Changing the Way the World Communicates

# IPNx Signalling & Media Gateway

# Technical Data

## Version 1.12

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# Introduction

The IPNx switch provides a range of integrated communications services including TDM voice, VoIP, Calling Card, voice VPNs, call routing and billing. The importance of the IPNx as a media & signalling gateway between C7 / SS7 networks and the IP world is growing.

Solutions based on ISDN E1 signalling work for the simple termination of voice traffic but present limitations in terms of the PSTN supplementary services that can be passed from the customer network to the carriers' networks. Even with wholesale agreements in place the rates offered to service providers connecting to carriers using ISDN PRI's are generally higher than those rates offered when the service provider can connect using C7 / SS7. Also when connecting to incumbent operators (such as B.T. in the UK) C7 / SS7 becomes a necessity to achieve any kind of wholesale rates or to provide services such as indirect access / carrier selection.

# C7 / SS7

### Protocols supported

Which variants of C7 / SS7 and which versions of these variants does the proposed product support? Please state and describe any functional limitations associated with their implementation of each of the supported variants.
MTP layer
The IPNX is conformant to ITU Q704 specification.
ISUP layer
The IPNX is conformant to ITU-T Q763 and Q764 1992 version. The IPNx supports ETSI ISUP as the core signalling and different country or carrier variants are controlled by parameters on the core.

No functional limitations are known.

### Distributed signalling

Please state if it is possible for the system to be split into two physically separate nodes which will, when connected via an IP network, act as a single redundant system. If this is possible state if it is possible for this geographically split system to have node A handle one signalling link from an interconnect which uses two signalling links and for node B to handle the second signalling link. If the proposed solution is capable of such functionality vendors should describe any problems or difficulties, if any, which their products have when installed and configured in such a way.
It is possible to configure the IPNx to run in a redundant manner as described here.
IPNx A will handle one of the signalling links from a carrier and IPNx B will handle the other. These 2 nodes may be geographically separate. They must be connected by a reliable IP network. The A and B nodes will operate in a load sharing fashion rather than having one node operational and the other in standby mode.

### Number of point codes supported

Please state the number of national and (if applicable to the variant) international point codes supported for each C7 / SS7 variant. Please also state and describe any factors, if any, which will reduce the number of point codes supported. I.e. the number of point codes supported is reduced for all protocols if a specific C7 / SS7 variant is enabled, etc.
The maximum number of destination point codes supported is 128. The maximum number of local point codes supported is currently 2. This could be increased to 8 if required. The maximum number of point codes is unaffected by factors such as choice of signalling variant.

### Multiple signalling protocols

Please state if the proposed product is able to operate using different C7 / SS7 variants at the same time. Also state if there is a limit on the number of C7 / SS7 variants that can be utilised at

the same time or if it is possible for each signalling link or at least each pair of signalling links to be configured with a different C7 / SS7 variant at the same time.
There is no limit on the different SS7 variants which can be used at the same time.

### Scalability

Please provide clear answers to the following:

Busy Hour Call Attempts (BHCA) supported by a redundant system.

BHCA supported is 300,000

Maximum number of signalling links a) per single switch and b) per single redundant system.

a. 56 signalling links is the maximum number per single chassis

b. 112 signalling links is the maximum number per single redundant system

Maximum number of signalling links per interconnect.

The maximum number of signalling links per interconnect is 16 (as in SS7 standards).

Maximum number of M.S.U.'s supported by a single redundant system.

A single redundant system can support up to 8000 MSUs per second.

Maximum number of sustained calls for a single redundant system.

Maximum number of sustained calls per single chassis is 1,680. This means all 56 E1s are fully active.

Maximum number of sustained calls is 3,360. This means all 112 E1s are fully active.

Maximum number of E1's supported by a single redundant system.

112 E1's supported by a single redundant system.

Maximum number of STM-1s supported by a single redundant system.

1 STM-1 supported by a single redundant system.

Note: All of the answers here assume that a single redundant system is made up of a redundant pair of IPNx switches, each capable of supporting up to 56 E1 ports (therefore, 112 for the redundant system). However, these redundant pairs can in turn be combined to create switches up to 2048 E1s. Since this means adding new switches with their own processing then all of the above values increase in a linear fashion.

### Bearer capabilities

Please list the bearer capabilities that the C7 / SS7 solution supports. Also list what bearer capability can be mapped from C7 / SS7 to the other protocols supported by the solution. For example can a call with a 64K UDI bearer capability / Telephony Media Request (TMR) be mapped to an H.323 call using the proposed solution. Also list what bearer capabilities can be mapped from other protocols to the C7 / SS7 side using the proposed solution.

Supported Capabilities

| Function/service | IPNX |
|---|---|
| Basic call | |
| Speech | Y |
| 3.1 kHz audio | Y |
| 64 kbit/s unrestricted | Y |
| | |

The above bearer capabilities can be mapped to all other signalling protocols. It is also possible to override requested capabilities (for example these are often wrongly formatted by H323 equipment).

## Licensing

Please clearly state if a license fee, in addition to any hardware or software costs, is associated with any part of the C7 / SS7 solution. For example does a signalling link card require a license to be purchased before that card can be used (either used practically or legally).

Every E1 port that carries a SS7 signalling link must be licensed. The list price of this license is €2400

## C7 / SS7 approvals

Clearly state the C7 / SS7 approvals that the product has achieved. This explanation should clearly show the organisation granting the approval, the country that the approval applies to, the date of the approval and the C7 / SS7 variant(s) and version(s) that that approval applies to. It should also be clearly stated if the approvals granted apply to all current and future software levels on the C7 / SS7 solution or if any of the approvals will cease as software levels increase. If any of the approvals expire after a specific date this date(s) should be clearly shown.

| **Organisation** | **Country** | **Date** | **Software Version** |
|---|---|---|---|
| BT | UK | 2002 | TXISUP V1.36 |
| Belgacom | Belgium | 1999 | TXISUP V1.24 |
| Cable & Wireless | UK | 2000 | TXISUP V1.30 |
| CYTA | Cyprus | 2007 | TXISUP V1.65 |
| GT | Ghana | 2007 | TXISUP V1.65 |
| KPN | Holland | 2001 | TXISUP V1.33 |
| Mauritius Telecom Group | Mauritius | 2007 | TXISUP V1.65 |
| Nitel | Nigeria | 2004 | TXISUP V1.50 |
| NTC | Pakistan | 2002 | TXISUP V1.36 |
| TeleDenmark | Denmark | 2000 | TXISUP V1.29 |
| Telefonica | Spain | 2000 | TXISUP V1.30 |
| Telenor | Norway | 2001 | TXISUP V1.33 |
| PTT Austria | Austria | 1999 | TXISUP V1.26 |

### C7 / SS7 interconnect

Clearly state the number of C7 / SS7 interconnects and with which variant and version this interconnect took place that their product has passed. Vendors should also state the following for each interconnect:

a) Country that interconnect took place.

b) Carrier interconnected too.

c) Number of signalling links used.

d) Number of E1 bearers used.

e) Traffic loading in terms of calls per second (CPS) and Erlangs or minutes per month after the interconnect had been completed and had gone live.

f) If traffic was incoming, outgoing or both.

g) Main use of interconnect I.e. terminating traffic, indirect access, etc.

| | **Country** | **Carrier** | **Signalling Links** | **E1 Bearers** | **Traffic (In, Out, Both)** | **Main Use** |
|---|---|---|---|---|---|---|
| 1. | Algeria | Algeria Telecom | 2 | 8 | Both | VoIP Trunking |
| 2. | Algeria | Djezzy | 2 | 8 | Both | VoIP Trunking |
| 3. | Algeria | Mobilis | 2 | 8 | Both | VoIP Trunking |
| 4. | Austria | Mobilkom | 2 | 4 | Both | Call Back |
| 5. | Austria | TeleKom Austria | 2 | 300 | Both | Carrier Select |
| 6. | Bahrain | Batelco | 2 | 4 | Both | VoIP Trunking |
| 7. | Belgium | B3G | 2 | 10 | Both | Least Cost Routing |
| 8. | Belgium | Belgacom | 2 | 20 | Both | Least Cost Routing |
| 9. | Belgium | BT Ignite | 2 | 8 | Both | Least Cost Routing |
| 10. | Belgium | Colt | 2 | 30 | Both | Least Cost Routing |
| 11. | Belgium | Deutsche Telekom | 2 | 20 | Both | Least Cost Routing |
| 12. | Belgium | EADS | 2 | 10 | Both | Least Cost Routing |
| 13. | Belgium | Enertel | 2 | 20 | Both | Least Cost Routing |
| 14. | Belgium | France Telecom | 2 | 60 | Both | Least Cost Routing |
| 15. | Belgium | Global UK | 2 | 24 | Both | Least Cost Routing |
| 16. | Belgium | IDT | 2 | 30 | Both | Least Cost Routing |

| | | | | | | |
|---|---|---|---|---|---|---|
| 17. | Belgium | InterRoute | 2 | 50 | Both | Least Cost Routing |
| 18. | Belgium | Kedra | 2 | 30 | Both | Least Cost Routing |
| 19. | Belgium | KPN | 2 | 40 | Both | Least Cost Routing |
| 20. | Belgium | LD Com | 2 | 16 | Both | Least Cost Routing |
| 21. | Belgium | Primus | 2 | 24 | Both | Least Cost Routing |
| 22. | Belgium | RSL Com | 2 | 30 | Both | Least Cost Routing |
| 23. | Belgium | Swisscom | 2 | 24 | Both | Least Cost Routing |
| 24. | Belgium | Telia | 2 | 20 | Both | Least Cost Routing |
| 25. | Belgium | TTG | 2 | 4 | Both | Least Cost Routing |
| 26. | Belgium | Versatel | 2 | 80 | Both | Least Cost Routing |
| 27. | Belgium | Worldcom | 2 | 60 | Both | Least Cost Routing |
| 28. | Cyprus | CYTA | 2 | 2 | Both | VoIP Trunking |
| 29. | Denmark | TeleDenmark | 2 | 2 | Both | Least Cost Routing |
| 30. | Denmark | Netcom | 2 | 2 | Both | Least Cost Routing |
| 31. | France | Colt | 2 | 4 | Out | Least Cost Routing |
| 32. | France | Intercall | 2 | 2 | Out | Least Cost Routing |
| 33. | France | Verizon / MCI | 2 | 8 | Both | Least Cost Routing |
| 34. | France | WaveCrest | 2 | 8 | Both | Least Cost Routing |
| 35. | Germany | Colt | 2 | 4 | Both | Least Cost Routing |
| 36. | Germany | Energis | 2 | 4 | Both | Least Cost Routing |
| 37. | Germany | Eutex | 2 | 4 | Both | Least Cost Routing |
| 38. | Germany | Global Crossing | 2 | 4 | Both | Least Cost Routing |
| 39. | Germany | Materna | 2 | 4 | Both | Least Cost Routing |
| 40. | Germany | MCI | 2 | 4 | Both | Least Cost Routing |
| 41. | Germany | Tropolys (Versatel) | 2 | 4 | Both | Least Cost Routing |
| 42. | Germany | T-Systems | 2 | 4 | Both | Least Cost Routing |
| 43. | Germany | Telia | 2 | 4 | Both | Least Cost Routing |

| | | | | | | |
|---|---|---|---|---|---|---|
| 44. | Holland | AT&T | 2 | 2 | Out | Least Cost Routing |
| 45. | Holland | BBNED | 2 | 8 | Both | Least Cost Routing |
| 46. | Holland | BT | 2 | 2 | Out | Least Cost Routing |
| 47. | Holland | Carrier1 | 2 | 2 | Out | Least Cost Routing |
| 48. | Holland | Colt | 2 | 8 | Both | Least Cost Routing |
| 49. | Holland | KPN | 2 | 8 | Both | Least Cost Routing |
| 50. | Holland | Telia | 2 | 4 | Both | Least Cost Routing |
| 51. | Holland | TTG | 2 | 4 | Both | Least Cost Routing |
| 52. | Holland | Verizon / MCI | 2 | 8 | Both | Least Cost Routing |
| 53. | Iceland | Iceland PTT | 2 | 4 | Both | GSM Callback |
| 54. | Iceland | Vodafone | 2 | 4 | Both | GSM Callback |
| 55. | Mauritius | Cellplus | 2 | 8 | Both | VoIP Trunking |
| 56. | Mauritius | Emtel | 2 | 8 | Both | VoIP Trunking |
| 57. | Mauritius | Mauritius Telecom | 2 | 8 | Both | VoIP Trunking |
| 58. | Morocco | Maroc Telecom | 4 | 16 | Both | Connect SIP server |
| 59. | Nigeria | ITN | 2 | 4 | Both | Least Cost Routing |
| 60. | Spain | Telefonica | 2 | 60 | Both | VoIP Trunking |
| 61. | Switzerland | Global Crossing | 2 | 4 | Both | Least Cost Routing |
| 62. | Switzerland | Swisscom | 2 | 4 | Both | Least Cost Routing |
| 63. | Sweden | Telia | 2 | 4 | Both | Least Cost Routing |
| 64. | UK | Arbinet | 2 | 4 | Both | Least Cost Routing |
| 65. | UK | BT | 2 | 4 | Both | Least Cost Routing |
| 66. | UK | Cable & Wireless | 2 | 4 | Both | VoIP Trunking |
| 67. | UK | Energis | 2 | 2 | Both | Least Cost Routing |
| 68. | UK | Global | 2 | 2 | Both | Least Cost |

| | | | | | | |
|---|---|---|---|---|---|---|
| | | Crossing | | | | Routing |
| 69. | UK | Kingston | 2 | 2 | Both | VoIP Trunking |
| 70. | UK | MCI | 2 | 4 | Both | Least Cost Routing |
| 71. | UK | Swisscom | 2 | 2 | Both | Least Cost Routing |
| 72. | UK | Telecom New Zealand | 2 | 2 | Both | Least Cost Routing |
| 73. | UK | KPN | 2 | 4 | Both | Least Cost Routing |
| 74. | UK | Reach Telecom | 2 | 4 | Both | Least Cost Routing |
| 75. | UK | T-Liaison | 2 | 4 | Both | Least Cost Routing |

Other SS7 Equipment Interconnects

| **Country** | **Manufacturer** | **Year** |
|---|---|---|
| Switzerland | Teles | 2002 |
| Holland | Clarent | 2004 |
| | | |

### DDI numbers

Clearly state if the proposed C7 / SS7 solution has the ability to route incoming DDI numbers to the correct CPE customer. State any limitations with regards to this functionality. I.e. depends on C7 / SS7 variant, depends if routing to a VoIP protocol or ISDN PRI port on the C7 / SS7 gateway, can route to H.323 devices but not to SIP devices, etc.

The IPNx can route incoming DDI numbers based on part or all of the DDI number. The called number may also be modified if required. Calls may be routed to ISDN PRI ports or to individual IP addresses. No limitations exist on the use of this function. There is no distinction made between H323 and SIP.

### Indirect access

Clearly state if the proposed solution supports an indirect access (IDA) service. Describe the basic operation of this service. For example do all IDA calls (excluding those actually destined for a directly connected customer) pass through the C7 / SS7 TDM solution only or is the call routed from the C7 / SS7 solution to the IP network and back to the C7 / SS7 side.

Also describe any additional software and / or hardware that is required to enable IDA on the product. Any licensing requirements should also be clearly stated and explained.

State if the IDA implementation allows full C7 / SS7 signalling transparency (so far as any differences between the incoming C7 / SS7 variant and the outgoing C7 / SS7 variant allow).

IDA is supported as a standard feature of the IPNx. No additional hardware or licensing is required to enable it. This may be by 2 stage dialling or by carrier select codes. Number translation features are available which give great flexibility in the case of carrier select. Carrier select codes may be stripped or inserted as required.

The IPNx is a fully functional telecom switch. Calls are routed in the most efficient way regardless of traffic type. Therefore, calls will be routed directly from one SS7 carrier to another with complete signalling transparency.

#### Carrier pre-selection

Clearly state if the proposed solution supports a Carrier Pre-Select (CPS) service (the ability for the platform to receive and correctly handle CPS calls from subscribers (passed via an intermediate network) using CPS). Describe the basic operation of this service on the product. For example, do all CPS calls (excluding those actually destined for a directly connected customer) pass through the C7 / SS7 TDM solution only or is the call routed from the C7 / SS7 solution to the IP network and back to the C7 / SS7 side.

Also describe any additional software and / or hardware that is required to enable CPS on their product. Any licensing requirements should also be clearly stated and explained.

State if the CPS implementation allows full C7 / SS7 signalling transparency (so far as any differences between the incoming C7 / SS7 variant and the outgoing C7 / SS7 variant allow).

CPS is supported as a standard feature of the IPNx. No additional hardware or licensing is required to enable it.

The IPNx is a fully functional telecom switch. Calls are routed in the most efficient way regardless of traffic type. Therefore, CPS calls will be routed directly from one SS7 carrier to another with complete signalling transparency.

# VoIP

#### Protocols supported

Which VoIP protocols and which versions of these protocols does the proposed product support (signalling transport mechanisms / protocols such as SIP-T & Sigtran / SCTP should be included here)? Vendors should state and describe any functional limitations associated with their implementation of each of the supported variants.

The IPNx supports SIP and H323 Versions 1, 2, 3 and 4.

#### H323 Support

The following components of H323 are supported:

H225 version 4 including the following features:

- multiple calls per call signalling channel
- separate H245 connection
- fast start
- early H245 connection
- tunneling of H245 and fallback to separate connection procedure if the peer does not support tunneling
- enbloc/overlap sending/receiving
- SS7/Q931 mapping of the following information elements:
    - called party number
    - calling party number
    - progress indicator
    - bearer capability
    - user to user information
    - called subaddress
    - cause value
    - notification indicator
    - plus all parameters passed in the Access Transport Parameter.
- support of the following optional Q931 parameters:

  - date/time
  - call state
  - keypad for DTMF sending/receive
- support all the mandatory and optional Q931 messages:
  - Setup
  - Setup Acknowledge
  - Information
  - Call Proceeding
  - Progress
  - Alerting
  - Connect
  - Status
  - Status inquiry (inbound only)
  - Notify
  - Facility
  - ReleaseComplete
- support of all the mandatory H323_UserInformation parameters and the following optional parameters:
  - destinationAddress
  - sourceAddress
  - mediaWaitForConnect
  - releaseComplete.reason
- support of all RTP/RTCP audio procedures:

G723.1, G726, G728, G729, G729a, G729b, G729ab, G711 Alaw/uLaw, GSM

  - RTCP generation
  - silence detection
  - silence packet generation/suppression
  - dynamic payload type
  - programmable concatenation of voice packets
  - T38 fax relay
  - G711 fax bypass

H245 version 6

- support of the following procedures:
  - terminalCapabilitySet
  - sendTerminalCapabilitySet
  - masterSlaveDetermination
  - openLogicalChannel (unidirectional and bidirectional)
  - closeLogicalChannel
  - requestMode
  - requestChannelClose
  - roundTripDelay
  - flowControlCommand
  - userInput (conversion between inband DTMF and userInput)
  - endSession
- following messages automatically rejected:
  - multiplexEntrySend
  - requestMultiplexEntry
  - maintenanceLoopRequest
  - logicalChannelRateRequest

All other messages replied with functionNotSupported

## SIP Support

| Feature | Supported Y/N? |
|---|---|
| Layer 4 protocols and are they mandatory or optional? (UDP, TCP, TLS, SCTP)? | UDP only |
| UDP fragmentation for both send and receive | Yes. |
| Layer 4 protocols planned? | TCP planned for Q4 07 |
| Uri's types supported? (SIP Uri, Tel Uri without phone context, Tel Uri with phone context) | SIP Uri (assuming that user name is a phone number). Tel Uri planned to be added later. |
| Maximum length of SIP-URI? | 46 characters |
| Maximum length of User Name | 21 characters |
| Maximum length of Password | 8 characters |
| RFC 3261 SIP signalling? | Yes. |
| Are 7, 10, or 11 digit numbers sent as dialled (i.e. not converted to e.164)? | Yes. |
| Does WTL SIP service require support of P-RACK | No (will be added as part of SIP-T support). |
| Voice codecs are supported? | G723.1, GSM-FR, G726-32, G729, G711 A-Law, G711 u-Law |
| Can Silence Suppression be disabled? | Yes. |
| Is Diffserve supported? | Yes. (Note: DiffServe only supported on RTP). |
| Does network provide ring back or does it require this to be locally provided? | Can generate ring back (pass transparently from the SS7 network or generate locally within the SoIP gateway) |
| Do you require early media capability (180/183 with SDP) in the CPE SIP endpoints? | We do not require early media capability but is preferred. |
| Yype of Authentication used i.e. digest, with or without "qop=auth-int", etc. | CLI, IP, prefix or Digest without qop |
| What services/features are provided by the network, that requires feature support on the CPE in order to interoperate e.g. Message waiting indication, hold, transfer… | Only basic SIP features are required |
| Can a particular contact be removed through unregistration leaving all other registrations intact? | Unregister is supported |
| Is RFC 3323 Privacy header supported? | Yes. User/CLI can be withheld or spoofed "Anonymous" requests. |
| Equipment operates behind a NAT? If so, do you provide SBC/ALG functionality for endpoints to traverse through NAT? | Yes, operation behind NAT is supported, can redirect RTP stream based on 'real' IP address. However, we recommend use of a STUN server. SBC functionality provided: we can have multiple IP connections (one private, one public) and switch RTP streams between. |
| How many numbers can you allocate per trunks? Can you describe the trunk account that we will be given? Is there for each number an equivalent PSTN number that we can dial to perform loop-back testing? | No limit. Up to 60 types of trunks are possible with different characteristics (local address binding, authentication etc). The loop-back question is a function of the number translation in the routing plan. |
| Do you use refresh on an existing call (RFC 4028, support for session timers)? Do they support UPDATE (RFC 3311)? | Session timer supported but not according to RFC. UPDATE supported for purpose of Session Timer. |

| | |
|---|---|
| Fax support? If so, can Fax be offered at the beginning of the session? Or does fax transmission occur only in the middle of a call? | Fax supported by fallback to G711. Fax transmission occurs only in the middle of a call. |
| Invite with no SDP? | Yes. |
| ReInvite? | Yes. |
| G729 Annex B or Annex A or both? Do you send and accept attribute for Annex B or Annex A in SDP? | Annex A & B supported for both send & accept. |
| RFC 2833 for DTMF tones? | Yes. DTMF can be sent inband with G.711 or as named events with G.711 and G729a |
| Can receive full and compact SIP headers (ie. "From:" & "f:") | Yes |
| RFC 2543 call hold (ie. a=sendonly or c=0.0.0.0) | Yes |
| Can receive RFC 3262 call hold (ie. a=sendonly) | Yes |
| RFC 3325 P-Asserted-Id header | No |
| SIP Early Offer/Answer w/SDP (i.e. Request-URI w/SDP and 183 response w/SDP) | Yes |
| SIP Diversion Header (draft-levy-sip-diversion-08 ) | No |
| Support/complies with mid-session codec changes using SIP Re-Invites | Yes |
| Supports SIP Re-Invites for Transfers | No |
| Sends INVITEs with SIP URLs (not tel URLs) | No |
| | |
| | |

SIP RFCs 2327, 2543, 2617, 2833, 3261 supported.
SIP methods currently supported:
- INVITE
- REINVITE with SDP redirect
- SDP in provisional responses for early media
- REFER
- REGISTER (client side)
- proxy (client side)
- RPORT (response)
- UPDATE (for keepalive, no SDP redirect)
- DTMF over INFO (Cisco method)

SIP methods currently under development. Please contact WTL for current status.

- REGISTER (server side)
- fax over T38 (media line in SDP)
- SIP-T (SIP to SS7 mapping)
- UPDATE (with SDP redirect)

SIP methods that are not planned at present but may be supported later:
- reliable provisionnal responses (PRACK)

### SIP RFC Support

The equipment is compliant with the following IETF standards:

| RFC Number | Description | Supported: (Yes, No, Partial) | Notes |
|---|---|---|---|
| 2327 | Session Description Protocol | yes | |
| 2543 | User Agent | yes | |
| 2617 | HTTP Authentication | yes | |
| 2806 | URLs for Telephone Calls | partial | only tel: scheme |
| 2833 | for DTMF transmission | yes | |
| 2976 | The SIP INFO Method | yes | |
| 3204 **(updated by RFC 3459)** | MIME media types for ISUP and QSIG Objects | no | |
| 3261 **(obsoletes RFC 2543/ updated by RFC 3853,RFC 4320)** | Session Initiation Protocol | yes | |
| 3262 **(obsoletes RFC 2543)** | Reliability of Provisional Responses in SIP | partial | we send PRACK, we don't request them |
| 3263 **(obsoletes RFC 2543)** | SIP: Locating SIP Servers | yes | |
| 3264 | An Offer/Answer Model with the Session Description Protocol (SDP) | yes | |
| 3265 **obsoletes RFC 2543)** | SIP-Specific Event Notification | partial | only in the context of Refer |
| 3310 | Hypertext Transfer Protocol (HTTP) Digest Authentication Using Authentication and Key Agreement (AKA) | no | |
| 3311 | The Session Initiation Protocol UPDATE Method | yes | |
| 3312 **(updated by RFC 4032)** | Integration of Resource Management and SIP | no | |
| 3313 | Private Session Initiation Protocol (SIP)Extensions for Media Authorization | no | |
| 3319 | Dynamic Host Configuration Protocol (DHCPv6)Options for Session Initiation Protocol (SIP) Servers | no | |
| 3323 | A Privacy Mechanism for the Session Initiation Protocol (SIP) | yes | |
| 3325 | Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks | yes | |

| | | | |
|---|---|---|---|
| 3326 | The Reason Header Field for the Session Initiation Protocol (SIP) | yes | |
| 3327 | Session Initiation Protocol Extension for Registering Non-Adjacent Contacts | no | |
| 3329 | Security Mechanism Agreement for the Session Initiation Protocol (SIP) Sessions | no | |
| 3361 | DHCP Option for SIP Servers | no | |
| 3389 | Real-time Transport Protocol (RTP) Payload for Comfort Noise | yes | |
| 3420 | Internet Media Type message/sipfrag | yes | |
| 3428 | Session Initiation Protocol Extension for Instant Messaging | no | |
| 3486 | Compressing the Session Initiation Protocol | no | |
| 3515 | The Session Initiation Protocol (SIP) Refer Method | yes | |
| 3581 | An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing | yes | |
| 3608 | Session Initiation Protocol Extension Header Field for Service Route Discovery During Registration | no | |
| 3666 | Session Initiation Protocol Extension Header Field for Service Route Discovery During Registration | yes | |
| 3840 | Indicating User Agent Capabilities in the Session Initiation Protocol (SIP) | no | |
| 3841 | Caller Preferences for the Session Initiation Protocol (SIP) | no | |
| 3853 | S/MIME Advanced Encryption Standard (AES) Requirement for the Session Initiation Protocol (SIP) | no | |
| 3891 | The Session Initiation Protocol (SIP) "Replaces" Header | no | |
| 3892 | The Session Initiation Protocol (SIP) Referred-By Mechanism | no | |
| 3893 | Session Initiation Protocol (SIP) Authenticated Identity Body (AIB) Format | no | |

| | | | |
|---|---|---|---|
| 3903 | Session Initiation Protocol (SIP) Extension for Event State Publication | no | |
| 3911 | The Session Initiation Protocol (SIP) "Join" Header | no | |
| 3959 | The Early Session Disposition Type for the Session Initiation Protocol (SIP) | yes | |
| 3960 | Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP) | yes | |
| 3968 | The Internet Assigned Number Authority (IANA) Header Field Parameter Registry for the Session Initiation Protocol (SIP) | yes | |
| 3969 | The Internet Assigned Number Authority (IANA) Universal Resource Identifier (URI) Parameter Registry for the Session Initiation Protocol (SIP) | yes | |
| 4028 | Session Timers in the Session Initiation Protocol (SIP) | yes | |
| 4040 | RTP Payload Format for a 64 kbit/s Transparent Call | yes | |

### Sigtran and SCTP

Signal transport is via WTL proprietary mechanism known as OSST. WTL developed OSST as a highly efficient method for SS7 signalling to be carried through IP networks. This protocol encodes the full range of SS7 messages in a secure, compact way which has been optimised for IP transport. OSST then uses smart routing algorithms to decide which device in the network to send the SS7 message to. SS7 call control can thus be extended to remote SS7 trunks anywhere in the network or converted to be compatible with VoIP protocols.

Sigtran and SCTP are not supported.

### Multiple signalling protocols

State if the proposed product is able to operate using different VoIP protocols at the same time.

The IPNx can operate with different VoIP protocols at the same time.

### Scalability

Provide clear answers to the following:

a) Busy Hour Call Attempts (BHCA) for VoIP calls to C7 / SS7 supported by a redundant system. If the VoIP protocol in use affects this figure then separate figures for each VoIP protocol should be provided.

The BHCA for VoIP calls to SS7 is 100,000 per single chassis.

The BHCA for VoIP calls to SS7 is 200,000 per single redundant system.

b) Maximum number of sustained VoIP to C7 / SS7 calls for a single redundant system.

Maximum number of sustained VoIP to C7 / SS7 calls for a single chassis is 960.

Maximum number of sustained VoIP to C7 / SS7 calls for a single redundant system is 1,920.

### Voice and video CODEC's supported

State which voice and video CODEC's are supported by the proposed solution.

G711, G723.1, G726 (ADPCM), G728, GSM and G729A voice codecs are supported. Also Netcoder, proprietary high quality, low bandwidth codec.

### Fax & Modem over IP support

State which fax over IP (FoIP) and modem over IP protocols the solution supports.

T.38 real time fax over IP is supported or fallback to G711. Modem traffic is supported by fallback to G711

### Licensing

Clearly state if a license fee, in addition to any hardware or software costs, is associated with any part of the VoIP side of the solution. For example does a VoIP IP interface card require a license to be purchased before that card can be used (either used practically or legally).

A license fee is chargeable per E1 for traffic which is translated into VoIP. List price for this one-off software license is €5,200.

### VoIP interconnect

VoIP interconnects between commercial carriers will become more and more commonplace in the near future. As such the ability to interconnect to VoIP carriers is a necessity.

Clearly describe the functionality that the solution contains to allow VoIP interconnects to take place and list the interconnects that have been achieved.

Customers have used the IPNx to connect to many different VoIP carriers. However, this is such a routine (and informal) part of a carrier's business that WTL are frequently not aware of the details. There is also the issue of our clients' confidentiality. The following is a list of some of the carriers to whom the IPNx has been successfully connected but without the detail requested above:

| **Carrier** | **Country** |
|---|---|
| Frontier | UK |
| Citrus | UK |
| Talk24 | UK |
| IBasis | Holland |
| Batelco | Bahrain |
| Sahranet | Turkey |
| GlobeTel | USA |
| LC Link | USA |
| Verscom | Turkey |

| | |
|---|---|
| Singtel | Singapore |
| Telecom Italia | Italy |
| Verizon/MCI | Belgium |

# SBC & Security features

### SBC Support

State the SBC (Session Border Controller) features supported and whether these are included in the product or subject to extra software licensing

The WTL SBC is included as standard in all products. It allows the network operator to control the kinds of calls that can be placed through the network on which it resides, to fix or change protocols and protocol syntax to achieve interoperability, and also to overcome some of the problems that firewalls and NAT (Network Address Translation) cause for VoIP calls.

Additionally, the WTL SBC can allow VoIP calls to be set up between two devices using different VoIP signaling protocols (SIP, H.323, NOP) as well as performing transcoding of the media stream when different codecs are in use*. The WTL SBC also provides certain firewall features for VoIP traffic (denial of service protection, call filtering, bandwidth management, etc...).

The WTL SBC function allows operators to offer multiple different business models in complete safety:

- Carrier-to-carrier
  - Bilateral or multilateral
  - Domestic and international
  - Interconnect and peering
- Carrier to enterprise
  - SIP trunking
  - Hosted IP PBX and IP Centrex
  - SIP Application services (conferencing, IVR etc)
  - Number manipulation
  - Correct 'Nature of Address' handling
  - Privacy/anonymous CLI
  - Call throttling
  - PABX environment (WTL acts as pseudo-local exchange)
- Carrier to Subscriber
  - VoBB (Voice over Broadband)
  - SIP registration
  - Call transfer
  - Voice mail support – redirect on no answer, on busy, unconditional, call forwarding
  - Pre-Paid services
  - Combined voice and data services

For all models a generic set of features provide protection for the operator's network:

- Overload protection (inbound, outbound or total call limits can be imposed)
- Call usage / Rate limiting
- Per session authentication and authorisation
- Call rating and mediation

**WTL SBC Feature List**

- Firewall Traversal
- NAT Traversal
- Topology hiding
- IP Address Resolution/Management
- SIP to H.323 conversion
- Voice codec conversion*
- Built-in firewall
- Authenticate VoIP calls and callers
- Session Admission Control
- Prevention of DoS attacks

- RTP termination and regeneration
- SSL tunnels
- H.323 V2 & 3 ( + partial V4 support)
- H.323 Fast Start & Slow Start
- H.323 ToS support
- Qos Marking, DiffServe support
- SIP transaction rate limiting (limit number of SIP Invites)
- Detect and drop malformed packets
- Per trunk bandwidth RTP policing
- CDR generation
- G.711 / T.38 Fax relay for SIP and H.323
- RADIUS Support
- Digit matching/manipulation
- Deep packet inspection
- H.245 tunnelling support
- H.225 RAS messages for alternative gatekeeper functionality
- Source and destination trunk group support
- Simultaneous peering with multiple gatekeepers and gateways

* Codec conversion requires hardware-based voice resources

### VoIP Security

State the VoIP security measures available and how these are implemented.

WTL equipment provides a wide range of features to keep the operator's network secure from threats to operational service (DoS attacks) or to revenues (fraud attacks). The WTL SBC adds the following features:

- **'Anti-spoofing' techniques** confirm that a source IP address is genuine. The WTL call control uses the source IP address not the IP address 'claimed' in the message.
- **Flood attack prevention:** SIP sessions may be limited based on source IP address. WTL implement the concept of a 'Max Users' field per account. An account can be tied to a source IP address.
- **NAT traversal** is supported by use of a callers' source IP address.
- **2 levels of IP filtering** are implemented to allow only known/trusted source IP addresses into the network. This filtering is flexible and works with a) specific approved addresses or b) ranges of addresses. In addition, IP address may be associated with an account. Accounts can in turn be validated by CLI or using Digest Authentication.
- **DOS attacks** are limited by using a first level of IP filtering which prevents an attack entering the higher levels of the application. Thus they are rejected with minimal resource utilisation.
- **CLI Spoofing:** a double level of verification is offered by WTL switches to ensure that an incoming CLI matches the IP address that it is supposed to come from.
- **Intelligent access lists**
- **IP address and port translation**: The WTL SBC provides access control and private-to-public IP network and port address translation.
- **Topology hiding** is supported. SIP message transiting WTL switches have all 'via lines' removed.
- **Traffic separation:** All equipment supports 2 Ethernet interfaces allowing the separation of administration and user traffic.
- **SIP signaling attacks** are dealt with using the 2 level IP filtering described above.
- **Blacklisting of endpoints:** the equipment contains tables of IP address / CLIs that are not allowed to call.
- **Malformed SIP messages** are efficiently dealt with and discarded.
- **Alarms** are recorded in the WTL equipment log files for audit or diagnostic purposes.

### Protocol Matching

State what support is included in the product for protocol matching.

WTL switches are built around a central core which uses a standard representation of a call or session – and all the attributes associated with it – regardless of the protocol or media used by

the call. This makes WTL switches ideal in a ‘transit’ environment, since they are able to support multiple simultaneous protocols and route calls between them with no limitations.
VoIP calls may be made seamlessly between SIP and H,323, and also to WTL’s own bandwidth optimisation protocol, NOP. Since SIP is an open and extensible protocol, many different variations exist which may, at times, be incompatible. WTL’s SBC function addresses this and ‘normalises’ the signalling variations so that calls may be successfully connected between different vendors’ equipment and amongst many operator networks.
The SoIP gateway is also capable of performing the same service for VoIP to TDM (SS7, ISDN and R2) calls.

### IP Addressing & Port Management

Describe any features available to assist with IP Addressing & Port Management.
The WTL SBC allows a range of operations using the IP address of inbound calls. These include caller authentication, address translation, and call routing.
To improve security, the following SIP-specific method is supported: if a call is received via a known SIP proxy the WTL SBC will detect that the calls come from the proxy’s IP address, and will look in the second via line for the actual customer IP address. Also, to counter CLI spoofing, an incoming CLI may be checked to ensure that the IP address is the one that ought to be associated with it.
In common with other gateways, WTL will allow the SIP port to be configurable. However, WTL can also support multiple SIP ports being used simultaneously. Different ports can have different settings: different proxies, forms of verification, choice of codecs etc because each SIP trunk has its own independent SIP signaling stack.
In order to overcome blocking of SIP by NAT devices, the WTL SBC can map SIP traffic to a non-standard IP port if required.
The WTL SBC supports IP address resolution for outbound calls according to DNS standards. This can be configured to be with or without external DNS, or to friendly DNS servers only. This means that the switch can route using a host name which can be useful for load sharing.

## Call routing

Describe the methods available for call routing.
At the core of all WTL products is a robust, high capacity, intelligent Call Routing Engine. This allows a very flexible system of dial plans based on dialed E.164 digits.
Numbering features include the following:

- Called number translation
- Calling number verification/manipulation
- Trunk capacity limits
- Hunt groups
- Flexible Dial plans
- Day & time based routing
- Release cause based routing
- Calling number translation
- Call load sharing
- Session (Call) Detail Reporting
- LCR
- Registration-based routing
- Traffic type based routing
- Destination grouping

Sophisticated Least Cost Routing (LCR) models may be created with multi-level fallback to up to 5 alternate carriers per destination dialling code (or code group).
The WTL routing engine also supports a ‘trusted’ mode to be set up for simple IP exchange applications. In this case, traffic comes from ‘trusted’ sources and authentication rules may be relaxed.
Session (Call) Detail Reporting includes protocol information, QoS metrics, release causes, actual route taken etc.

Clearly state on what field / parameter / information their solution / product can make routing decisions. The following table lists a number of these fields / parameters / information basis.

 

| Basis of routing decision | Supported by solution |
|---|---|
| Time of day | Yes |
| Day of week | Yes |
| Date of month | No |
| Special dates / holidays | Yes |
| Bearer capability / TMR | Yes |
| DDI number | Yes |
| A number / CLI | Yes |
| Based upon inbound route | Yes |
| Protocol used to present call | No |

State if the solution is able to make all routing decisions that it can support on all protocols that it supports.

The routing decisions are not affected by the protocol used to receive the traffic.

### Load balancing on outbound carriers

Are facilities available to control the way calls are shared between outbound carriers?

Yes, both simple, sequential routing and more sophisticated algorithms are available.

In the simple case calls are presented to the carriers in the routing table in the fixed, priority order that they appear in the rerouting list for a destination. The switch sends all the calls to the carrier on the top of the list and then, if that call attempt fails (carrier is down or full), to the second carrier, etc.

The other algorithms allow:

- Sending a fixed percentage of calls to a carrier
- Load balancing the calls across a number of carriers

The Route table allows 5 carriers in the rerouting list. This table allows control of the order and the content of the carrier rerouting list. For example, X% of the calls may be sent to the top carrier and 100-X%% of the calls will skip the top carrier and will go directly to the second carrier.

These features may also be combined to create a complex rerouting scheme: load sharing with fixed overflow routing, load sharing with overflow to another load sharing, etc.

Note that in this case a Carrier may be a VoIP trunk to any IP address.

## Protocol support

### Protocols supported other than C7 / SS7 and VoIP protocols

List which non C7 / SS7 TDM protocols their solutions supports. Examples of such protocols includes Euro ISDN, DPNSS, 1TR6.

Other protocols supported are: Euro ISDN (many country variants), CAS (many MFC/R2 variants)

### Inter-working of TDM to TDM protocols

Describe which supported TDM signalling protocols can inter-work with each other. For example can Euro ISDN signalling inter-work with DPNSS? Can C7 ETSI ISUP Version 1 inter-work with UK ISUP?

All supported TDM protocols can fully inter-work with each other.

### Inter-working of VoIP to VoIP protocols

Describe which supported TDM signalling protocols can inter-work with each other. For example can SIP inter-work with H.323 version 2?

All supported VoIP protocols can fully inter-work with each other. SIP signalling inter-works with any H323 version.

### Inter-working of TDM to VoIP / VoIP to TDM protocols

State which of the supported TDM signalling protocols can inter-work with which supported VoIP signalling protocols and vice versa.

All supported VoIP and TDM protocols can fully inter-work with each other. This is one of the design principles of the IPNx switch.

### Inter-working with other vendors

a) State and describe any interoperability testing between other vendors.

Very little formal interoperability testing has been done with other vendors. However, WTL have yet to encounter another vendor in our extensive installed base with whom we have been unable to interconnect.

## Supplementary services

### TDM supplementary services supported

State which supplementary services each implementation of a TDM protocol supports.

| SS7 Function/service | IPNX |
|---|---|
| **Basic call** | |
| Speech/ | Y |
| 3.1 kHz audio | Y |
| 64 kbit/s unrestricted | Y |
| Multirate connection types | |
| | |
| Signalling procedures for connection type allowing fallback capability | Note 1 |
| Compatibility procedure | Y |
| Confusion procedure | Y |
| Simple segmentation | |
| User part availability control | Y |
| Propagation delay determination procedure | Note 1 |
| Dynamic echo control procedure | Note 1 |
| Tones and announcements | Y |
| MTP pause and resume | Y |
| Access delivery information | Note 1 |
| Transportation of User teleservice information | Y |
| **Supplementary services** | |
| DDI | Y |
| MSN | Y |
| CLIP/CLIR | Y |
| COLP/COLR | Note 1 |
| MCID | |
| Sub-addressing | Y |
| Terminal portability | Y |
| Call forwarding | Note 1 |
| Call deflection | Note 1 |
| Call waiting | |
| Call hold | |
| Conference calling | |
| Three party service | |
| CUG | Note 1 |
| MLPP | |
| UUS, Service 1 (implicit) | Note 1 |
| UUS, Service 1 (explicit) | |
| UUS, Service 2 | |
| UUS, Service 3 | |

Note 1: Although the IPNX does not implement these services, it passes transparently the associated parameters.

ISDN Services: basic call procedures are all supported plus Charging Pulse and Date & Time on Connect.

### VoIP supplementary services supported

State which supplementary services each implementation of a VoIP protocol supports.

All SIP basic procedures including Registrar are supported.

All H323 basic procedures including RAS are supported.

### Inter-working of supplementary services on TDM to TDM protocols

State which supplementary services can be inter-worked between different TDM protocols. For example can CLI be inter-worked between ETSI ISUP version 2 and Euro ISDN.

Supplementary services are passed transparently by the IPNx allowing full inter-working. For ISDN this includes CLI, Subaddressing, Bearer Capability, all contents of the ATP (including Display Information and Progress Indication) the UTI and the HLC.

### Inter-working of supplementary services on VoIP to VoIP protocols

State which supplementary services can be inter-worked between different VoIP protocols. For example can CLI be inter-worked between SIP and H.323.

Supplementary services are passed transparently by the IPNx allowing full inter-working. For VoIP this includes CLI, Subaddressing, Bearer Capability, all contents of the ATP (including Display Information and Progress Indication) and the UTI but not the HLC.

### Inter-working of supplementary services on TDM to VoIP / VoIP to TDM protocols

State which supplementary services on which TDM signalling protocols can inter-work with which supplementary services on which supported VoIP signalling protocols and vice versa.

Supplementary services are passed transparently by the IPNx allowing full inter-working. For VoIP to TDM and vice versa this includes CLI, Subaddressing, Bearer Capability, all contents of the ATP (including Display Information and Progress Indication) and the UTI but not the HLC.

### Inter-working with other vendors

a) State and describe any interoperability testing with other vendors.

Very little formal interoperability testing has been done with other vendors. However, WTL have yet to encounter another vendor in our extensive installed base with whom we have been unable to interconnect.

## Features and services

### IP voice VPN

Describe any IP voice VPN solution provided along with a list and description of the features provided by the solution. Include the following in the description:

a) VoIP protocols supported.

b) Can the IP voice VPN solution be used for single users i.e. home workers.

c) Can the IP voice VPN solution be used to serve users on a traditional PABX who access the VoIP network and IP voice VPN solution via a CPE VoIP gateway.

d) What is the maximum number of customers that can be served by the proposed IP voice VPN solution.

e) What is the maximum number of users that can be served by the proposed IP voice VPN solution.

Also describe any additional hardware, software and licences that would be required to operate the IP voice VPN service.

IP voice VPN services can be created using the highly flexible numbering plans of the IPNx. Multiple overlapping plans may be created, distinguished by the Company ID, therefore, an independent numbering plan can exist for each company. Routing may be done on any length of number up to 16 character phone numbers.

 

### Number translation services

Describe any solution offered to enable number translation services such as freephone and premium rate numbers.

Number translation services are supported by the IPNx and are in use by many customers. Flexible number matching is used to translate the called number. This is tied to a highly flexible rating system to allow freephone or Premium Rate services. For Premium Rate services the auto announcement of the cost of the call is supported. This is required by most countries' telecom regulations.

### Local Number portability

Describe the solution provided to allow all UK and European regulatory requirements with regards to local number portability to be met. This description should include any additional hardware and software requirements.

Local Number portability supported

### Number portability

Describe any solution provided to allow the portability of non-local numbers. Examples include freephone and premium rate numbers.

Number portability not supported

### Emergency services

Describe how access to emergency services are provided from the IP / VoIP network to the PSTN side.

Emergency services not supported

### Legal intercept

Describe the solution provided to allow all UK and European regulatory requirements with regards to legal intercept to be met. This description should include any additional hardware and software requirements.

Legal intercept services not supported

### Voicemail / Unified messaging

Describe any solution provided which provides voicemail and unified messaging services.

Voicemail services not supported

### Network ACD

Describe any solution provided which allows a network based ACD service to be offered to customers. This description should provide information on statistics packages as well as the call routing functionality of an ACD system.

The IPNx implements a full automatic call distribution for customers who want to offer their audiotext or operator services via a single number.
Sophisticated options decide where incoming calls are routed for answering:

- Time of day – up to 7 time periods per day
- Day of week - 7 different types of day may be defined
- Capacity of destination
- CLI of caller
- DDI that call came in on
- Number of calls already in the queue waiting to be answered

Careful control over call queues is possible even if queues are spread over multiple IPNxs and even if they are in physically different locations. Because the algorithm runs on a central host there is a guarantee of global control on queuing and fallback. This allows all call answering resources to be used to their optimum. Call distribution is handled fairly so incoming calls are presented to each operator in turn (not always to the first in the list).

If a valid destination is not available the call may be held whilst playing music or a suitable voice message until an operator is available. There is also the capability to tailor these messages to the waiting caller. For example, “You are currently 3rd in the queue”. This is also an opportunity to play customised advertising messages to the caller.
Finally, if all the queues are full the call is routed to a final recorded message service which would normally ask the caller to try again later.

### Contact centre solutions

Describe any solution provided which allows a network based contact centre service to be offered to customers.

Contact centre solutions not supported

### IVR

Describe any IVR solution provided and how it integrates into other applications.

The IPNx offers extensive, integrated IVR support. This is mainly aimed at offering Pre-Paid and Call Back services to subscribers. The IVR system may support multiple languages and allows IPNx owners to record their own prompts.

### Voice conferencing

Describe any voice conferencing solution provided.

Voice conferencing services not supported

### Video conferencing

Describe any video conferencing solution provided. Examples include H.323 to H.320 gateways, H.323 gatekeepers, etc.

Video conferencing services not supported

### Multipoint Control Unit (MCU) / Bridging

Describe any MCU / Bridging products / solutions provided. This description should clearly state if the product / solution can be used for voice, video or both.

MCU services not supported

### Outbound calling

Describe any solution provided which allows the use of a network based outbound calling service. Include within any description information as to the applications use for home workers, small offices and larger offices. All of which may operate as a single team working from the same outbound dialling database.

The IPNx has an interface which allows third party call control (WebConnect). This has been used to create automated calling systems. WTL can assist the customer in the design of a PC program which passes information to the IPNx to allow it to make programmed calls and to monitor the results.

### Text to speech / Speech to text

Provide information on any applications, features and services which provide a text to speech and / or speech to text functionality.

TTS services not supported

### Voice XML

State if the solution supports Voice XML and if so which version(s) they support. Describe what functionality and control Voice XML scripts can have over the various components of the solution.

Voice XML services not supported

### API's & programming interfaces

State and describe what, if any, API's and programming interfaces the solution supports. Describe what functionality and control such API's and programming interfaces can have over the various components of the solution.

The IPNx has a number of APIs which allow the switch owner different levels of control over the switch:

Web Connect – allows external call control of the IPNx. Typical uses include pre-programmed outbound calls, Click to Talk buttons on web sites, on-line purchase of telephone services

Wt Agent – allows manipulation of the IPNx database by external processes. Typical uses include the reading and setting of customer account balances (especially in Pre-Paid environments).

CDR FTP - daily files exist of the very comprehensive Call Detail Records of the IPNx switch. External processes may collect these files. For example, a PC utility is also available which downloads these records reliably into a SQL database for further processing.

Log files – a series of detailed log files may be produced about most aspects of the switch operation. These are available for downloading into external processes.

Cslog – a streaming socket for real time applications which need immediate call information. Typical use of this is a real time billing process.

### Other applications and services

Provide information on other related applications, features and services offered.

The additional features offered by the IPNx are mainly in the areas of call rating, Pre-Paid services and VoIP compression.

## System architecture

### System architecture

Describe the system architecture of the proposed offering(s). This description should clearly show the following:

a) How CPE gateways are directed to the softswitch. I.e. static routing, gatekeepers, proxy servers, TRIP, etc.

b) Redundancy of components and nodes

c) System nodes and how the communicate with each other.

d) The purpose of each system node.

**IPNx**

The IPNx is an industrial computer integrated offering all the features required of a small to medium telecom switch in a single box. The modular architecture allows various interface or resource cards to be plugged in as the application requires. The switch operating software has been designed to run each chassis as a self-contained switch regardless of the mix of traffic it is handling. In addition any number of IPNx switches may be connected in a network. They will operate in a distributed fashion (each handles its own routing for example although routing information is shared between the switches).

Rittal - 8 Slot 9U Chassis
Quad Hot Swap 300W AC Power Supply

Sun Sparc Processor Card with

- 650MHz Ultra SPARC IIi CPU
- 512 KB Cache
- 1 GB Memory
- 4 MB Flash
- 2 x 10/100M Ethernet
- NEBS Level 3 compliant

2 x 160 GB Hard Disks
Ultra Wide SCSI RAID1 Subsystem

i. **Chassis Specification**.
   - **Rugged, rack-mountable system chassis with:**.
     - 8-slot CompactPCI backplane (supports PICMG CompactPCI Full Hot Swap standard).
     - Hot swap fans (dual fan trays optional).
     - EMC-tight front door.
     - IEEE 1001.11 rear transmission board support.

ii. **Physical characteristics**.
   - **Dimensions:**.
     - 9U 19" rack-mount standard.
     - 17" W x 18.5" D x 15.75" H.
     - 435 mm x 470 mm x 400 mm.
   - **Weight:**.
     - 78 lbs..
     - 35.4 kg.

iii. **System Power**.
   - Hot Swap dual/quad (n+1) AC or DC power supplies 300 W with dual fan and power connector per pair
   - Maximum inrush current: 30A
   - Maximum consumption (230V): 7.5A

iv. **Environmental**
   - **Operating Range:**.
     - Temperature: 5°C to 40°C.
     - Relative Humidity: 20% to 80% at +20℃ to +40℃, non-condensing.

v. **Agency Approvals**.
   - CE Declaration of Conformity (EN 50081-1, EN 50082-1, EN 60950).
   - FCC Part 15 Class A.
   - VCCI Class A.
   - UL 1950.

**Reliability**
Switch Availability: 99.9985% (based on field data for 2003)
Minutes down per 10,000 minutes running: 15.22 (based on field data for 2003)
MTBF Figure for the CP1500 Processor Card 151,479 Hours
MTBF Figure for TP1610 16 E1, 480 channel VoIP Card 174,672 Hours
MTBF Figure for TP1610 8 E1, 240 channel VoIP Card 232,774 Hours

**Communications Cards**
**ISDN, E1, SS7** - Audiocodes TP1610 and/or Brooktrout PRI-CPCI NS301
**VoIP** - Audiocodes TP1610 (120 - 480 channels)
**Inter-chassis** - Kallastra KeyTrunk313 (1024 channels) or Keytrunk315 (2048 channels)

**SoIP**

SuperMicro – 4 x PCI Slot 4U Chassis
Triple Redundant 760W Power Supply
ATX motherboard with

AMD Opteron DP 246 Processor 2GHz, 1MB Level

- 512 KB Cache
- 1 GB Memory 400MHz DDR ECC
- 2 x RJ45 ports, Gigabit Ethernet, Supports 10BASE-T, 100BASE-TX, and
- 1000BASE-T
- 2x64-bit 133/100MHz PCI-X, 2x64-bit 66MHz PCI-X, 2x32-bit 33MHz PCI
- ATI Rage XL 8MB PCI Graphics Controller
- PS/2 keyboard and mouse connectors, up to 5x USB 1.1 ports

4 x 160 GB Hard Disks
3 disks (SATA) RAID5 Subsystem
Pass-through disk (SATA also) for dual boot option

vi. **Chassis Specification**.
- **4U rack-mountable system chassis with:**.
  - 4x 5000 RPM Hot-Swappable Cooling Fans
  - 2x 5000 RPM Hot-Swappable Rear Exhaust Fan
  - Locking Bezel with Filter
  - Power Switch, Reset Switch and 6 LED Indicators

vii. **Physical characteristics**.
- **Dimensions:**.
  - 4U 19" rack-mount standard.
  - 17.2" W x 25.5" D x 7.0" H.
  - 437 mm x 648 mm x 178 mm.
- **Weight:**.
  - 65 lbs..
  - 29.5 kg.

viii. **System Power**.
  - 760W Triple-Redundant AC power supply with PFC [ 24-pin, (8-pin, 4-pin) =12V ]
  - AC Voltage: 100 - 240V, 50-60Hz, 14 Amp Max
  - DC Output: 5V + 3.3V ≤ 200W
  - +5V: 36.0 Amp, +5V standby: 3.5 Amp, +12V: 50.0 Amp (combined), -12V: 1.0 Amp, +3.3V: 36.0 Amp

ix. **Environmental**
- **Operating Range:**.
  - Temperature: 10 - 35℃ (50° to 95° F)
  - Relative Humidity: 8 - 90% non-condensing

x. **Agency Approvals**.
  - CE Declaration of Conformity (EN 60950, EN55022: 1994, EN55024 : 1998, EN61000-3-2/EN61000-3-3)
  - Canada/USA (UL60950-CSA60950, Title 47 CFR Part 15)
  - International (IEC 60950, CISPR 22, CISPR 24, IEC61000-3-2)

**Reliability**
Switch Availability: 99.9985% (based on field data for 2003)
Minutes down per 10,000 minutes running: 15.22 (based on field data for 2003)
**Communications Cards**
**VoIP** / **ISDN** / **SS7** - Audiocodes TP260 (30 - 240 channels)

## Operating system

State which operating system(s) is used for each component within the system architecture. This also applies to platforms running any of the proposed applications.

The IPNx operating system is Solaris 9

# Quality of service

Describe what IP quality of service (QOS) mechanisms the products provide. This description should explain if it is possible to have separate QOS settings for the VoIP signalling and the media streams.

The IPNx can use the ToS bit to prioritise voice traffic in a mixed voice and data network. Also the IPNx always uses the same IP ports to transmit VoIP. This allows these ports to be prioritised by routers over other traffic on other ports.

# Security

### Call verification and authorisation

Describe how outgoing VoIP calls from the IP network to the PSTN are verified and authorised.

Outgoing VoIP calls may be authorised in a number of ways:

SIP Digest authentication (RFC 2617)

Caller's Ip address

DDI called

CLI of the user

A PIN number may be checked (this may be explicitly requested or may be built into the user's call request).

### Encryption of the VoIP signalling messages

Clearly state what if any encryption of the VoIP signalling is possible.

VoIP signalling messages are not encrypted.

### Encryption of the VoIP media stream

Clearly state what if any encryption of the VoIP media stream is possible.

VoIP media stream are not encrypted.

### Unauthorised access

Clearly describe how unauthorised access to the various components and nodes of the solution (C7 / SS7, IP Centrex and any applications) is provided.

The IPNx has a management and configuration interface which uses ssh for log in and is password protected.

# Scalability

### Call processing scalability for softswitch

Provide information regarding the performance of the core softswitch solution (excluding applications). This information should include but is not necessarily limited to:

a) Maximum calls per second / BHCA
b) Maximum number of sustained calls.
c) Storage capacity of any databases related to the core solution
d) Maximum number of users per node.
e) Maximum number of E1's per media gateway (M.G.).

f) Maximum number of VoIP calls per media gateway.

g) Maximum number of media gateways per media gateway controller (MGC).

Any other information concerning core softswitch solution scalability should be provided.

a) BHCA 300,000

b) Max calls 3360

c) Storage: no relevant limit

d) Max users: same as max number of calls

e) Max E1s: 112 for single redundant system expandable to 2048 E1s for multiple linked redundant systems

f) Max VoIP calls: 960

g) System is not based on MG and MGC. Maximum of linked systems is 64.

### Applications scalability

For each application, feature or service provided information concerning its scalability should be provided. This information should include (where applicable) but is not necessarily limited to the following:

h) Maximum calls per second / BHCA

i) Maximum number of sustained calls.

j) Storage capacity

k) Maximum number of users.

l) Maximum number of groups (for applications such as ACD, etc)

m) Maximum number of users per group

Any other information concerning application scalability should be provided.

h) Maximum calls per second / BHCA : 300,000

i) Max calls 3360

j) Storage: 4Gb

k) Max users: same as max number of calls

l) Max groups:128

m) Users per group: no limit

# Reliability

Clearly describe the reliability of the products, applications, features and services described. This description should include (but is not necessarily limited to):

a) Average down time per year.

The only single point of failure for the IPNx is the CPU card (although this may be avoided in the redundant configuration outlined here). The MTBF for the CPU card is 151,479 Hours.

The other critical components are the VoIP and E1 cards:
MTBF for TP1610 16 E1, 480 channel VoIP Card 174,672 Hours
MTBF for TP1610 8 E1, 240 channel VoIP Card 232,774 Hours

 

# Approvals

a) Provide a list of all approvals gained for any of the products, applications, features or services that described.

The IPNx and the communication cards that make it up have the relevant telecom and safety approvals. This includes:

- CE Declaration of Conformity (EN 50081-1, EN 50082-1, EN 60950).
- FCC Part 15 Class A.
- VCCI Class A.
- UL 1950.

# Billing

### Customer billing

Describe how billing for CPE customers is accomplished.

Also provide a description of the possible CDR fields that can be output and also explain if it is possible to configure what fields are output within CDR's.

The SoIP Gateway generates a CDR for every call wherever it is originated (on VoIP, via TDM or SS7). Each CDR contains 50 fields. It is not possible to configure the CDR contents. The fields are defined below.

The CDRs are located in two directories on the switch depending upon the service:

- *Non-calling card* service CDRs are located in /usr2/CDR directory
- CDRs for *calling cards* are located in /usr2/DEBIT directory

CDRs are accumulated in a daily file with the name YYYYMMDD where YYYY is the year, MM the month and DD the day.
CDRs are retained on the switch for 31 days.
The CDRs are also exported (at a configurable interval) to a management PC for off-line processing. This processing may be done by the WTL Billing application or a 3rd party billing package. To allow the latter case a scriptable utility is available which can reformat some or all of the CDR into the style required by the external Billing package.
There is no limit to the length of time CDRs may be retained in the Billing database.

The CDR format is:

## Voice Call CDR format

| Field | Name | Description | # of chars, type, fixed? [note] |
|---|---|---|---|
| 1 | CID | Company ID or provider ID if pre-paid service | 8, A, Y |
| 2 | Unixtime | Time in seconds since 1970 (Unix time) | 9, N, N |
| 3 | Duration | Duration of call in seconds; 0 if no connect | 4, N, N |
| 4 | Switchunit | Switch # on which call originated: 5 digits, first 4 = node #, last digit = sw # | 5, A, Y |
| 5 | Time | Time connect was started or time of disconnect if not connected | 8, T, Y |
| 6 | Date | Date in DD.MM.YY (US format) or DD.MM.YYYY | 10, D, Y |

 

| | | | |
|---|---|---|---|
| 7 | Jobnumber | Job number (for billing purposes. It is also an extra level of security - password - provided by the switch.) | 19, A, N |
| 8 | PIN or CLI | CLI, PIN code or credit card number (depending upon service) | 25, A, N |
| 9 | B-Number | Destination number (international format) | 24, A, N |
| 10 | Outbound line | Out bound Line # (1 - N) per switch | 3, N, N |
| 11 | Outbound carrier | Outbound carrier # on which call was placed | 1, A, Y |
| 12 | Initial rate | Call rate in cents/minute (cents = 1/100 of currency unit) | 5, N, N |
| 13 | Cost | Call cost in cents (cents = 1/100 of currency unit) | 5, N, N |
| 14 | DNIS | Dialed Number Identification Source = City code + DDI (Direct Dial in Number) on which call was received | 24, A, N |
| 15 | Operator number | Telephone number of operator for operator assisted calls. It contains the telephone number of the operator position for operator-assisted calls (debit service only). Although this field is currently unused for business service, it will be present as an empty field in the extended CDR of business calls. | 20, A, N |
| 16 | Balance left | Balance in cents remaining on the account after the call | 9, N, N |
| 17 | Inbound (A-Leg) carrier | For SCX calls (not reported in CDR file), this will be 0 | 4, N, N |
| 18 | Inbound line | According to PORT table | 3, A, Y |
| 19 | Cause value | Disconnection cause value in INX format (ISDN cause=1512+ISDN cause), 0 if none | 5, N, N |
| 20 | Transmit Medium Requirement (TMR) | This numeric parameter represents the type of call. It can take the following values:<br>0: the call is a speech only call (no fax or modem signals)<br>2: the call is a 64Kb/s unrestricted digital call<br>3: the call is an audio call. The audio signal can be speech, fax or modem and any other inband (0-3KHz) signal<br>6: the call is a 64Kb/s unrestricted "preferred" call. This is treated by the INX as a TMR 2 but with the possibility for a subsequent switch to fallback to an audio call quality. The CDR does not reflect whether the fallback has occurred or not. | 1, N, Y |
| 21 | Duration of the A-leg connection | This field will be set for call-through and callback and represents the total connected time since beginning of call. If there are several CDRs recorded during the call (LCR attempts or joint call feature), they will all have this field set with increasing values. The duration is counted from the time the A-leg is connected, it does not include ringing | 8, N, N |

| | | | |
|---|---|---|---|
| | | time for direct or carrier select services where the inbound call is not connected before the B-leg leg is connected | |
| 22 | Telephone number in international format of A-leg leg when there is a callback | This field will only be set on the last CDR when the A-leg is disconnected and only if the A-leg is a callback call. It is intended to facilitate the collection of callback calls for accounting. Note that if the callback call fails, it is reported as a normal outbound call failure (the callback number is in the destination number field). | 24, A, N |
| 23 | Number of busy lines on the A-leg carrier (field 17) at the time of the CDR time stamp (field 5 and 6) | This is the total number of busy lines (inbound or outbound calls, connected or progressing calls) on all lines to the carrier on the switch. In case of callback, the A-leg carrier is the carrier on which the callback call has been successfully placed. | 4, N, N |
| 24 | Number of busy lines on the B-leg carrier at the time of the time stamp | This is the total number of busy lines (inbound or outbound calls, connected or progressing calls) on all lines to the carrier on the switch. | 4, N, N |
| 25 | Total outbound call time in seconds | This is the time between the initial dialing and the disconnection of the B-leg. The duration of the dialing and ringing stage can be deduced by subtracting the B-leg connection time (field 3). The duration of the dialing stage alone cannot be deduced. | 8, N, N |
| 26 | Disconnection origin | 0 = the disconnection is initiated internally (timeout, balance expiry, ...)<br>1 = the disconnection is coming from the A-leg<br>2 = the disconnection is coming from the B-leg | 1, N, Y |
| 27 | Type of PIN | Specifies content of CDR field 8 First letter:<br>A for credit card call (when field 8 contains an Account Number)<br>D debit number from DEBIT.DEBIT_NR (can be a ordinary debit number or a debit CLI (starts with letter 'O'))<br>P pin code from PIN.PIN<br>O CLI from PIN.PIN<br>R redirecting number from PIN.PIN<br>H H323 pin code from PIN.PIN<br>I IP address from PIN.PIN formated as xxx.xxx.xxx.xxx<br>C CSTA from PIN.PIN formated as SSSSSCCC where SSSSS is the switch | 2, A, N |

| | | | |
|---|---|---|---|
| | | name in 5 characters and CCC is the inbound line CSTA in 3 characters.<br>Second letter:<br>N if the PIN or DEBIT number has been created by the switch<br>The field is appended with N if field 8 contains a newly created PIN or DEBIT record. This is used in services where the switch can generate PIN or DEBIT record by itself (REGISTER, RECHARGE)<br>If field 8 is empty, field 27 will also be empty | |
| 28 | CLI | CLI of the call obtained from different sources:<br>for C,O,D inbound carriers, replaced A-number from PIN.CLI according to PIN.CLIUSE.<br>for callback service with PIN.CLIUSE=B or b, replaced A-number from PIN.CLI<br>for tdial test calls, from –a option<br>otherwise unconverted A-number from inbound call | 16, A, N |
| 29 | B-Leg node number | When the B-leg carrier is a VoIP carrier, this field contains the node number to which the call has been sent. Empty otherwise. | 5, A, N |
| 30 | PIN service code | When the call is authenticated by a PIN record, this field contains the PIN.SERVICE field:<br>O for CLI based authentication on O,P carriers<br>C for CLI based authentication on C carrier<br>D for CSTA based authentication on D carrier<br>H for PIN based authentication on DDI service H,U or on H carrier<br>S for CLI based authentication on DDI service S<br>I for CLI and DNIS based authentication on DDI service I<br>B for DNIS based authentication on DDI service B<br>If the call is not authenticated by a PIN record, this field is empty | 1, A, Y |
| 31 | A-leg rate | Initial A-leg rate or 0 if the A-leg is not charged. The A-leg rate does not change during the call. | 6, N, N |
| 32 | A-leg cost | Total cost of A-leg or 0 if the A-leg is not charged. | 9, N, N |

| | | | |
|---|---|---|---|
| 33 | Type of CDR | Indicates the position of the CDR in the call.<br>0 intermediate CDR, A-leg still in progress, B-leg disconnected<br>1 terminating CDR, A-leg and B-leg disconnected<br>2 intermediate CDR, callback A-leg disconnected<br>3 intermediate CDR, no A-leg or B-leg change, just updated balance<br>4 tdial test CDR<br>5 webconnect terminating CDR leg 1<br>6 webconnect terminating CDR leg 2<br>7 transit intermediate CDR (carrier V,W,X)<br>8 transit terminating CDR (carrier S,V,W,X)<br>9 TNS calls<br>Note: when debit cards are used to recharge an account, a CDR will be created in debit CDR file with the balance of the recharging cards, There will be no A-leg and B-leg, just an updated balance. The type of CDR will be 3 with no A-leg and B-leg number, cost or rate. The company ID (field 1) will be set from the card being recharged (the active card), not the recharging card. | 2, N, N |
| 34 | Type of balance | Indicates the type of balance in field 16<br>0 no balance is charged<br>1 debit balance is charged<br>2 pin balance is charged<br>3 company balance is charged | 2, N, N |
| 35 | B-leg format | Indicates if the B-leg number in field 9 is in international format or not<br>B-leg number is unformated<br>B-leg number is formated (processed by inrule1 table) and should be in international format. | 2, N, N |
| 36 | B-leg number heading | Content of ROUTE1.AREACODE field (ROUTE.AREACODE if slice routing is used) which has been used to route the B-leg. Empty if there was not B-leg or if the ROUTE1.AREACODE (ROUTE.AREACODE) field was empty. | 16, A, N |

| | | | |
|---|---|---|---|
| | | This is useful for statistical purposes to sort the calls by selected destinations without having to run pre-processing routines on the CDR.<br><br>Note that the route table should have a meaningful list of destinations for this field to contain useful information, even if the routing is simple: all redundant route records can point to the same carriers. | |
| 37 | B-leg post dial delay | Time is seconds between the end of dialling and the start of the ringing phase.<br><br>The ringing phase starts when an Alerting (or equivalent) message is received from the B-leg. If no Alerting is received, the PDD ends when the call fails or is connected.<br><br>Note that for calls with overlap sending, the dialling ends when the switch 1) detects a timeout on the incoming digits 2) detects a "end-of-dialling" signal from the inbound side 3) receives a Proceeding/Alerting/Connect message from the outbound side.<br><br>In case 1, the timeout is counted as part of the PDD, in case 2, the PDD starts with the "end-of –dialling" signal, in case 3, the PDD will be 0 if an Alerting/Connect message is received. | 8, N, N |
| 38 | CLI nature of address | Holds the CLI nature of address parameter received from the inbound carrier. It defines the type of number stored in field 28. Possible values:<br><br>0 unknown type of number<br>no CLI is provided<br><br>3 national number<br><br>4 international number | 1, N, Y |
| 39 | B-leg routing ID | Content of ROUTE1.ROUTEID field (ROUTE.ROUTEID if slice routing is used) which has been used to route the B-leg. Empty if there was no B-leg or if the ROUTE1.ROUTEID (ROUTE.ROUTEID) field is empty.<br><br>In case of indirect route (ROUTE_COD field used), this field holds the initial route ID, not the ROUTE_COD field. | 6, A, N |
| 40 | A-leg zone | Content of RATE.C_CODE field used to rate the A-leg. Empty if the A-leg is not rated or if RATE.C_CODE field is empty. | 16, A, N |
| 41 | A-leg rate ID | Content of RATE.RATE_ID field used to rate the A-leg. Empty if the A-leg is not rated or if RATE.RATE_ID field is empty In case of indirect rating (RATE_CLI or | 6, A, N |

| | | | |
|---|---|---|---|
| | | RATE_COD used), this field holds the iniital rate ID, not the RATE_CLI or RATE_COD. | |
| 42 | B-leg zone | Content of RATE.C_CODE field used to rate the B-leg. Empty if the B-leg is not rated or if RATE.C_CODE field is empty. | 16, A, N |
| 43 | B-leg rate ID | Content of RATE.RATE_ID field used to rate the B-leg. Empty if the B-leg is not rated or if RATE.RATE_ID field is empty.<br><br>In case of indirect rating (RATE_CLI or RATE_COD used), this field holds the iniital rate ID, not the RATE_CLI or RATE_COD. | 6, A, N |
| 44 | Agent ID | Holds the agentid used for the call. Empty if no agentid was selected. | 3,A,N |
| 45 | Call ID | Holds the call id assigned to the call by the switch software. This call id is a small number (1-9999) assigned cyclically by the switch and maintained for the entire duration of the Aleg. The call id is also used in the log file to tag all log entry of the call. The call id is unique per INX at a particular point of time but is not globally unique: two INX can have the same call id at the same time and the same call id can be reused every 10 minutes or so on a busy INX.<br>This field can be useful for statistical purposes to link several CDR together: consecutive CDRs from the same calling card number having the same calling id field are chained CDR | 4,N,N |
| 46 | Batch ID | Holds the batch id of the card copied from DEBIT.BATCHID. Only set for debit calls, empty otherwise. | 6,A,N |
| 47 | Caller Category | Holds the caller category as received from the inbound call data. It matches the SS7 definition for caller category:<br>10 : ordinary subscriber<br>15 : payphone<br>For non-SS7 incoming calls, the caller category is set by default to 10 unless a payphone is detected via the PIN table in which case the caller category is set to 15. This field can be used to determine if the call was coming from a payphone or not. | 2,N,N |
| 48 | Calling Card Status | Holds the debit card status as calculated by the switch at the start of the call. Copied from<br>DEBIT.STATUS. It is the status of the card which number is in field 8. Can take the following values:<br>- empty: for non-debit CDR | 1,A,Y |

| | | | |
|---|---|---|---|
| | | - F : the card is active and has been used<br>- Z : the card balance is below minimum<br>- E : the card has expired<br>- B : the card is blocked | |
| 49 | Expiration Date | Holds the expiration date of a credit card number as received from the SETUP message or the user. The format is normally MMYY.<br>This field is only used for credit card calls (field 27 = A) | 6,A,Y |
| 50 | Validation Code | Holds the authentication result for credit card (field 27 = A)<br>The following values are defined:<br>0 Passed<br>1 Validation unavailable<br>2 Invalid collect call<br>3 Invalid Account<br>4 Account Barred<br>5 Invalid Authorization Code<br>6 Invalid PIN<br>7 Invalid Account Code<br>8 Invalid Account, No Retry<br>9-255 Reserved | 3,A,Y |
| 51 | QOS A-leg coder | Holds the codec used on A-leg<br>0 The call is not VoIP or the codec could not be determined<br>1 Undefined<br>2 G711 uLaw<br>3 Undefined<br>4 G711 ALaw<br>5 V32 modem relay (deprecated)<br>6 G723.1 6.3b/s<br>7 G726 32kb/s<br>8 G729 8kb/s<br>9 Undefined<br>10 Undefined<br>11 GSM full rate<br>12 G728 16kb/s<br>13 G723.1 5.3kb/s<br>14 Undefined<br>15 T38 fax relay<br>16 G711 fax/modem bypass<br>Note: fax coders (15 and 16) indicates a change to fax mode during the call, it does not give any indication on the success or failure of the fax | 2,N,Y |
| 52 | QOS A-leg RX packet loss ratio | Holds the fraction, expressed in multiple of 1/256, of the RTP packets sent by the peer that were never received. Note that if the switch does not receive any packet, this field will be 0. | 3,N,Y |
| 53 | QOS A-leg RX packet jitter | Holds the average jitter, expressed in multiple of 0.125us, measured on the inbound RTP packets. | 4,N,Y |
| 54 | QOS A-leg TX packet | Holds the fraction, expressed in multiple of 1/256, of the RTP packets sent by the | 3,N,Y |

| | | | |
|---|---|---|---|
| | loss ratio | switch that the peer never received. This field will be 0 if we don't receive RTCP reports from the peer. | |
| 55 | QOS A-leg TX packet jitter | Holds the average jitter, expressed in multiple of 0.125us, which the peer measured on the RTP packets that it received. This field will be 0 if we don't receive RTCP reports from the peer. | 3,N,Y |
| 56 | QOS B-leg coder | Holds the codec used on B-leg. See field 51 for a list of possible values | 2,N,Y |
| 57 | QOS B-leg RX packet loss ratio | Holds the fraction, expressed in multiple of 1/256, of the RTP packets sent by the peer that were never received. Note that if the switch does not receive any packet, this field will be 0. | 3,N,Y |
| 58 | QOS B-leg RX packet jitter | Holds the average jitter, expressed in multiple of 0.125us, measured on the inbound RTP packets. | 4,N,Y |
| 59 | QOS B-leg TX packet loss ratio | Holds the fraction, expressed in multiple of 1/256, of the RTP packets sent by the switch that the peer never received. This field will be 0 if we don't receive RTCP reports from the peer. | 3,N,Y |
| 60 | QOS B-leg TX packet jitter | Holds the average jitter, expressed in multiple of 0.125us, which the peer measured on the RTP packets that it received. This field will be 0 if we don't receive RTCP reports from the peer. | 3,N,Y |
| 61 | Exact call duration | Holds the same information as in field 3 except that the duration is not rounded to the second (rounding threshold is configurable through switch software –XD option).<br>The field is always provided with 2 decimals, even if the duration is 0 (presented as 0.00) | 11,N,Y |
| 62 | IP address of caller | Holds the IP address of caller if the incoming call is VoIP.<br>When provided, the IP address is always presented with 15 digits: XXX.XXX.XXX.XXX<br>For data CDR, this is the IP address of the access router where the data session is established. | 15,A,Y |

| | | | |
|---|---|---|---|
| 63 | Original DDI number | Holds the called address as received from the network before any conversion in the switch software<br><br>Note that for some protocols (only SIP at present), a conversion can take place in the wn kernel before the call is presented to the switch software. | 23,A,Y |
| 64 | Original DDI nature of address | Holds the nature of address of the called address as received from the network before any conversion in the switch software.<br><br>The value of this field follows the SS7 standard:<br><br>0 = unknown<br><br>1 = subscriber number<br><br>2 = unknown<br><br>3 = national number<br><br>4 = international number<br><br>5-111 = spare<br><br>112-126 = reserved for national use | 3,N,Y |
| 65 | Total data transfer | Holds the amount of data (inbound+outbound) transferred during a data session up to the time of the CDR. The field is expressed in Kilobyte = 1024 bytes. | 10,N,N |
| 66 | Total charged data transfer | Holds the amount of data (inbound+outbound) paid so far during a data session. It differs from field 65 because of predictive charging and the rounding of data transfer to the upper multiple of PROFILE.INC_DATA. | 10,N,N |
| 67 | Original CLI number | Holds the calling address as received from the network before any conversion in the switch software<br><br>Note that for some protocol (only SIP at present), a conversion can take place in the wn kernel before the call is presented to the switch software. | 16,A,N |
| 68 | Original CLI nature of | Holds the nature of address of the calling address as received from the network | 3,N,N |

| | | | |
|---|---|---|---|
| | address | before any conversion in the switch software.<br>The value of this field follows the SS7 standard:<br>0 = unknown<br>1 = subscriber number<br>2 = unknown<br>3 = national number<br>4 = international number<br>5-111 = spare<br>112-126 = reserved for national use | |
| 69 | A-leg VoIP call ID | Holds the call ID of the A-leg VoIP call. This field is empty for non-VoIP A-leg carriers. For SIP calls, the call ID is formatted as <id>@<host>.<br>Currently this field is only provided for SIP A-leg carriers. | 80,A,N |
| 70 | B-leg VoIP call ID | Holds the call ID of the B-leg VoIP call. This field is empty for non-VoIP B-leg carriers. For SIP calls, the call ID is formatted as <id>@<host>.<br>Currently this field is only provided for SIP B-leg carriers.<br>Note: this field is shorter than field 69 because the current version of the wn kernel generates short call ID but it may change in future version. The target size of this field for all database application should be 80. | 26,A,N |

## Data Call CDR Format

The Data CDR was introduced to reflect cost applied on RADIUS data sessions. A CDR is generated at the end of the session with the total session details (data transfer and cost). In case of predictive charging, an operational CDR is also generated whenever the user balance is charged at the start or during the session.
Most fields have the same content/meaning as for voice calls with the following exceptions:

| | | | |
|---|---|---|---|
| 2 | Call UNIX time | The time of the start of the session (when the RADIUS Access-Request message was received). | 16,A,N |
| 3 | B-leg | The value of the Acct-Session-Time | |

| | | | |
|---|---|---|---|
| | connection duration | RADIUS attribute when it is present in the message that triggers the CDR. | |
| 5 | Call time | Time of the start of the session (when the RADIUS Access-Request message was received). | |
| 7 | Job code | Session ID obtained from the RADIUS attribute Acct-Session-Id. The session is uniquely identified with the combination of this field and the field 62 that contains the IP address of the access router. Both values form the search key in the session table. | |
| 8 | Pin code | Attribute value that was used to find the PIN record: Calling-Station-Id or User-Name | |
| 9 | B-leg telephone number | The network name to which the user logged on; it is taken unchanged from Called-Station-Id. | |
| 12 | initial B-leg rate | The cost per Megabyte charged on the session. Setup cost is not included. | |
| 13 | cost of the call | The total cost of the session, including setup cost. For data session update CDR (predictive charging in use), this is the cost applied on the balance as pre-charge at the time of the CDR, not including previous pre-charge. | |
| 14 | DDI | DDI used to service the call. formatted as DDI + <IP_Address_of_RADIUS _client_as_XXX.XXX.XXX.XXX>. The IP address of RADIUS client is the IP address of the originator of the RADIUS message, not to be confused with the IP address of the access router in field 62. | |
| 16 | Balance left | The exact balance at the end of the CDR after all cost applied | |
| 21 | Partial A-leg connection duration | The duration of the session measured by the switch (difference in second between the time of the CDR and the initial session creation time). | |
| 22 | A-leg callback number | The MSRN number if CLI-based rating is used. | |
| 26 | Disconnection flag | For data session end CDR, a value of 0 means that the switch disconnected or rejected the session. A value of 1 means the session ended normally. For data session update CDR, this is field is not meaningful and will be 0. | |

| | | | |
|---|---|---|---|
| 28 | CLI | The content of the Calling-Station-Id attribute. | |
| 33 | Type of CDR | 19 Data session update CDR: PIN/COMP balance updated because of predictive charging or setup cost. Field 13 is the cost applied.<br>20 Data session end CDR: PIN/COMP balance updated based on total session data consumption. Field 13 is the total session cost (including setup cost). Note that a session that has been rejected will generate a CDR of this type with 0 duration, cost and data transfer (fields 65 and 66). | |

### Interconnect billing for C7 / SS7

Describe how billing for C7 / SS7 interconnect is accomplished.

Also provide a description of the possible CDR fields that can be output and also explain if it is possible to configure what fields are output within CDR's.

The IPNx generates a CDR for every call wherever it is originated (on VoIP, via TDM or SS7).

Each CDR contains 60 fields. It is not possible to configure the CDR contents. The fields are defined above.

### Interconnect billing for VoIP

Describe how billing for VoIP interconnect is accomplished.

Also provide a description of the possible CDR fields that can be output and also explain if it is possible to configure what fields are output within CDR's.

The IPNx generates a CDR for every call wherever it is originated (on VoIP, via TDM or SS7). Each CDR contains 60 fields. It is not possible to configure the CDR contents. The fields are defined above.

# Fraud detection

Any fraud detection abilities / facilities of any component of the core softswitch, IP Centrex and other applications should be described. This ability may be real time or otherwise. If non-real time the time delay before possible frauds are announced should be stated.

This description should include any additional hardware and software requirements.

The IPNx has fraud detection mechanisms associated with subscriber authorisation. For example, the number of bad PIN number attempts allowed can be configured and an alarm raised if this is exceeded. Account limits may also be set (based on value or calling time).

# Operations and support

### Management system

Provide information relating to how any proposed products are managed. This information may relate to management systems, web interfaces or simple command line terminal sessions.

A number of PC applications are available to assist the management of the IPNx. In addition all operations may also be carried out using the command line interface of the switch.

### SW Admin - Multi-Level Telecom Business Management

A clear, easy-to-use PC-based business management application has been developed to work with the WTL range of switches. This package is designed to allow switch owners and their resellers immediate access to the key features driving their businesses. As usual with WTL it makes no difference if the traffic is VoIP or TDM and a wide mixture of services can be supported from one single application.

- New Customer sign-on
- Rate management
- New Reseller sign-on
- Invoice generation
- Secure, partitioned access
- Carrier reconciliation
- Remote access for Resellers
- Traffic reports
- Multi-level business model
- Balance/account management
- Calling Card Creation
- Batch & PIN Management

### SW Config – web based user friendly switch setup

The IPNx switch is well known for the enormous flexibility and many service options that it offers. Until now this has normally been configured using the switch tables and command line interface.

Using years of experience of how to set up telecoms equipment WTL has released a secure, web-based, real-time, GUI interface. SW Config is designed for engineers who need to set up or modify one or more WTL switches.

SW Config gives the network engineer a graphical view of the switch and allows them to click on the card and/or port to drill down and set the right parameters.

Intelligence is built into the application so that every data entry is value-checked to make sure that it is valid and that bad data cannot be sent to the switch. Drop down menus offer the valid options in a given situation and commonly used default configurations are available.

Parameter files are no longer edited directly and an improved table editor is also included.

Features:

- All entries value checked
- Supports IPNx
- Parameters can be changed in real time
- Works local or remotely
- Imports existing switch configurations
- Supports multiple switches
- Profiles available for common set-ups (for example, connection to Cisco)
- Remembers last 3 configurations (allows roll back)

Benefits:

- Easier interface makes reconfiguration quicker
- Value checking and drop-downs to pick only valid entries reduces errors
- Risk of switch down time due to bad configurations is almost zero
- Operators only have access to the information that they are allowed to see
- Reverting to previous known working configuration is always possible in case of problems
- Default configurations have been pre-loaded to speed up common tasks
- Hot re-configuration is now supported – all changes can be made on the fly OR
- Configuration files can be prepared in advance for activating at a later time.

# SW HelpDesk

SW HelpDesk allows operators to answer typical call-related user questions. Call records can be viewed, filtered and queried. The application works in two modes: (i) querying directly to the WTL switch for current call information (real time, for calls from the last 30 days); or for archive enquiries, (ii) querying an SQL database which has already been populated with CDR data from the switch. In archive mode there is no limit to the period of historical data held.

The examination and analysis of call traffic on one or more WTL switches is supported, allowing a centralised customer support function for a large network.

For most simple problems SW HelpDesk searches the CDR file. However, for more detailed fault finding, it is possible to drill down to see the more comprehensive information stored in the call log file.

SW HelpDesk also allows the operator to adjust customer balances & profiles (subject to correct login), which means that if a problem is found, it can be fixed immediately.

## Features

- CDRs or logs can be downloaded directly from the switch(es), or from SQL Server database.
- Filter call information by PIN number, destination number, outbound carrier or company ID.
- Check balance and rates for a particular PIN and change (if authorised using the Security feature).
- Search for any text string comprising alphabetic and numeric characters.
- Searches can be performed on displayed information, or “live” on the switches.
- Also filter the displayed information using standard SQL queries.
- Supports calls by ‘hosted’ customers

## Display Options

- Default shows last 10 minutes activity
- Also search log by following filters:
- On finding desired call, double-click to get full details

    - Time / Date
    - PIN
    - Destination Number
    - Outbound Carrier
    - Company
    - Zero cost calls
    - Errored calls only
    -

- Allows selection of which switches to include in search (up to 16 simultaneously). Switches may be grouped – allows creation of geographical zones.
- Export highlighted call details in comma separated file to Excel or other application

## Security

SW HelpDesk supports 3 levels of Security:

a) Read Only,
b) Modify user balances up to set limit,
c) Modify balance with no limit. An audit log is kept of all balance modifications by login of the user who made them.

### SW Monitor

Running a telecoms network relies on accurate, up to date information. All relevant data must be displayed in easy to understand ways. SW Monitor from World Telecom Labs does this by monitoring the health and operation of any number of our switches.

- Graphical display of switches and trunks
- Simple colour coding to indicate alarms received
- Ability to view recent alarms
- Choose which switches to monitor and how frequently
- Instant view of total calls, calls & ASR per switch and calls per trunk
- Emails sent if certain events occur

Sophisticated Email-based reporting supported:
SW Monitor is also designed for unattended operation. There is a highly configurable system of email-based reporting.

- Configurable limits before emails are sent
- Different action rules for day and night (so emails can be sent when engineers off-duty)
- Multiple engineers' email addresses can be included
- Different alarm levels trigger emails to different people (for example, only send major alarms to manager)

### SNMP Support

Describe how the proposed equipment can be incorporated into a 3rd party SNMP based management console.

At present, WTL has not developed its own SNMP based management system and we do not have a preferred 3rd party supplier for a SNMP management console (customers have successfully used their own choice of SNMP management consoles).
The decision was taken that a proprietary method of equipment management would be used for all WTL equipment (wn_mon). This has been developed to be highly efficient, robust, secure and, most importantly, uses low bandwidth when managing remote equipment.
However, it is possible to include WTL equipment in an SNMP-based management scheme. This is achieved by the use of standard MIBs contained within elements of the WTL equipment. These MIBs can be used to give a real-time view of equipment status, trunk state, IP networking, packets sent and received, data errors and many other characteristics.

**MIB Support by Equipment Type**
The WTL MIB support is organized as follows:

| Equipment Type | System MIB | Trunk MIB | RTP MIB |
|---|---|---|---|
| **IPNx** | **Y** | **Y** | **Y** |
| **SoIP** | **Y** | **Y** | **Y** |
| **PVx / Soft IVR** | **Y** | **N** | **N** |
| **IPNx DG** | **Y** | **Y** | **Y** |
| **IPNx AG** | **N** | **N** | **N** |

The Solaris Operating System used in WTL switch products supports MIB I and MIB II (RFCs 1156 & 1213). This includes query-able information on the following equipment attributes:

| | |
|---|---|
| ▪ System (system name, time etc) | ▪ IP |
| ▪ Interfaces | ▪ TCP |
| ▪ Address Translation | ▪ UDP |

 