Contents

Asterisk - the Open Source PBX!

Asterisk is a complete PBX in software. It runs on Linux[1] and provides all of the features you would expect from a PBX and more. Asterisk does voice over IP in many protocols, and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware.

Asterisk provides Voicemail services with Directory, Call Conferencing, Interactive Voice Response and Call Queuing. It has support for three-way calling, caller ID services, ADSI, SIP and H.323 (as both client and gateway). Check the Features section later in this document for a more complete list.

Asterisk needs no additional hardware for Voice over IP. For interconnection with digital and analogue telephony equipment, Asterisk supports a number of hardware devices, most notably all of the hardware manufactured by Asterisk's sponsors, Digium™. Digium has single and quad span T1 and E1 interfaces for interconnection to PRI lines and channel banks as well as a single port FXO card and a one to four-port modular FXS and FXO card.

Easily build your own multiprotocol PBX on Linux!

Asterisk supports a wide range of TDM protocols for the handling and transmission of voice over traditional telephony interfaces. Asterisk supports US and European standard signalling types used in standard business phone systems, allowing it to bridge between next generation voice-data integrated networks and existing infrastructure. Asterisk not only supports traditional phone equipment, it enhances them with additional capabilities.

Using the Inter-Asterisk eXchange (IAX™) Voice over IP protocol, Asterisk merges voice and data traffic seamlessly across disparate networks. While using Packet Voice, it is possible to send data such as URL information and images in-line with voice traffic, allowing advanced integration of information.

[1] Asterisk is primarily devel-oped on GNU/Linux for x/86. It is known to compile and run on GNU/Linux for PPC along with OpenBSD, FreeBSD, and Mac OS X Jaguar. Other platforms and standards-based UNIX-like operating systems should be reasonably easy to port for anyone with the time and requisite skill to do so.

Asterisk provides a central switching core, with four APIs for modular loading of telephony applications, hardware interfaces, file format handling, and codecs. It allows for transparent switching between all supported interfaces, allowing it to tie together a diverse mixture of telephony systems into a single switching network.

Anatomy of an Asterisk PBX

The transition from circuit-switched PBX systems to server-based systems is possible because individual components are now modular, conformable and faster. Computer-telephony integration means that the same network of servers can process both data and voice – and the nature of Asterisk's modular design allows for extensive customization. For instance, dial plans for extensions can be configured to route traffic to either digital or analogue endpoints, including phones.

Wildcard PCI hardware from Digium provides access to the telephone company and to analogue endpoints such as traditional phones and networked devices such as the facsimile machine or printer. To protect customer investment and allow incremental migration, Digium provides PCI cards that interface with both analogue and digital phones.

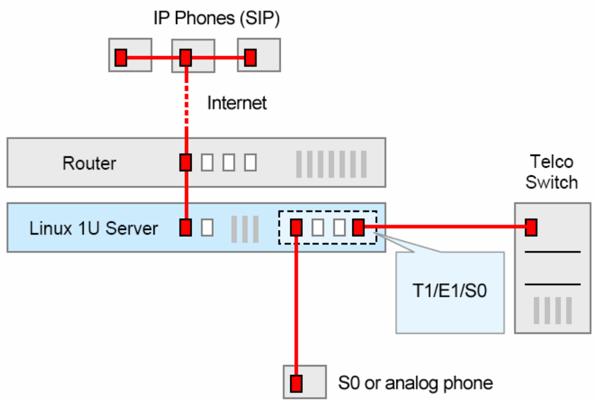
Asterisk also complements existing PBX systems by avoiding costly replacement of existing PBX investment while eliminating further vendor lock-in. In this fashion, Asterisk delivers hybrid PBX capability for purposes from simple additional lines to special features like branch-to-branch calling. Fully compatible with a wide range of IP and analogue protocols and codecs, Asterisk translates between them on the fly.

An Asterisk-based solution can support larger numbers of endpoints and services with little or no expense of adding additional PBX hardware. The key is having a Linux-capable administrator or service provider available to manage the computer telephony needs of the business. But a competent administrator or service provider working on the LAN or business applications has the same set of skills.

There are also companies (e.g. LocaNet oHG) that offer Asterisk system integration, administration consoles, and software packaging, support plans, and training.

A possible Asterisk PBX is illustrated below. The Asterisk application runs on the Linux server. IP phones connect directly to the LAN router, both internally and over the internet. Normal analogue or digital phones connect to interface cards in the Linux server. Asterisk can provide an outside line or a line to a PBX via different interfaces, both digitally or analogue.

The PBX components shown can typically be purchased for about € 2500, which is probably one-tenth the cost of other commercial alternatives. Actual performance of an Asterisk-based solution depends on traffic. CPU utilization will quickly increase when multiple calls or codec transcodings are in progress, placing larger demands on the server.



Open Source Asterisk Hardware

Asterisk offers a large set of capabilities. In addition to traditional PBX features such as voice mail, call conferencing, automatic call distribution and interactive voice response, Asterisk delivers caller ID, call queuing, bridging and toll-bypass. For example, intelligent routing features can automatically redirect calls, making service representatives more accessible, responsive and productive – a win for both the company and the caller.

Asterisk provides video call support, message-waiting indicators, call parking and transfer, paging, and intercom, among others features. It includes over 500 recorded professional voice prompts. Incoming voice mails are recorded in Wave format for storage or forwarding to a user mailbox, or forwarding by email for playback on a personal computer.

Supported VoIP Protocols

Asterisk supports three VoIP protocols, two industry standards and one originally developed specifically for Asterisk and adopted by number of other hardware and software devices.

* *Inter-Asterisk Exchange (IAX)*: IAX is the de facto standard VoIP protocol for Asterisk networking. IAX differentiates itself through transparent interoperation with NAT and PAT (IP masquerade) firewalls. This allows plug-and-play portability of PBXs and phones. IAX is extremely low-overhead (four bytes of header, as compared to at least 12 bytes of header for RTP based protocols like SIP and H.323).

* *Session Initiation Protocol (SIP)*: SIP is the IETF standard for VoIP. Its call control syntax resembles SMTP, HTTP, FTP and other IETF protocols. SIP is widely regarded as the replacement standard for H.323 VoIP due to its relative simplicity and human-readability.

* *H.323*: H.323 is the ITU standard for VoIP.

Important advantages of Asterisk

The Asterisk software package delivers a number of unparalleled advantages:

Substantial cost reduction - Combined with low-cost PCI-Card telephony hardware and a Linux PC server, Asterisk can be used to create a PBX at a fraction of the price of traditional PBX and key systems, while providing a level of functionality exceeding that of many of the most expensive systems available.

Control - Asterisk allows the user to take control of their phone system. Once a call is in a Linux server with Asterisk, anything can be done to it. In the same way that Apache gives the user fine-grained control over virtually every aspect of its operation (and configuration), the same applies to Asterisk.

Rapid deployment and development - Asterisk allows PBX's and interactive voice response (IVR) applications to be rapidly created and deployed. Its powerful command line interface (CLI) and text configuration files facilitate both rapid configuration and real-time diagnostics.

Customization - Through its support for internationalization, configuration files, and open source code, every aspect of Asterisk can be configured or modified. For example, codes for call features can be modified in order to accommodate proprietary protocols.

Dynamic content deployment - As web servers like Apache allow a developer to deploy dynamic content on the web, such as account information, Asterisk permits such dynamic content deployment over the telephone, with programming ease similar to the Common Gateway Interface (CGI), a vitally important web technology.

Flexible dial plan - Asterisk's unusually flexible dial plan allows seamless integration of IVR and PBX functionality. Many of Asterisk's existing features can be implemented simply with extension logic.

Myths Versus Reality

The transition to IP-based systems like Asterisk is unstoppable, but some myths persist. In reality, enterprise adoption is pushing aside the myths, technology is outpacing concerns, and first movers are demonstrating solid benefits.

Myth	Reality	Internal Use Strategy
Existing infrastructure cannot support the QoS standards necessary for real-time voice communications.	Ethernet switches support far more efficient voice traffic than traditional dedicated telephony, reducing operational cost.	Server and LAN infrastructure investment at most companies has largely outpaced traditional circuit switched voice communications systems, making this much newer than existing PBX systems. Opportunity exists for IP telephony to leverage these platforms.
The fundamental con-cern for IP telephony is voice quality.	Conversation de-grades only when one-way delay exceeds 150 ms.	About half of normal voice interaction is silence. This means that 50% of the capacity of the traditional TDM network remains unused due to silence alone.
The traditional PBX has high reliability through stable components and built in manufacturer redundancy.	The traditional PBX deliver zero vendor redundancy and is inflexible and prohibitive to extend.	Eliminating the separate telecommunications network, and instead managing PBX servers in the same LAN cabinets and on the same operating systems as the data network, creates significant advantages in reliability, maintenance, and ease of administration.

Asterisk™ Features

Asterisk-based telephony solutions offer a rich and flexible feature set. Asterisk offers both classical PBX functionality and advanced features, and interoperates with traditional standards-based telephony systems and Voice over IP systems.

Call Features

* ADSI On-Screen Menu System
* Alarm Receiver
* Append Message
* Authentication
* Automated Attendant
* Blacklists
* Blind Transfer
* Call Detail Records
* Call Forward on Busy
* Call Forward on No Answer
* Call Forward Variable
* Call Monitoring
* Call Parking
* Call Queuing
* Call Recording
* Call Retrieval
* Call Routing (DID & ANI)
* Call Snooping
* Call Transfer
* Call Waiting
* Caller ID
* Caller ID Blocking
* Caller ID on Call Waiting
* Calling Cards
* Conference Bridging
* Database Store / Retrieve
* Database Integration
* Dial by Name
* Direct Inward System Access
* Distinctive Ring
* Distributed Universal Number Discovery (DUNDi™)
* Do Not Disturb
* E911
* ENUM
* Fax Transmit and Receive (3rd Party OSS Package)
* Flexible Extension Logic
* Interactive Directory Listing
* Interactive Voice Response (IVR)
* Local and Remote Call Agents
* Macros
* Music On Hold
* Music On Transfer
    - Flexible Mp3-based System
    - Random or Linear Play
    - Volume Control
* Predictive Dialler
* Privacy
* Open Settlement Protocol (OSP)
* Overhead Paging
* Protocol Conversion
* Remote Call Pickup
* Remote Office Support
* Roaming Extensions
* Route by Caller ID
* SMS Messaging
* Spell / Say
* Streaming Media Access
* Supervised Transfer
* Talk Detection
* Text-to-Speech (via Festival)
* Three-way Calling
* Time and Date
* Transcoding
* Trunking
* VoIP Gateways
* Voicemail
    - Visual Indicator for Message Waiting
    - Stutter Dial tone for Message Waiting
    - Voicemail to email
    - Voicemail Groups
    - Web Voicemail Interface
* Zapateller

Computer-Telephony Integration

* AGI (Asterisk Gateway Interface)
* Graphical Call Manager
* Outbound Call Spooling
* Predictive Dialler
* TCP/IP Management Interface

Scalability

* TDMoE (Time Division Multiplex over Ethernet)
    - Allows direct connection of Asterisk PBX
    - Zero latency
    - Uses commodity Ethernet hardware
* Voice-over IP
    - Allows for integration of physically separate installations
    - Uses commonly deployed data connections
    - Allows a unified dialplan across multiple offices

Codecs

* ADPCM
* G.711 (A-Law & μ-Law)
* G.723.1 (pass through)
* G.726
* G.729 (through purchase of commercial license through Digium)
* GSM
* iLBC
* Linear
* LPC-10
* Speex

Protocols

* IAX™ (Inter-Asterisk Exchange)
* H.323
* SIP (Session Initiation Protocol)
* MGCP (Media Gateway Control Protocol)
* SCCP (Cisco® Skinny®)

Traditional Telephony Interoperability

* E&M
* E&M Wink
* Feature Group D
* FXS
* FXO
* GR-303
* Loopstart
* Groundstart
* Kewlstart
* MF and DTMF support
* Robbed-bit Signaling (RBS) Types

PRI Protocols

* 4ESS
* BRI (ISDN4Linux)
* DMS100
* EuroISDN
* Lucent 5E
* National ISDN2
* NFAS

Asterisk™ Architecture

Asterisk is carefully designed for maximum flexibility. Specific APIs are defined around a central PBX core system. This advanced core handles the internal interconnection of the PBX, cleanly abstracted from the specific protocols, codecs, and hardware interfaces from the telephony applications. This allows Asterisk to use any suitable hardware and technology available now or in the future to perform its essential functions, connecting hardware and applications.

The Asterisk core handles these items internally:

* **PBX Switching** - The essence of Asterisk, of course, is a Private Branch Exchange Switching system, connecting calls together between various users and automated tasks. The Switching Core transparently connects callers arriving on various hardware and software interfaces.

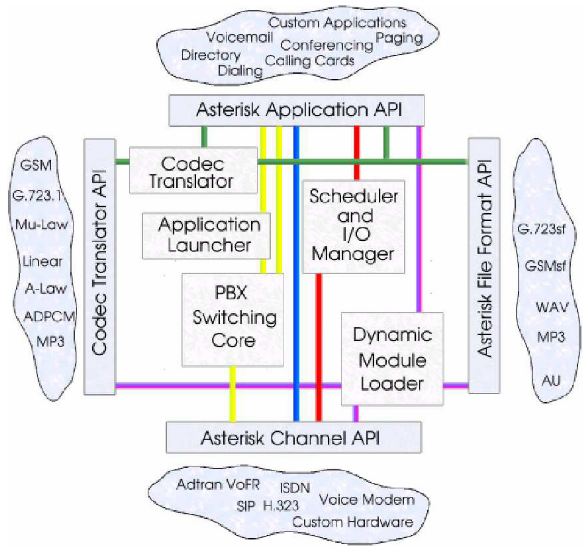
* **Application Launcher** - launches applications which perform services for uses, such as voicemail, file playback, and directory listing.

* **Codec Translator** - uses codec modules for the encoding and decoding of various audio compression formats used in the telephony industry. A number of codecs are available to suit diverse needs and arrive at the best balance between audio quality and bandwidth usage.

* **Scheduler and I/O Manager** - handles low-level task scheduling and system management for optimal performance under all load conditions.

Loadable Module APIs

Four APIs are defined for loadable modules, facilitating hardware and protocol abstraction. Using this loadable module system, the Asterisk core does not have to worry about details of how a caller is connecting, what codecs are in use, etc.



* **Channel API** - the channel API handles the type of connection a caller is arriving on, be it a VoIP connection, ISDN, PRI, Robbed bit signalling, or some other technology. Dynamic modules are loaded to handle the lower layer details of these connections.

* **Application API** - the application API allows for various task modules to be run to perform various functions. Conferencing, Paging, Directory Listing. Voicemail, In-line data transmission, and any other task which a PBX system might perform now or in the future are handled by these separate modules.

* **Codec Translator API** - loads codec modules to support various audio encoding and decoding formats such as GSM, Mu-Law, A-law, and even MP3.

* **File Format API** - handles the reading and writing of various file formats for the storage of data in the file system.

Using these APIs Asterisk achieves a complete abstraction between its core functions as a PBX server system and the varied technologies existing (or in development) in the telephony arena. The modular form is what allows Asterisk to seamlessly integrate both currently implemented telephony switching hardware and the growing Packet Voice technologies emerging today. The ability to load codec modules allows Asterisk to support both the extremely compact codecs necessary for Packet Voice over slow connections such as a telephone modem while still providing high audio quality over less constricted connections.

The application API provides for flexible use of application modules to perform any function flexibly on demand, and allows for open development of new applications to suit unique needs and situations. In addition, loading all applications as modules allows for a flexible system, allowing the administrator to design the best suited path for callers on the PBX system and modify call paths to suit the changing communication needs of a going concern.

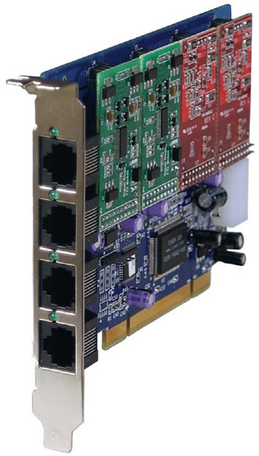
Asterisk™ Hardware

Digium TDM400P

Scalable and Effective SOHO Solution

The Wildcard TDM400P is a half-length PCI 2.2-compliant card that supports FXS and FXO station interfaces for connecting analogue telephones and analogue POTS lines through a PC. Using Digium's TDM hardware, Open Source Asterisk PBX software, and a standard PC, users can create a Small Office Home Office (SOHO) telephony environment which includes all the sophisticated features of a high-end PBX/Voicemail platform.

The TDM400P takes the place of an expensive channel bank and brings the system price point to the lowest in the industry. The FXO and FXS modules are interchangeable to create various combinations of interfaces. To scale this solution, just add additional TDM400P cards. This revolutionary solution has an unprecedented price point in the industry.



Target Applications

* Small Office Home Office (SOHO) applications
* Gateway Termination to Analog Telephones
* Add Inexpensive Analog Phones to Existing PBXs
* Wireless Point-to-Point Applications between Asterisk Servers

Services & Features

* Caller ID and Call Waiting Caller ID
* ADSI Telephones
* PCI Half-length Slot
* RJ-11C Connector

Standard Configurations

* TDM10B: 1-port FXS bundle
* TDM40B: 4-port FXS bundle
* TDM01B: 1-port FXO bundle
* TDM04B: 4-port FXO bundle
* TDM11B: 1-port FXS &1-port FXO bundle
* TDM22B: 2-port FXS &2-port FXO bundle
* TDM31B: 3-port FXS &1-port FXO bundle
* Other configurations available on request

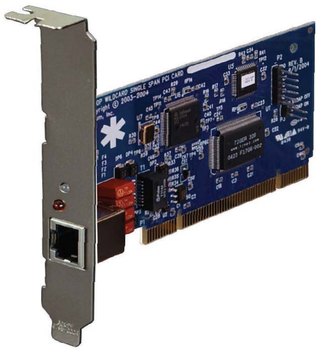
Digium Wildcard TE110P

High-performance and Cost-effective T1 and E1

The Wildcard TE110P brings a high-performance, cost-effective, and flexible single span togglable T1/PRI or E1/PRI interface to the Digium line-up of telephony interface devices. The TE110P is a compact and powerful interface supporting voice and data transmission over T1, E1, and Primary Rate ISDN (PRI) connections.

The TE110P supports industry-standard telephony and data protocols, including Robbed Bit Signalling (RBS), GR-303, and PRI protocols including NFAS (Non-Facility Association Signalling) for voice and Cisco HDLC, PPP and Frame Relay for data transmission. A low profile, half-length PCI form factor allows this device to fit within a 2U rack mount case or equivalent chassis, offer excellent density for call center, service provider, and other space-sensitive applications.

Of course, the TE110P is fully supported by Digium's open source Asterisk PBX software. Used in conjunction with Asterisk, the TE110P offers the power to create a seamless network, interconnecting traditional telephony systems with the emerging Voice-over IP technologies. The TE110P can be used to deliver a wide range of PBX and IVR services to the network or handset including Voicemail, Call Conferencing, Three-Way Calling, and VoIP Gateways.



Target Applications

* Packet Voice Gateways and Switches
* Calling Card Services
* One Number Services
* Message Services
* Conferencing
* Customized and Web Telephony
* Voice/Data Integration
* Futureproof PBX
* ISDN Remote Access Servers

PRI Switch Capability

* AT&T 4ESS
* DMS 100
* Lucent 5E
* National ISDN 2
* Euro ISDN
* Network or CPE
* NFAS

RBS Voice Modes

* GR-303
* A-Law, μ-Law, and Linear
* Modes Supported
* E&M
* E&M Wink
* Feature Group D
* Groundstart (FXO and FXS)
* Loopstart (FXO and FXS) with
* Optional Disconnect Supervision

Data Modes

* SyncPPP (both Fixed and Dialup)
* Frame Relay
* Cisco HDLC

Digium Wildcard TE410P

Ultimate Density and Performance in both T1 and E1

The TE410P is the next generation of Digium hardware that improves performance and scalability through bus mastering architecture. The TE410P supports both E1 and T1 environments and is selectable on a per-card or per-port basis. This feature enables signaling translation between E1 and T1 equipment and allows inexpensive T1 channel banks to connect with E1 circuits. Because the TE410P improves I/O speed by up to 10 times, the result is reduced CPU usage and increased card density per server.

Digium has designed the TE410P to be fully compatible with existing software applications and it is fully integrated with the Asterisk Open Source PBX/IVR platform. Also, the open source driver supports an API interface for custom application development. With the combination of Digium Hardware and Asterisk software, numerous combinations of telephony configurations are possible. From the traditional PBX to VoIP Gateways, Digium solutions are paving the way for a new generation of worldwide communications.

The TE410P supports industry standard telephony and data protocols, including Primary Rate ISDN (both N. American and Standard Euro) protocol families for voice, PPP, Cisco, HDLC, and Frame Relay data modes. Both line-side and trunk-side interfaces are supported, also included are advanced call features.

Target Applications

* Legacy PBX/IVR services
* Voice-over Internet Protocol (VoIP) services
* Complex IVR Trees
* "Meet-me-Bridge" conferencing
* VoIP Gateways (supports SIP, H.323 and IAX)
* Calling Card Platforms
* Voice/Data Router (replace expensive routers)

PRI Switch Capability

* EuroISDN (PRI or PRA) - Q.931/Q.921
* AT&T 4ESS
* DMS 100
* Lucent 5E
* Network or CPE
* National ISDN 2

CAS Voice Modes

* Feature Group D
* E&M Wink
* A-Law, Mu-Law, and Linear Modes Supported

Data Modes

* SyncPPP (both Fixed and Dialup)
* Frame Relay
* Cisco HDLC
* Multi-Link PPP

The TE410P is for use only with a 3.3 Volt PCI slot. For use with 5.0 Volt PCI slots, the otherwise feature-identical TE405P should be considered.

Digium Wildcard TE411P

Ultimate Density and Performance with on-board echo cancellation

The TE411P is the next generation of Digium hardware that offers an onboard echo cancellation module. It supports E1, T1 and J1 environments and is selectable on a per-card or per-port basis.

The echo cancellation module supports all T1, E1 and J1 channels and improves voice quality in environments where software echo cancellation is not sufficient. The TE411P reduces CPU overhead required for software echo cancellation thereby, freeing resources for other processes, such as codec translation. By supporting 16ms with 128 channels or 64ms on 32 channels this card will perform in the most difficult of environments.

Digium has designed the TE411P to be fully compatible with existing software applications and it is fully integrated with the Asterisk Open Source PBX/IVR platform. Also, the open source driver supports an API interface for custom application development. With the combination of Digium Hardware and Asterisk software, numerous combinations of telephony configurations are possible. From the traditional PBX to VoIP Gateways, Digium solutions are paving the way for a new generation of worldwide communications.

The TE411P supports industry standard telephony and data protocols, including Primary Rate ISDN (both N. American and Standard Euro) protocol families for voice, PPP, Cisco, HDLC, and Frame Relay data modes. Both line-side and trunkside interfaces are supported, also included are advanced call features.

Target Applications

* Legacy PBX/IVR services
* Voice-over Internet Protocol (VoIP) services
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PRI Switch Capability

* EuroISDN (PRI or PRA) - Q.931/Q.921
* AT&T 4ESS
* DMS 100
* Lucent 5E
* Network or CPE
* National ISDN 2

CAS Voice Modes

* Feature Group D
* E&M Wink
* A-Law, Mu-Law, and Linear Modes Supported

Data Modes

* SyncPPP (both Fixed and Dialup)
* Frame Relay
* HDLC
* Cisco HDLC
* Multi-Link PPP

The TE411P is for use only with a 3.3 Volt PCI slot. For use with 5.0 Volt PCI slots, the otherwise feature-identical TE406P should be considered.

BeroNet BN4S0 ISDN Board

with 4 S/T Interfaces

The BN4S0 is an ISDN card with four S/T interfaces for Basic Rate Interface (BRI) based on Cologne Chip HFC-4S. It is designed for all kind of data applications and voice applications such as PC based PABX, VoIP gateways and ISDN monitoring/recording (possible with another card version with a Cologne Chip HFC-8S and different component insertion).

ISDN Conformity

The ISDN BN4S0 board family is compliant to the hardware specifications of the ISDN standards (I.430, CTR3). The BN4S0 board is certified by an authorized test laboratory (TÜV Rheinland Product Safety). A copy of the Layer 1 Measurement Report can be obtained upon request.

S/T Interface

* 4 BRI ports
* each port can be configured separately to TE/NT mode
* TE/NT mode setting can be detected by software for each port
* all ports of the card can be fed independently by one external power supply unit (permitted in NT mode only; unit optional available)
* short circuit protection for the ISDN cabling by fuses (no blowing, auto-recovery)
* line termination (100 Ω) is independently selectable for each port by DIP switch

PCI Interface

* PCI interface is suitable for 3.3V as well as 5V PCI 2.2 slots (5V to 3.3V regulator on board)

PCM Bus

* daisy chaining possible by two connectors (2x10 pins, 2.54cm pin pitch) on the card
* flat ribbon cable for connection of several cards optional available
* 2/4/8 Mbit/s data transfer rate
* Report can be obtained upon request.

Chipset

* Cologne Chip HFC-4S ISDN
* high precision 49.152 MHz quartz oscillator
* 512 bit x 8 serial EEPROM for the storing of PCI configuration information (e.g. PCI vendor or device ID)
* 512k x 8 SRAM for enlarging FIFO buffer (soldering option)

General Purpose I/O

* 4 dual LEDs at the slot bracket that can be controlled by software with possible states green, red, off for each LED (controllable via GPIO – general purpose input-output)
* 3-DIP switch can be checked by software for e.g. identification of the card (controllable via GPI - general purpose input)

BeroNet BN8S0 ISDN Board

with 8 S/T Interfaces

The BN8S0 is an ISDN card with eight S/T interfaces for Basic Rate Interface (BRI) based on Cologne Chip HFC-8S. It is designed for all kind of data applications and voice applications such as PC based PABX or VoIP gateways. The RJ45 jacks have a non-standard pinning (2 ISDN ports per connector). There are 4 transformers on the backside of the PCB so that the standard PCI dimensions are exceeded.

ISDN Conformity

The ISDN board family is compliant to the hardware specifications of the ISDN standards (I.430, CTR3).

S/T Interface

* 8 S/T interfaces: 2 ISDN ports per RJ45 jack. The inner 4 pins belong to an ISDN port (accessible with a standard ISDN cable) while the outer pins build an additional ISDN port (special cable or adapter required). So the cabling has to be split by an adapter to give access to all 8 S/T interfaces.
* each port can be configured separately to TE/NT mode
* all ports of the card can be fed independently by one external power supply unit (permitted in NT mode only; unit optional available)
* short circuit protection for the ISDN cabling by fuses (nonblowing, auto-recovery)
* line termination (100 Ω) is independently selectable for each port by DIP switch

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* PCI interface is suitable for 3.3V as well as 5V PCI 2.2 slots (5V to 3.3V regulator on board)

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* 2/4/8 Mbit/s data transfer rate

Chipset

* Cologne Chip HFC-8S ISDN
* high precision 49.152 MHz quartz oscillator
* 512 bit x 8 serial EEPROM for the storing of PCI configuration information (e.g. PCI vendor or device ID)

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