can effectively eliminate echo effect in any situation.

● DMA data exchange

The use of PCI-based DMA technique for data reading and writing helps minimize the cost of the host CPU. 32-bit bus master DMA data exchanges across PCI interface at 132 MB/s for minimum host processor intervention, and optimized per channel DMA streams and hardware-level.

● Input/output Interface

Digital trunk interface: TEJ100/101: 1 RJ48C jack; TEJ200/201: 2 RJ48C jacks; TEJ400/401: 4 RJ48C jacks; TEJ800/801: 4 RJ48C jacks;

● Physical characteristics

Dimensions: 64 x 120 mm² (excluding L-bracket)

Weight: approx. 100g or 120g

Includes compatible mounting clips for installation in 2U rack-mount servers.

● Environment

Operating temperature: 0℃—55℃

Storage temperature: -20℃—85℃

Humidity: 8%—90% non-condensing

Storage humidity: 8%—90% non-condensing

● Operating Systems

Linux, FreeBSD, Solaris, Unix

● Certification, Environment standard & Production Quality

CE, FCC Part 15 Class A and Class B, EN 55022, EN 55044, CISPR 22, CISPR 24;

Safety: Lightning resistance: Level 4

ISO 9001:2000

RoHS

● Warranty

6-year warranty, refundable 2-month return, lifetime maintenance, free support.

● EMC

Has excellent electromagnetic compatibility



Decades of accumulated DSP technology for echo cancellation and voice processing, empowers the most competitive open source architectures with matchless price and robustness.

BRI Series

- Up to 16 ISDN BRI ports
- Mix TE and NT modes
- DSP-enabled echo cancellation of carrier-grade(32ms or 128ms optional)
- Six-years warranty, refundable two-month return, lifetime maintenance

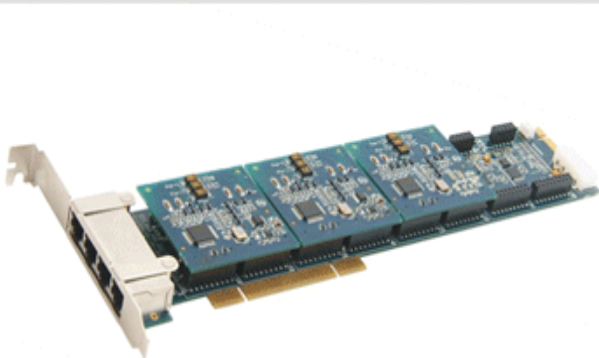
Introduction

Compliant with ISDN BRI, Synway BRI series card leverages SuPerForm echo cancellation technology (DSP-based, 32ms and 128ms echo tail) to assure superior voice quality in any complex networking environments.

Adopting modular design and expandability architecture, this series helps developers deliver high-flexibility, highly robust and cost effective systems in the hypercompetitive IP-PBX market.

S/T modules optional, the BRI series also enables developers to build up TE/NT hybrid systems, up to 16 channels per PCI slot, or up to 256 ports in single system.

BRI Architecture



To configure any density (2~16 channels), the BRI baseboard can be mounted on a backplane, which has dual-bus-connectors to mount additional daughterboard for higher density. For that, a BRI baseboard and its daughterboard share the synchronous clocking of the BRI baseboard's PCI/PCIe interface. This add-ons architecture(BRI series) is powered internally.



Each BRI baseboard or its daughterboard has four sockets to mount four S/T BRI modules. One S/T BRI module has two S/T four-wire interfaces, which supports TE or NT mode. 1~8 S/T BRI modules can be installed in BRI's baseboard/daughterboard(connected by a backplane) for a total 16 ISDN BRI lines.

Key features & Benefits

● Configurable freely

Each baseboard or daughterboard can mount four S/T BRI modules (each module supports two TE or NT lines) for system capacity up to 16 ISDN BRI TE/NT lines per PCI slot

● SuPerForm superior voice quality and echo cancellation

High-adjustability, original and complimentary SuPerForm, can be "automatic adaptability" optimized by site environments for the unmatched voice enhancements (up to 128ms echo tail), accurate DTMF/tone detection.

● Carrier-grade reliability and performance in any situation

32-bit bus master DMA data exchange at 132 MB/s across PCI(e) interface for data reading and writing helps minimize intervention over host CPUs, optimizing per channel DMA streams and maximizing system's reliability.

Technical Specifications

Technical Specifications

● Product models

BRI1610P, PCI, full length, 2 to 16ports configurable, 32ms EC

BRI1611P, PCI, full length, 2 to 16ports configurable, 128ms EC

BRI1610E, PCIe, full length, 2 to 16ports configurable, 32ms EC

BRI1611E, PCIe, full length, 2 to 16ports configurable, 128ms EC

● Environment

Operating temperature: 0°C—55°C

Storage temperature: -20°C—85°C

Humidity: 8%—90% non-condensing

Storage humidity: 8%—90% non-condensing

● Input/output Interface

Single motherboard/daughterboard: four RJ45 can be converted into eight S/T interface

Interface Type: S / T ITU-T I.430

Transmission code: AMI code

Terminal resistance: 100Ω optional

Output power: 38V ± 2V (NT only, optional)

● Audio Specifications

CODEC: CCITT A/μ-Law 64kbps

Distortion: ≤3%

Frequency response: 300-3400Hz (±3dB)

Signal-to-noise ratio: ≥38dB

Echo suppression: ≥40dB

● Physical characteristics

Dimensions: 270 x 64 mm² (excluding L-bracket)

Weight: approx. 120g (Excluding modules)

Includes compatible mounting clips for installation in 2U rack-mount servers.

● Fax capability optimized by Synway hardware

Resource saving: With Fax synchronization cable, Synway FXM analog card and BRI card can connect to the fax machine;

Error Correction Mode (ECM): The BRI series digital card can synchronize with Synway analog card and PSTN clocking for error-free fax and modem pass through, and optimize your Fax capability and avoid missed lines, blank sheets or page missing;

● Field-upgradeable firmware

On-board DSP algorithm can be loaded through driver, and other firmware is conveniently upgradeable through server.

● OS and open sources IP-PBX supported

Support Unix, Linux and Solaris; compatible with Zaptel, and support a broad range of open source PBX systems, including Asterisk, Trixbox, Yate, Freeswitch, CallWeaver, Elastix and more

● Maximum System Capacity

Up to 8 boards concurrently per system; up to 16 channels per board

● Power Requirements

Total Power Consumption includes the electricity use of all motherboards and daughterboards.

A single motherboard (with modules fully inserted)

+3.3V DC: 1100mA (power consumption: 3.63W)

+12V DC: 1000mA (power consumption: 12W, supplied by power socket, NT modules only)

+5V DC: 1000mA (power consumption: 5W, supplied by power socket)

A single daughterboard (with modules fully inserted)

+12V DC: 1000mA (power consumption: 12W, supplied by power socket, NT modules only)

+5V DC: 1000mA (power consumption: 5W, supplied by power socket)

● Audio Encoding & Decoding

A-Law 64kbps

μ-Law 64kbps

● Sampling Rate

8kHz

● Safety

Lightning resistance: Level 4

● Warranty

6-year warranty, refundable 3-month return, lifetime maintenance, free support.

Asterisk Appliance

- Up to 96 Asterisk-based FXO/FXS channels or 16 E1/T1/J1 ports or hybrid
- DSP-enabled echo cancellation of carrier-grade
- All-in-one architecture, Flexible configuration and hot swappable

Introduction

Based upon decades of field-proven voice technologies, Synway provides developers with a cost efficient, open programmable, integrated appliance to empower a range of Asterisk-based (other open-source) applications and services. In addition to superior voice enhancement feature, the appliance helps bridge existing wired and wireless networks with IP networks, and supports all of Asterisk-based applications, including IP-PBX, gateway, IVR, fax, conferencing in enterprise and service provider networks. Each chassis is based on single 1U, 2U or 6U architecture, and can be configured for digital or analog or hybrid open source systems.

Key features & Benefits

● Perfectly integrated

All-in-one architecture ensures the matchless interoperability and adopts to various wired and wireless networks; Help developers deliver high quality, cost effective open source applications.

● Rugged architecture

Embraces all of major PBX brands; support hot swappable, and could be changed or replaced over the front (or) back panel, with no need to open chassis cover.

● Flexible configuration for any network

Interface with high-scalability FXO/FXS/E1/T1 lines in Asterisk or other open source IP-PBX environment; support SuPerForm echo cancellation for voice enhancement.

● Adjustable capacity for customized needs

1U appliance enables 2~64 Asterisk-based FXO/FXS channels or 1~16E1/T1/J1 ports or hybrid;
2U appliance enables 2~96 Asterisk-based FXO/FXS channels or 1~24E1/T1/J1 ports or hybrid;
6U appliance enables 2~256 Asterisk-based FXO/FXS channels or 1~24E1/T1/J1 ports or hybrid;

● Carrier-grade reliability

Special power system with standby redundancy; advanced cooling system to reassure long-standing robustness; special air cleaner to protect against dust accumulation inside chassis; inside temperature control and alert system; no need to change wiring when changing functional modules as backplane's wiring outlet is separate from backplane's functional-module slots.

● Processing capability powered by Intel

Adopt Intel(R) X86, the Mini-ITX industrial embedded platform as host control unit (main board) to execute high-capacity, high-complexity applications.

● Robust engineering design

The motherboard has optional 4.3 inches WVGA(800*480) TFT LCD; support diverse interfaces(port) on the rear panel, including RJ11, RJ21, RJ48C, USB, etc.

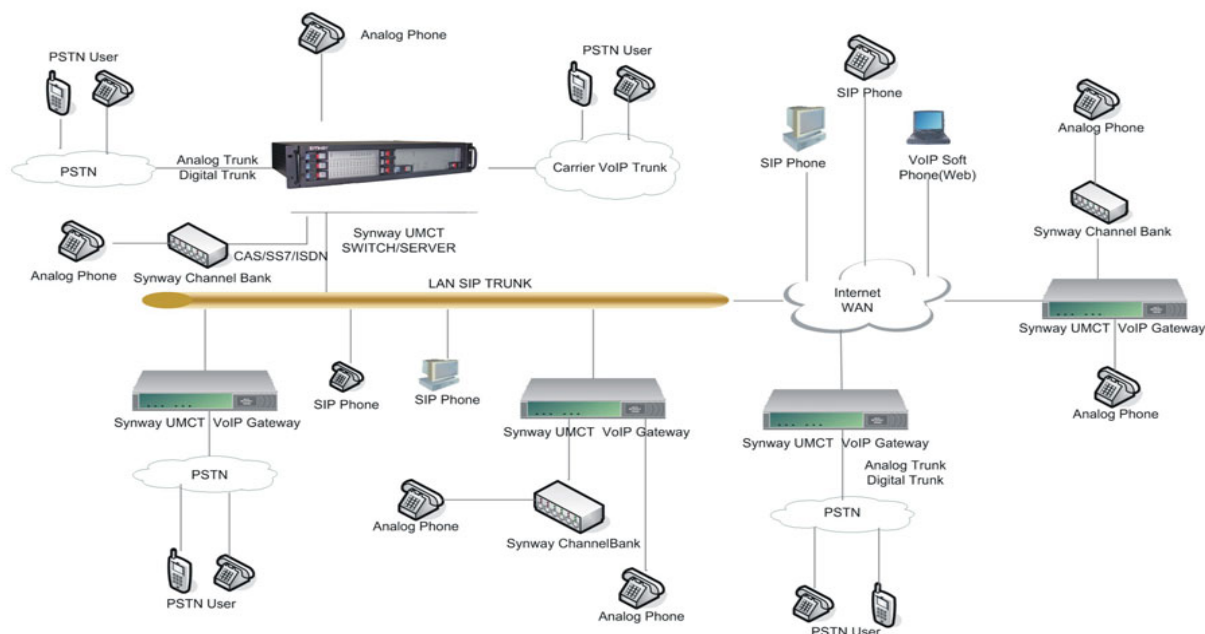
● Compliant with your applications and Asterisk

Support Unix, Linux and Solaris; compatible with Zaptel, and a range of open source PBX systems, including Asterisk, Trixbox, Yate, Freeswitch, CallWeaver, Elastix and more.



Asterisk Appliance

Typical Application



Technical Specifications

● FXO/FXS/E1(T1 or J1) configurable

Each Asterisk analog module can support 2-32 FXO/FXS ports for density up to 96 FXO/FXS lines per appliance;
Each Asterisk digital module can support 1-8 ports for density up to 16 E1/T1/J1 ports per appliance;

● SuPerForm echo cancellation

The self-adaptive echo cancellation technology taps 128 ms tails (1024 taps/128 ms tail per channel) and can effectively eliminate echo effect in any situation. Embedded in the card, without any physical space requirement.

● DMA data exchange

The use of PCI-based DMA technique for data reading and writing helps minimize the cost of the host CPU. 32-bit bus master DMA data exchanges across PCI interface at 132 MB/s for minimum host processor intervention, and optimizes per channel DMA streams and hardware-level.

● Physical characteristics

1U appliance

Dimensions: 44.5mm(H) x 480mm(W) x 390mm(L)

Weight: about 5kg (different for the number of optional modules)

2U appliance

Dimensions: 89mm(H) x 482.6mm(W) x 390mm(L)

Weight: about 9kg (different for the number of optional modules)

● Audio Specifications

CODEC: CCITT A/μ-Law 64kbps

Distortion: ≤3%

Frequency response: 300-3400Hz (±3dB)

Signal-to-noise ratio: ≥38dB

Echo suppression: ≥40dB

● Power Consumption

AC: 90-120V or 200~240V (SELECT RIGHT RANGE ON SHELF),
Frequency: 50~60Hz;
Power consumption: different for configuration, please refer to user manual;

● Environment

Ventilation: normal

Operating temperature: 0℃—40℃

Relative humidity: 10%—85%

Avoid dust accumulation

Anti-electrostatic: please Grounded

Installation recommendation: mounted on standard 19-inches rack;

● Operating Systems

Linux, FreeBSD, Solaris, Unix.

● Quality and Warranty

ISO 9001:2000

1U/2U/6U chassis (including mother board): 3-year warranty

● Security and Certifications

Lighting-proof grade: Level 4;

CE, FCC Part 15 Class A and Class B, EN 55022, EN 55044, CISPR 22, CISPR 24;

For RoHS compliance, please contact Synway's sales representatives;

CDCSeries

DSP-dedicated transcoding helps you build robust IP or Asterisk-based systems to handle high traffic calling as well as strategic expansion without constraints of host processor.

Introduction

The CDC series is DSP-enabled hardware with dedicated DSP resources for transcoding. Combined with the CDC series, Asterisk, any other open sources or host-based IP platform can handle complex codec translation between G.729A and G.711(A-law or μ -law) for the purposes of call origination or termination in the hybrid VoIP and TDM networks, with minimal hitting to host CPU.

Under constraint of processor capability, the CDC series leverages dedicated DSP algorithm to help you design versatile, high scalability open source solutions in complex networking environments where minimal utilization of host processor, high-traffic call processing or system expansion for higher density is required. It not only improves performances of Asterisk-based application, but also eliminates limitation of existing bandwidth. The CDC series has been considered as more cost effective and high-performance transcoding algorithm when compared with CPU's consumption for that purpose.

The PCI-2.2 compliant, half-length CDC series cards are the perfect substitute to Digium's TC400B for its cost advantages and certified DSP algorithm. For a variety of environments, each card is designed to support 50, 100 or 150 bi-directional codecs transformations and be seamlessly compatible with Synway's TEJ and FXM series.

Technical Specifications

● Physical characteristics

Dimensions: 120×95mm² (excluding L-bracket)

Weight: approx. 100g

● Environment

Operating temperature: 0°C—55°C

Storage temperature: -20°C—85°C

Humidity: 8%—90% non-condensing

Storage humidity: 8%—90% non-condensing

● Audio Specifications

CODEC: CCITT A/ μ -Law 64kbps

Distortion: $\leq 3\%$

Frequency response: 300-3400Hz (± 3 dB)

Signal-to-noise ratio: ≥ 38 dB

Echo suppression: ≥ 40 dB

● Power Requirements

Maximum power consumption: ≤ 8 W

● Audio Decoding & Encoding

From G.729A to A-Law, μ -Law

Key features & Benefits

● Seamlessly compatible with all Asterisk features

The CDC series is perfectly compatible with all Asterisk, helping Asterisk-based applications developers deliver cost-effective, feature-rich and highly accessible solutions.

● PCI 2.2 Bus compliant

Includes PCI 2.2 bus with burst data transmission rate up to 132 MB/s; PNP (plug and play) feature eliminates the need for jumper leads; the universal PCI design supports 3.3V/5V PCI slot and PCI-X slot.

● DMA data exchange

32-bit bus master DMA data exchanges across PCI interface at 132MB/s for minimum host processor intervention. The use of PCI-based DMA technique for data exchange minimizes utilization of the host CPU.

● Transcoding capability

The dedicated DSPs process the codec translations among G.711 A-Law, μ -Law and G.729a, which minimizes consumption of host CPU in all demanding situations and guarantees high scalability when system expansion is required. Each can handle 50, 100, or 150 bi-directional decompression (a-law to G.729a) or compression (G.729a to a-law).

● Audio CODEC

A-Law 64kbps

μ -Law 64kbps

G.729A 8kbps

Sampling Rate: 8kHz

● Certification

CE, FCC Part 15 Class A and Class B, EN 55022, EN 55044, CISPR 22, CISPR 24;

Safety: Lightning resistance: Level 4

● Warranty

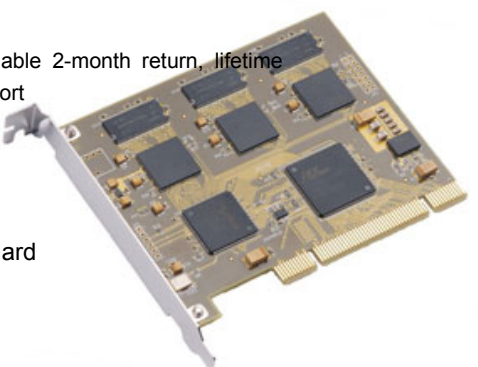
6-year warranty, refundable 2-month return, lifetime maintenance, free support

● Production Quality

ISP 9001:2000

● Environment standard

RoHS



Synway Products Family

Multimedia Switch Platform



UMCT integrated multimedia switch platforms are open programmable platforms with integrated multimedia processing and signaling capabilities. In addition to rich media resources, the switch platforms help bridge existing wired and wireless networks with IP networks, and integrate for IP (SIP and H.323) / TDM (SS7/ISDN/CAS) / mobility protocols with IVR, fax, conferencing, compression, echo cancellation and other media processing resources. 1U,2U,6U available, the UMCT supports up to 64E1/T1/J1 or 128FXO/FXS channels per system.

IP & TDM Series



IP & TDM boards consist of three product lines: SHN series for IP network, SHD series for E1/T1/J1 digital network, and SHT series with analog interface. Besides a complete range of media resources, such as fax, conferencing, compression, echo cancellation for voice enhancement, they also combine various built-in signaling protocol software packets, which include SIP, H.323, SS7 (MAP/SCCP/ISUP/TCAP/TUP/MTP1~3), ISDN variants, CAS and more

Call Recording Series



Synway offers the broadest range of call recording hardware platforms. For two decades Synway has consolidated its position as a leading international call recording hardware solution vendor. Having worked with our products over time, our clients have helped Synway achieve: the most diverse product ranges, greatest scalability, greatest compliance with multi-protocols and multi-networks, and the most installations in Asia and Europe. Our offerings can passively tap analog/digital/PBX/IP lines.



For more.....
[Http://ww.synway.net](http://www.synway.net)



VALUE MAXIMUM BEYOND TOMORROW

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