Product Line Overview

From the leader in network access, connectivity, VoIP and mobile video-surveillance gear

Smart\lode, VoIP solutions that are more than just talk,



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VoIP Starts with SmartNode...

SmartNode integrates IP and TDM communications for Enterprise and Carrier access networks, offering VoIP gateways combined with IP access routing, WAN transmission, and transcoding functionality. SmartNode scales from 1 to 2,048 VoIP or fax calls with various telephony interfaces including analog FXS/FXO and digital ISDN BRI, PRI, DS3 and STM-1.

Award-winning SmartNode™ equipment delivers state-ofthe art VoIP technology that integrates seamlessly with existing analog PSTN, digital ISDN, and IP infrastructures. Providing any-to-any multipath switching, SmartNode supports simultaneous SIP, H.323, ISDN and PSTN calling—plus T.38 SuperG3 FAX and modem over IP.

SmartNode's proven interoperability with all major brands of softswitches and IP PBXs makes it easy for carriers and enterprises to deploy future-proof VoIP systems quickly and profitably. As VoIP Gateway pioneers since 1998, tens of thousands of SmartNode™ products are up and running in enterprise and carrier networks worldwide, and have proven interoperability with all major IP PBX, Softswitch and VoIP service providers (visit www.patton.com/partners for details).

✓ Telephony interfaces on Gateways include:

- 1 to 32 FXS/FXO
- 1 to 8 ISDN BRI
- 1 to 64 T1/E1/PRI
- 1 to 3 DS3
- 1 STM1
- ✓ 1 to 2016 VoIP or fax call capacity



✓ Supports simultaneous SIP, H.323, ISDN and PSTN calling—plus T.38 faxing

Why SmartNode?

- ✓ Patton Quality and Reliability
- ✓ Unrivalled Customer Support
- ✓ Free Support and Software Upgrades
- ✓ Robust Enterprise Feature Sets and Functionality
- ✓ Proven Interoperability
- ✓ Swiss Engineered. Made in the USA.





SmartNode[™] Branch eXchange (SNBX) 3CX Phone System and Windows 7 Pre-loaded



A fully-loaded Enterprise IP PBX appliance made easy for any business. Supporting up to 64 simultaneous calls, the SNBX is plug-and-play to fit any network—bringing you Unified Communications without breaking the bank.

- Pre-installed with 3CX Phone System and Windows 7
 - Up to 64 Concurrent Calls
 - Up to 500 SIP Extensions
 - Built-in Config Tool for SmartNode Gateways



SmartNode Software

Depending on the model, SmartNode comes with three different software packages developed for different applications:

- SmartLink™—Basic SoHo feature sets. Running on the M-ATA.
- SmartWare[™]—Small/medium enterprise feature sets (SIP, PSTN, Routing, QoS, transcoding, etc.). Running on SmartNode 4XXX/5XXX models.
- TrinityTM—Small/medium enterprise feature sets (including next-gen features like TLS and SRTP, stateful firewall, TACACS+, etc.). Running on SN4660/70, SN4970/80/90 and SN5480/90.
- SmartMedia[™]—Large enterprise and carrier network feature sets (SIP, PSTN, Routing, QoS, transcoding, SS7, etc.). Running on SmartNode 10XXX models.

(See SmartNode™ Software Features on pg 18, for more details.)

FXS/FXO Gateways							
	M-ATA	SN4110	SN4520	SN4830	SN4300	SN4400	SN4900*
Software	SmartLink	SmartWare	SmartWare	SmartWare	SmartWare	SmartWare	SmartWare
Telephony Interfaces	FXS	FXS & FXO	FXS & FXO	FXS & FXO	FXS or FXO	FXS or FXO	FXS or FXO
# of Telephony Ports	1	2, 4, 6 or 8	2, 4, 6 or 8	2, 4, 6 or 8	12, 16, 24 or 32	12, 16, 24 or 32	12, 16, 24 or 3
VoIP Gateway: Converts TDM to IP	Yes	Yes	Yes	Yes	Yes	Yes	Yes
IP Router: IP Routing, QoS, VPN, etc.	No	No	Yes	Yes	No	Yes	Yes
Number of Ethernet Ports	1	1	2	2	1	2	2
WAN Access	No	No	No	G.SHDSL Serial X.21 Serial V.35 ADSL	No	No	(Optional) Serial X.21 Serial V.35

^{*}The SN4900 differs from the SN4400 in that it has a redundant internal power supply, optional WAN access, and 0 to 50° C operating temperature

ISDN BRI Gateways						
	SN-DTA	SN4120	SN4630	SN4650	SN4660	SN4670
Software	SmartWare	SmartWare	SmartWare	SmartWare	IPv6 SmartWare/Trinity	IPv6 SmartWare/Trinity
Telephony Interfaces	BRI (NT or NT+TE)	BRI (TE)	BRI	BRI	BRI with FXS/FXO	BRI with FXS/FX
# of Telephony Ports	1 or 2	1 or 2	3 or 5	3 or 5	Up to 12	Up to 12
# of VolP Channels	2 or 4	2 or 4	4 or 8	4 or 8	Up to 24	Up to 16
VolP Gateway: Converts TDM to IP	Yes	Yes	Yes	Yes	Yes	Yes
IP Router: IP Routing, QoS, VPN, etc.	No	No	Yes	Yes	Yes	Yes
Number of Ethernet Ports	1	1	2	1	4	4
WAN Access	No	No	No	ADSL, G.SHDSL Serial X.21 Serial V.35	No	Fiber, EFM ADSL, G.SHDSL



		1/E1/PRI Ga	teways				
	SN4970	SN4980	SN4990	SN10100		SN10200	
Software	SmartWare/Trinity	IPv6 SmartWare/Trinity	IPv6	SmartMedia		SmartMedia	
Telephony Interfaces	T1/E1/PRI	T1/E1/PRI	T1/E1/PRI	T1/E1/PRI	T1/E1/PRI	DS3	STM-
# of Telephony Ports	1 or 4	1 or 4	1 or 4	4 to 8	16 to 64	1, 2, or 3	1
# of VoIP Channels	15, 24, 30, 48, 60, 96 or 120	15, 24, 30, 48, 60, 96 or 120	15, 24, 30, 48, 60, 96 or 120	up to 256		up to 2,048	
VolP Gateway: Converts TDM to IP	Yes	Yes	Yes	Yes		Yes	
IP Router: IP Routing, QoS, VPN, etc.	No	Yes	Yes	No		No	
Number of Ethernet Ports	1	2	2	2		2	
WAN Access	No	No	Fiber, ADSL, EFM, G.SHDSL, Serial X.21, Serial V.35	No		No	
Transcoding: Interconnect multiple VoIP networks	No	Optional	Optional	Yes		Yes	

Note: SmartNode 4970/4980/4990 are next-generation models of the SmartNode 4940/4950/4960.

	SN5200	SN5480	SN5490
Software	SmartWare	SmartWare/Trinity	SmartWare/Trinity
Telephony Interfaces	None	None	None
SIP Sessions	Up to 32	Up to 196	Up to 196
Transcoded Calls	N/A	Up to 64	Up to 64
VoIP Gateway: Converts TDM to IP	No	No	No
IP Router: IP Routing, QoS, VPN, etc.	Yes	Yes	Yes
Number of Ethernet Ports	5	2	2
WAN Access	X.21	No	Fiber, G.SHDSL, EFM, X.21
Transcoding: Interconnect multiple VoIP networks	No	Yes	Yes

For more information visit: www.patton.com/smartnode





Enterprise Solutions

SIP Trunking with a Legacy PBX

VoIP service offers most users substantial savings (sometimes up to 50–70%), scalability and flexibility. But many businesses are tied to existing telephone equipment because of the cost of installing new IP equipment, retraining employees, and not realizing investment in their current equipment.

The solution? Patton provides their PBX a gateway to VoIP.

Using a Patton SmartNode Gateway you can provide the cost savings and added functionality of SIP Trunking to your existing legacy PBX. Patton's SmartNode supports all varieties of legacy PBXs, regardless of the type of trunk supported. This includes analog, BRI, PRI, DS3 or STM-1 connections.



Business Case

An outbound long-distance call center has an Avaya legacy PBX system. Currently they have a 23-channel PRI with from a local Carrier and 2 PSTN lines for fax machines running them at \$1,200 to \$1,300 per month.

They were informed that they could cut their phone bill in half by a VoIP service provider while also adding next-generation IP features. However, moving to the VoIP service provider would mean throwing away their perfectly functional Avaya PBX and 100+ analog phones (a \$50,000 investment 4 years ago) and re-investing another \$50,000 in an IP PBX and IP phones. Not to mention, they would have to re-wire the office for Ethernet.

There's no justification for spending \$100,000 to save \$600 a month (ROI in 14 years).

But by installing a PRI SmartNode for roughly a few thousand dollars, this call center can connect their existing phone equipment to a VoIP service provider and achieve the \$600 per month savings after only 3 months when the hardware is paid off. Not only do they achieve \$7,200 in savings each year, they don't have to teach their employees how to use a new phone system.

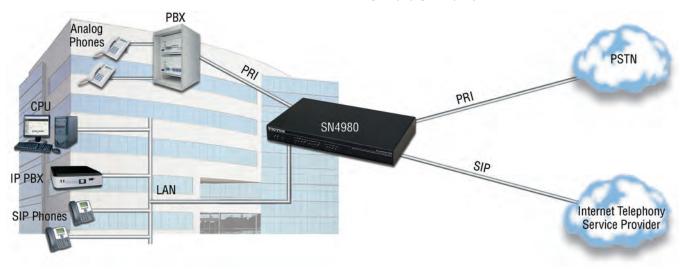


Enterprise Solutions

Migration (Legacy PBX to IP PBX)

Why throw away perfectly functional phone equipment? A SmartNode can allow you to utilize your existing phone equipment while migrating to IP equipment at your own pace. As your business grows, start investing in next-generation IP telephones while at the same time utilizing your perfectly good analog/digital phone equipment until it needs to be replaced.

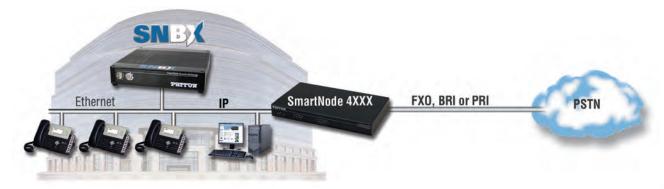
- Allow a smooth migration from old equipment to new equipment
- Stage your investments
- Preserve investment in existing infrastructure and extend useful life of capital equipment
- Any-to-any flexible call routing between legacy PBX, IP PBX, PSTN and SIP Trunks.



Gateway to the PSTN for IP PBX

Many businesses are migrating to feature-rich IP PBX phone systems but wish to keep their existing traditional phone service provider because they are satisfied with their current service, stuck in a contract or possibly don't trust VoIP. In these cases, offering Patton's SmartNode Gateways in conjunction with an all-IP setup enable you to provide the benefits of SIP while allowing businesses to continue using their trusted PSTN lines and existing telephony service providers.

For businesses uneasy about switching to a VoIP service provider, you can use a SmartNode to reliably implement VoIP with a phased approach. First, install your IP setup with a SmartNode to use your existing POTS lines. Once passing calls successfully, you can add a basic VoIP account (some providers even offer a free month trial) to be used for outbound long-distance calls only. If the business is satisfied, you can switch them over completely to the VoIP provider and possibly keep one or two POTS lines as a fallback line using the SmartNode (see next section).



Enterprise Solutions

SIP Trunking with Fallback/Survivability

The best VoIP providers provide 99% reliability given the necessary bandwidth requirements. A SmartNode ensures businesses telephony continuity by setting their VoIP phone network up with survivabil-

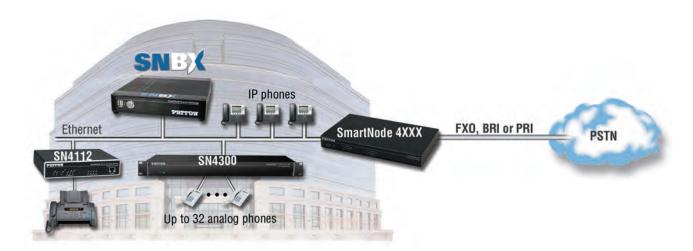
ity. A SmartNode VoIP Gateway can be used to fallback to your PSTN lines (99.999% reliability) in cases where the Internet telephony service provider or Internet connection goes down.



Connecting Legacy Terminals

When migrating to a VoIP network, many businesses find it difficult to replace every legacy terminal with a brand new IP terminal all at once. The SmartNode Gateway gives SMBs and hotels the ability to connect this existing equipment to their IP phone system.

As the world transitions to VoIP, fax has become a particularly hot topic as it has proven to be an indispensable technology. When sending faxes over a VoIP network, even the slightest interruption in the data stream will drop a fax. Hence, to ensure reliable fax delivery, SmartNode offers T.38 and G.711 fax-over-IP.



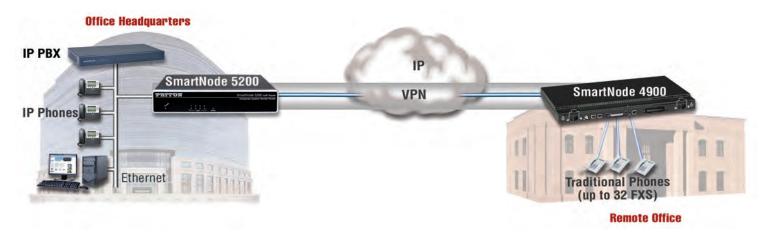


Enterprise Solutions

Connecting Remote Offices

The SmartNode Gateway can be used to connect analog, ISDN and IP telephony from a remote office to your IP PBX phone system in your headquarters.

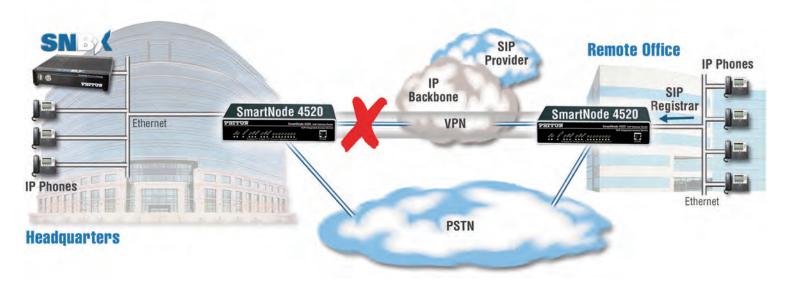
With integrated routing, QoS and VPN features, the SmartNode can ensure a secure connection and maximum voice quality between offices.



Remote Office Survivability

A SmartNode provides a SIP Registrar for your IP phones in your remote office to register with the SmartNode directly rather than just the IP PBX in your headquarters. This way, in the case where

you lose your connection to your headquarters, the remote office can act independently making and receiving calls over PSTN lines.

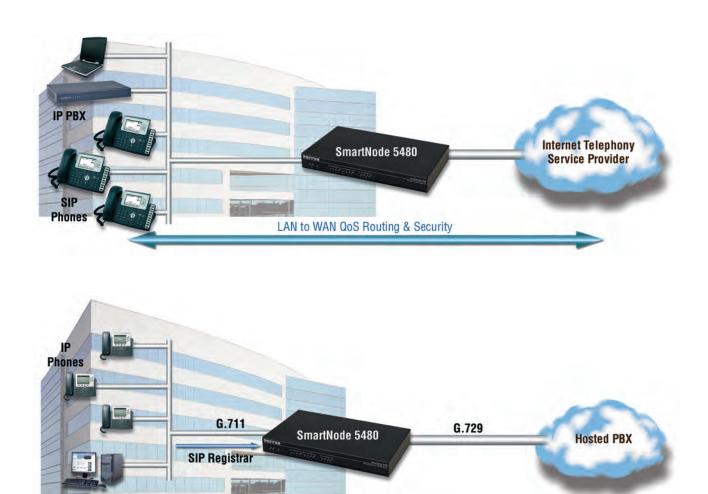


Enterprise Solutions

Enterprise Session Border Routers

For all-IP environments, a SmartNode Enterprise Session Border Router (ESBR) can be placed at the edge of your network to provide several key voice quality, survivability, and security features:

- Back-to-back-user-agent (B2BUA): Isolates your network from the outside world to hide your private addresses and network topology for security purposes.
- Transcoding: Provide any-to-any codec conversion allowing efficient use of bandwidth and improved voice quality. Reduce WAN-access bandwidth requirements by converting high band-
- width G.711 codec on your LAN to low bandwidth G.729 for WAN transport. Transcoding also provides seamless integration between multiple VoIP providers using different VoIP codecs.
- Quality-of-Service (QoS): Ensures voice quality by managing data communications so that voice packets and other real-time data are prioritized.
- **SIP Registrar**: Enables any registration and authentication scenario between IP PBX on one side and SIP Trunk provider on the other.



LAN to WAN QoS Routing & Security



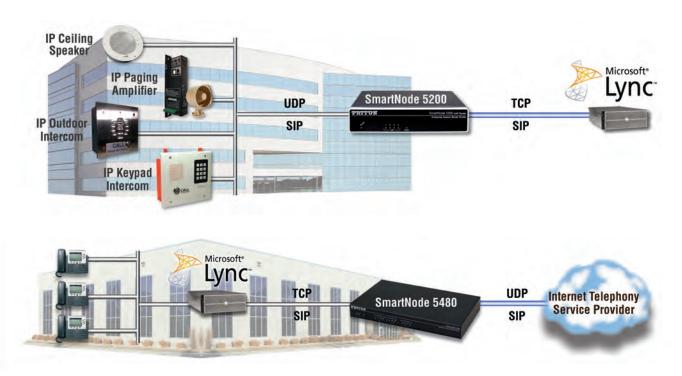
The Gateway to Lync



ertified for Microsoft Lync Server 2010 and 2013, SmartNode provides a gateway to Lync for **all** pieces of your voice infrastructure including:

- Legacy (PBX, phones, fax, PSTN lines)
- Non-certified SIP (SIP trunks, IP PBX, IP phones, and IP paging equipment)

Not only can you integrate legacy equipment with Lync, a SmartNode can be used to provide the translation between your standard SIP that uses UDP signaling and Microsoft Lync which uses TCP signaling.



Carrier Solutions

Carrier CPE: Legacy PBX

As a growing number of Service Providers are offering VoIP services, they face the challenge of connecting a wide array of legacy PBX systems. The SmartNode will always provide the same SIP to the service provider regardless of the type of legacy PBX attached to it. Patton's

SmartNode supports all varieties of legacy PBXs, regardless of the type of trunk supported. This includes Analog, BRI, PRI, DS3 or STM-1 connections.



Carrier CPE: IP PBX

The major problem for Carriers is the wide selection of IP PBXs available on the market today, and this number is growing every day. And even though each IP PBX provider says they are SIP standards based, each system does something different with the SIP protocol. This means the service provider must have a unique softswitch profile for *each* type of IP PBX system which ties directly to OAM (operations, administration and maintenance) costs.

This device receives SIP requests from the customer premise equipment and reformats the SIP to meet the requirements for the service provider softswitch.

This SIP normalization standardizes setup because no matter what vendor IP PBX is used at the customer premise, the SIP presented to the service provider is *always* the same—eliminating the need for individual service profiles for each vendor's IP PBX.

Also moving the typical SBC features of these VoIP Routers from the Carrier network to the customer premise provides the customer with QoS, transcoding and a security demarcation point.





Carrier Solutions

Carrier CPE: Hosted PBX

With the wide variety of SIP phones available on the market today, the same number of softswitch SIP profiles is required to support each vendor's SIP phone. By utilizing the SmartNode Enterprise Session Border Router with the SIP Back-to-Back user agent, a single softswitch profile can be used to support all varieties of SIP phones. This will allow control of OAM costs by not utilizing support personnel's valuable time to develop and implement multiple SIP phone profiles.

A second problem is that most SIP phones require registration to some type of softswitch or Session Border Router to be able to

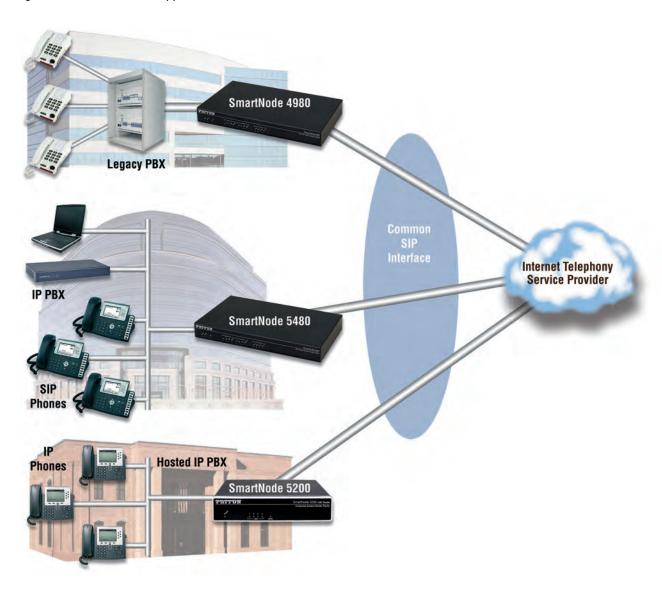
function. Most SBCs on the market today support a SIP registrar function to allow SIP to continue operation in the case of a softswitch failure. But in the case of a broadband connection failure, the SIP phones will not be able to reach the SBC to registrar and will not function. The solution for this problem is to move that SBC functionality to the customer premises. By complementing the SBC functionality with the SmartNode ESBR that supports SIP registra tion, all SIP phones will be able to operate locally without the broadband connection.



Carrier CPE: Connect Any Enterprise

The Patton SmartNode[™] product line supports IP PBX and hosted PBX, as well as legacy PBX systems. The feature-rich capabilities of the Patton SmartNode VoIP CPE means the service provider can have a single vendor solution that supports all VoIP services to the

end user. By providing a single and consistent SIP interface for all three of the above services, Patton can help you simplify VoIP deployment while controlling your OAM costs.



The Patton SmartNode $^{\text{m}}$ solution operates under a single software platform allowing ease of installation and configuration, regardless of the application being supported. And the specific features explained above are available on **all** SmartNode products. Other

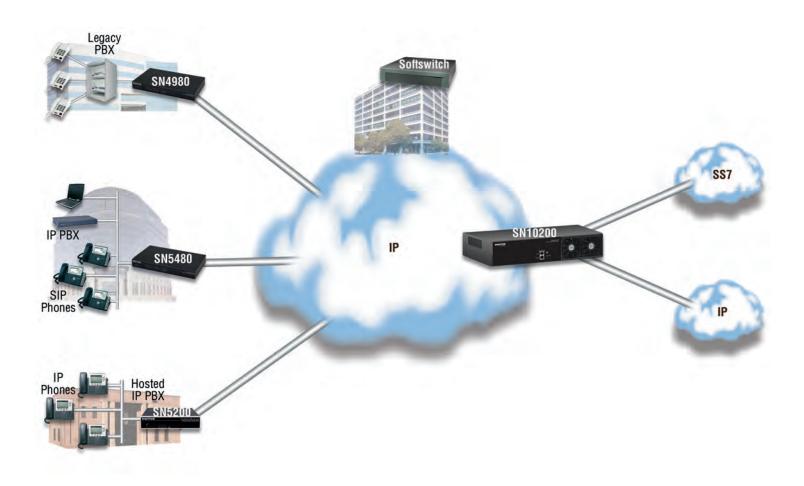
vendors cannot support these features because of limited feature sets or by providing a gateway only solution that cannot support IP-to-IP applications.



Carrier Network

Service providers are adding VoIP capabilities to their networks, whether to reduce costs when interconnecting with other carriers, to cost-effectively build out their network footprints, or simply to transport voice traffic across their IP backbones. This can be best accomplished using a SmartNode 10XXX Series media gateway, that enables the delivery of VoIP services by bridging voice traffic

between the public switched telephone network (PSTN)—based on time-division multiplexing (TDM)—and IP networks such as the Internet. Whether sitting at the network core or at the edge, SmartNode media gateways enable service providers to introduce VoIP into their networks while maintaining the quality and the reliability of traditional TDM networks.



	Sma	rtNode™ Software Featur	es	
	SmartLink (SmartLink 4022 & M-ATA)	SmartWare (SmartNode 4000/5000 Series)	Trinity (SmartNode 4000/5000 Series)	SmartMedia (SmartNode 10000 Series)
VoIP Features	,	,	,	
Signaling Type				
SIP v2	✓	✓	✓	✓
H.323 v4	N/A	✓	Roadmap	External H.323 (Converter box)
SIGTRAN	N/A	N/A	N/A	(IUA, M2UA, M2PA, M3UA
Voice Codec				
G.711	✓	✓	✓	(Universal Channel)
G.722.2 (AMR-WB)	N/A	N/A	N/A	√
G.723	1	√	,	✓
U.723	•	V	✓	(Universal Channel)
G.726	✓	✓	✓	(Universal Channel)
G.728	N/A	N/A	N/A	1
G.729AB	✓	✓	✓	(Universal Channel)
G.729EG	N/A	N/A	N/A	1
AMR	N/A	N/A	N/A	✓
GSM-FR	N/A	N/A	N/A	✓
GSM-EFR	N/A	N/A	N/A	✓
GSM-HR	N/A	N/A	N/A	✓
EVRC	N/A	N/A	N/A	✓
QCELP	N/A	N/A	N/A	✓
T.38 V.17	✓	✓	✓	(Universal Channel)
T.38 V.34	N/A	✓ (G.711 Bypass)	✓ (G.711 Bypass)	✓
G.711 Bypass	(Not Modem)	✓	1	(Auto-switch to Modem Passthrough)
RFC4040 (data passthrough)	N/A	✓	✓	✓
Basic Call Features				
AOC (Advice of charge)	N/A	√	√	1
Call-Hold	IN/A	<i>J</i>	√	N/A
		•	•	√
Call-Routing	N/A	~	✓	(Requires Ruby Scripting
Call-Transfer	✓	✓	✓	Partial (REFER [relay/rerou ing], ISDN AT&T 4ESS/5ES Toll free transfer connect
Call-Waiting	✓	✓	✓	N/A
Dejitter Buffer	✓	✓	✓	✓
Echo Cancellation	✓	✓	✓	(128 msec Universal)



SmartNode™ Software Features (continued)						
	SmartLink	SmartWare	Trinity	SmartMedia		
11.040	· ·		(SmartNode 4000/5000 Series) N/A	(SmartNode 10000 Series)		
H.248	N/A	N/A N/A	N/A N/A	√ N/A		
Local 3 Party Conference MGCP	1	N/A	N/A	N/A		
NAT Traversal	1					
	N/A	√	√	√ N/A		
SIP Registrar		√	√			
SIP Registree STUN	1	✓ N/A	√ N/A	√ N/A		
31011	✓	IV/A	IV/A	IV/A		
SIP Related Compliance						
RFC2327 SDP	√	√	J	1		
RFC2833 (DTMF Relay)	y	√	y	1		
	√	•	<u> </u>	1		
RFC2976 (SIP INFO Messages)	(DTMF Tones)	✓	✓	(DTMF Tones)		
RFC3261 (SIP)	✓	✓	✓	✓		
RFC3262 (PRACK)	N/A	✓	✓	✓		
RFC3264	✓	✓	✓	('indicating' capabilities not supported)		
RFC3265 (SIP Notify Messages)	N/A	✓	✓	N/A		
RFC3311 (SIP Update Method)	N/A	✓	✓	(For session timer refresh)		
RFC3323 (P-Header)	N/A	✓	✓	√		
RFC3325 (A-Header)	N/A	✓	✓	(No Preferred-Identity)		
RFC3372 (SIP-T, SS7 to SIP)	N/A	N/A	N/A	(implemented SIP-I and as a result met RFC Standards)		
RFC3398 (SIP to ISDN)	N/A	N/A	N/A	1		
RFC3515 (Refer Method)	N/A	✓	✓	✓		
RFC3581 (rport)	N/A	✓	✓	✓		
RFC3665 SIP Basic Call Flow	✓	✓	✓	✓		
RFC3666 PSTN Call Flow	✓	✓	✓	✓		
AoR	N/A	N/A	N/A	√		
RFC3842 (MWI)	✓	✓	✓	N/A		
RFC3891 (Replaces Header)	N/A	✓	✓	✓		
RFC3892 (Referred-By Mechanism)	N/A	✓	✓	N/A		
RFC4028 (Session Timers in the SIP)	N/A	✓	✓	✓		
RFC4240 (Media Mixing Server)	N/A	✓	✓	N/A		
RFC4694 (LNP for tel URI)	N/A	N/A	N/A	√ (For rn/npdi Relay to SS7 only)		
RFC5806 (SIP Diversion header)	N/A	N/A	N/A	(Unconditional Forward Scenario)		
SIP-I (SS7 to SIP)	N/A	N/A	N/A	√		

		™ Software Features <i>(cd</i>		
	SmartLink (SmartLink 4022 & M-ATA)	SmartWare (SmartNode 4000/5000 Series)	Trinity (SmartNode 4000/5000 Series)	SmartMedia (SmartNode 10000 Serie
TDM	(SHIAITEHK 4022 & WI-ATA)	(Silial livoue 4000/3000 Selles)	(Silialtivode 4000/3000 Selles)	(Sinartivode 10000 Serie
Signaling				
4ESS and 5ESS	N/A	N/A	N/A	1
CC-FXO	N/A	IV/A	Roadmap	N/A
CC-FXS	N/A	✓	Roadmap	N/A
DMS-100	N/A	√	Roadmap	√ ·
DMS-250	N/A	N/A	N/A	√
DSS1	N/A	√	√	4
NI-2	N/A	√	y	4
NTT INSnetI500	N/A	N/A	N/A	√
Q.SIG	N/A	IV/A	√	N/A
Q.OIQ	IV/7\	V	V	IV/A
R2	N/A	✓	Roadmap	(Scripteable Variants) RJ/TE Only
RBS	N/A	✓	Roadmap	N/A (Hardware Supported
SS7	N/A	N/A	N/A	✓
V5.2	N/A	N/A	N/A	N/A
Data Connectivity/IP				
Standard QoS	N/A	✓	✓	N/A
VPN Capabilities	N/A	√ √	<i>J</i>	N/A
ACL	N/A	√ ·	<i>J</i>	N/A
DownStream QoS	N/A	√	√	N/A
IP Router	✓	✓	✓	N/A
IPv6	N/A	N/A	Roadmap	("WIP for VoIP Calls, alre works for data connectivi
Frame Relay	N/A	✓	Roadmap	N/A
DSL	N/A	✓	Roadmap	N/A
V.35	N/A	✓	Roadmap	N/A
X.21	N/A	✓	Roadmap	N/A
VLAN support (802.1p/Q)	N/A	✓	√	N/A
IP Tables	N/A	N/A	✓	N/A
DHCP server/DHCP client	✓	✓	✓	(DHCP Client)
Stateful Firewall	N/A	N/A	✓	N/A
Ethernet Bridging	N/A	N/A	✓	N/A
Syslog Client	✓	✓	✓	N/A
Related RFC Compliance				
RFC791 (IP)	✓	✓	✓	√
RFC792 (ICMP)	√ ·	√	√	1
RFC1035 (DNS)	√ ·	√	√	1
RFC1058 (RIP)	√	√	√	N/A



SmartNode™ Software Features (continued)						
	SmartLink	SmartWare	Trinity	SmartMedia		
	<u>'</u>	(SmartNode 4000/5000 Series)	(SmartNode 4000/5000 Series)	(SmartNode 10000 Series)		
RFC1349 (ToS)	N/A	✓	✓	✓		
RFC1483 (MPE ATM)	N/A	✓	Roadmap	N/A		
RFC1490 (MPI over FrameRelay)	N/A	✓	Roadmap	N/A		
RFC1631 (NAT)	✓	✓	✓	N/A		
RFC1661 (PPP)	✓	✓	✓	N/A		
RFC2030 (SNTP)	✓	✓	✓	(Control host)		
RFC2132 (DHCP Server)	1	✓	✓	N/A (Atom supports but not used in such a way)		
RFC2401 (IPsec)	N/A	✓	Roadmap	(Atom host only. Hardware Supported on VoIP)		
RFC2453 (RIPv2)	✓	✓	✓	N/A		
RFC2571 (SNMP)	✓	✓	✓	✓		
RFC4004 (TACACS+)	N/A	N/A	✓	N/A		
RFC2865 (RADIUS)	N/A	✓	Roadmap	✓		
RFC3022 (NAT)	✓	✓	✓	N/A		
RFC4251 (SSH)	N/A	✓	1	(Control host)		
RFC4252 (SSH)	N/A	✓	1	(Control host)		
RFC4253 (SSH)	N/A	✓	1	(Control host)		
RFC4254 (SSH)	N/A	✓	✓	(Control host)		
Call Routing Functions						
Regex routing	N/A	√	1	1		
Scripting routing	N/A	N/A	N/A	1		
List-driven longest-prefix match routing (NPANXX)	N/A	N/A	N/A	√		
Routing Read/Write Accessible Parameters						
ISDN: Called Name (subscriber name)	N/A	✓	1	✓		
ISDN: Called Number Plan	N/A	✓	✓	✓		
ISDN: Called Party Number IE	N/A	✓	✓	✓		
ISDN: Called Type of Number	N/A	✓	✓	1		
ISDN: Calling Name (subscriber name)	N/A	✓	✓	✓		
ISDN: Calling Number Plan	N/A	✓	✓	✓		
ISDN: Calling Party Number IE	N/A	✓	✓	√		
ISDN: Calling Redirect Reason	N/A	✓	✓	√		
ISDN: Calling Type of Number	N/A	✓	✓	√		
ISDN: Display IE	N/A	✓	✓	√		
ISDN: ITC (Bearer Capability)	N/A	✓	✓	N/A		

Resulting fleast/Mirite SmartLink SmartVarie Trinity SmartIndidia Accessible Parameters ISDNC Prigning in Information (FISSS only) N/A N/A N/A N/A V V V V V V V V V	SmartNode™ Software Features (continued)						
SSDR, Origin line information (SESS) only) N/A N	Routing Read/Write	SmartLink	SmartWare	Trinity			
(SESS Only) INA INA INA INA INA INA INA IN	Accessible Parameters	(SmartLink 4022 & M-ATA)	(SmartNode 4000/5000 Series)	(SmartNode 4000/5000 Series)	(SmartNode 10000 Series)		
Confignal call number N/A		N/A	N/A	N/A	1		
If (redirecting)		N/A	✓	✓	✓		
Other: Day of week N/A J J Other: Time N/A J J Outgoing Trunk GroupPNAP N/A N/A N/A J R2CAS: Calle Party Category N/A J Roadmap J R2CAS: Called Number N/A J Roadmap J R2CAS: Calling Number N/A J Roadmap J SIP: Sand SRTP N/A N/A N/A N/A SIP: Alled Green N/A N/A N/A N/A SIP: Called (IP) N/A N/A N/A N/A SIP: Calling (IP) N/A N/A N/A N/A A J J J J J J J		N/A	✓	✓	✓		
Ottoging Trunk Group/NAP N/A N/A N/A N/A N/A N/A N/A V Color of the property o	Other: Date	N/A	✓	✓	✓		
Outgoing Trunk Group/NAP N/A N/A N/A V R2CAS: Call Party Category N/A J Roadmap J R2CAS: Called Number N/A J Roadmap J R2CAS: Called Number N/A J Roadmap J R2CAS: Called Number N/A J N/A N/A SIP TLS and SRTP N/A N/A N/A J SIP: Called (IP) N/A N/A N/A J SIP: Called (IP) N/A N/A N/A N/A J SIP: Called (User-info) N/A N/A N/A N/A J	Other: Day of week	N/A	✓	✓	✓		
R2CAS: Called Number N/A J Roadmap J R2CAS: Called Number N/A J Roadmap J R2CAS: Calling Number N/A J Roadmap J SIP: TLS and SRTP N/A N/A N/A N/A SIP: Any Headers N/A N/A N/A N/A SIP: Called (IP) N/A N/A N/A N/A SIP: Called (IP) N/A N/A N/A N/A SIP: Called (IP) N/A N/A N/A N/A SIP: Called (user-info) N/A N/A N/A N/A SIP: Called (user-info) N/A N/A N/A N/A J SIP: Calling (IP) N/A N/A N/A N/A J	Other: Time	N/A	✓	✓	✓		
R2CAS: Called Number	Outgoing Trunk Group/NAP	N/A	N/A	N/A	✓		
R2CAS: Calling Number N/A	R2CAS: Call Party Category	N/A	✓	Roadmap	✓		
SIP TLS and SRTP	R2CAS: Called Number	N/A	✓	Roadmap	✓		
SIP: Any Headers	R2CAS: Calling Number	N/A	✓	Roadmap	✓		
SIP: Called (IP)	SIP TLS and SRTP	N/A	N/A	✓	N/A		
SIP: Called (m) N/A N/A N/A V/A SIP: Called (user-info) N/A N/A N/A V V V SIP: Calling (spc) N/A N/A V V V V V V SIP: Calling (spc) N/A V	SIP: Any Headers	N/A	N/A	N/A	(insertion only for		
SIP: Called (user-info)	SIP: Called (IP)	N/A	✓	✓	✓		
SIP: Called Name (subscriber name) N/A J J SIP: Calling (cpc) N/A N/A N/A J SIP: Calling (IP) N/A J J J SIP: Calling (Ing (Ing name-addr) N/A J J J SIP: Calling (user-info) N/A N/A J J J SIP: Calling (user-info) N/A N/A N/A J J J SIP: Calling Redirect Reason N/A N/A N/A N/A J <t< td=""><td>SIP: Called (rn)</td><td>N/A</td><td>N/A</td><td>N/A</td><td>✓</td></t<>	SIP: Called (rn)	N/A	N/A	N/A	✓		
SIP: Calling (cpc) N/A N/A N/A V SIP: Calling (IP) N/A J J J SIP: Calling (name-addr) N/A J J J SIP: Calling (oli) N/A J J J SIP: Calling (user-info) N/A N/A N/A J SIP: Calling Redirect Reason N/A N/A N/A J SIP: Diversion (original call number) N/A N/A N/A J SIP: Diversion (redirecting) N/A J J N/A J SIP: Diversion (redirecting) N/A J J N/A J N/A J N/A J J N/A J J N/A J N/A J J N/A J N/A N/A N/A J J N/A N/A J J J J J J J J J J J J J J J J<	SIP: Called (user-info)	N/A	N/A	N/A	✓		
SIP: Calling (IP)	SIP: Called Name (subscriber name)	N/A	✓	✓	✓		
SIP: Calling (name-addr)	SIP: Calling (cpc)	N/A	N/A	N/A	✓		
SIP: Calling (oli)	SIP: Calling (IP)	N/A	✓	✓	✓		
SIP: Calling (user-info) N/A N/A V N/A N/A V V V V V N/A <	SIP: Calling (name-addr)	N/A	✓	✓	✓		
SIP: Calling Redirect Reason N/A J J SIP: Diversion (original call number) N/A N/A N/A J SIP: Diversion (redirecting) N/A J J J SIP: Diversion (redirecting) N/A J J J SIP: ITC (Bearer Capability) N/A J J N/A SIP: P-asserted-identity (name-addr) N/A J J J SIP: p-asserted-identity (user-info) N/A J J J SIP: Redirecting Number N/A J J J SIP: Redirecting Number IE (redirecting) N/A J J J SIP: Request URI N/A J J J J SIP: Request URI (rn) N/A J J J J J J SS7: Called Party Number IE N/A N/A N/A N/A J J J J J J J J J J J J J <t< td=""><td>SIP: Calling (oli)</td><td>N/A</td><td>✓</td><td>✓</td><td>✓</td></t<>	SIP: Calling (oli)	N/A	✓	✓	✓		
SIP: Diversion (original call number) N/A N/A N/A ✓ <td>SIP: Calling (user-info)</td> <td>N/A</td> <td>N/A</td> <td>N/A</td> <td>✓</td>	SIP: Calling (user-info)	N/A	N/A	N/A	✓		
SIP: Diversion (redirecting) N/A ✓ ✓ ✓ ✓ N/A ✓ N/A ✓	SIP: Calling Redirect Reason	N/A	✓	✓	✓		
SIP: ITC (Bearer Capability) N/A ✓ N/A SIP: p-asserted-identity (name-addr) N/A ✓ ✓ SIP: p-asserted-identity (user-info) N/A ✓ ✓ SIP: Redirecting Number IE (redirecting) N/A ✓ ✓ SIP: Request URI N/A ✓ ✓ SIP: Request URI (rn) N/A ✓ ✓ SS7: Called Party Number IE N/A N/A N/A SS7: Calling Party Number IE N/A N/A N/A SS7: Charge Number IE (ANSI) N/A N/A N/A SS7: Display Information IE N/A N/A N/A SS7: Generic Address Parameter IE (ANSI) N/A N/A N/A SS7: Origin Line Information IE (ANSI) N/A N/A N/A SS7: Redirecting Number IE N/A N/A N/A N/A	SIP: Diversion (original call number)	N/A	N/A	N/A	√		
SIP: p-asserted-identity (name-addr) SIP: p-asserted-identity (user-info) N/A SIP: Redirecting Number IE (redirecting) N/A SIP: Request URI N/A SIP: Request URI (rn) N/A SIP: Request URI (rn) N/A SIP: Request URI (rn) N/A N/A N/A N/A N/A N/A SS7: Called Party Number IE N/A N/A N/A N/A N/A N/A N/A N/	SIP: Diversion (redirecting)	N/A	✓	✓	√		
SIP: p-asserted-identity (user-info) N/A J J SIP: Redirecting Number IE (redirecting) N/A J J SIP: Request URI N/A J J SIP: Request URI (rn) N/A J J SIP: Request URI (rn) N/A J J SS7: Called Party Number IE N/A N/A N/A SS7: Calling Party Number IE N/A N/A N/A SS7: Charge Number IE (ANSI) N/A N/A N/A SS7: Display Information IE N/A N/A N/A SS7: Generic Address Parameter IE (ANSI) N/A N/A N/A SS7: Origin Line Information IE (ANSI) N/A N/A N/A SS7: Redirecting Number IE N/A N/A N/A	SIP: ITC (Bearer Capability)	N/A	✓	✓	N/A		
SIP: Redirecting Number IE (redirecting) SIP: Request URI N/A SIP: Request URI (rn) N/A SIP: Request URI (rn) N/A SS7: Called Party Number IE N/A N/A N/A N/A N/A N/A N/A N/	SIP: p-asserted-identity (name-addr)	N/A	✓	✓	√		
IE (redirecting) SIP: Request URI N/A SIP: Request URI (rn) N/A SS7: Called Party Number IE N/A N/A N/A N/A N/A N/A N/A N/	SIP: p-asserted-identity (user-info)	N/A	✓	✓	√		
SIP: Request URI (rn) SS7: Called Party Number IE N/A N/A N/A N/A N/A N/A N/A N/		N/A	✓	✓	✓		
SS7: Called Party Number IE SS7: Calling Party Number IE N/A N/A N/A N/A N/A N/A N/A SS7: Charge Number IE (ANSI) N/A N/A N/A N/A N/A N/A N/A N/	SIP: Request URI	N/A	✓	✓	✓		
SS7: Called Party Number IE SS7: Calling Party Number IE N/A N/A N/A N/A N/A N/A SS7: Charge Number IE (ANSI) N/A N/A N/A N/A N/A N/A N/A N/	SIP: Request URI (rn)	N/A	✓	✓	√		
SS7: Calling Party Number IE N/A N/A N/A N/A N/A N/A N/A N/	SS7: Called Party Number IE	N/A		N/A	√		
SS7: Display Information IE SS7: Generic Address Parameter IE (ANSI) SS7: Origin Line Information IE N/A N/A N/A N/A N/A N/A N/A N/	-	N/A	N/A	N/A	√		
SS7: Display Information IE SS7: Generic Address Parameter IE (ANSI) SS7: Origin Line Information IE N/A N/A N/A N/A N/A N/A N/A SS7: Redirecting Number IE N/A	o s						
SS7: Generic Address Parameter IE (ANSI) SS7: Origin Line Information IE (ANSI) N/A N/A N/A N/A N/A N/A N/A N/							
SS7: Origin Line Information IE (ANSI) SS7: Redirecting Number IE N/A N/A N/A N/A N/A N/A N/A N/	SS7: Generic Address Parameter						
SS7: Redirecting Number IE N/A N/A N/A N/A	SS7: Origin Line Information	N/A	N/A	N/A	✓		
SS7: Redirection Information IE N/A N/A N/A		N/A	N/A	N/A	✓		
	SS7: Redirection Information IE	N/A	N/A	N/A	✓		



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