SAMIP® Communication Server

SAM4000E

The SAM4000E is a high-capacity, integrated communication solution designed for large enterprises requiring robust and scalable telephony systems. Key features include:

- **High Scalability:** Supports up to 2,560 TDM ports, accommodating extensive organizational growth.
- **VoIP Integration:** Enables up to 320 simultaneous VoIP channels, facilitating seamless networking across multiple sites and reducing communication costs.
- **Redundancy for Reliability:** Features hot redundancy for central units and power supplies, ensuring continuous operation in mission-critical environments.
- **Flexible Connectivity:** Compatible with traditional telephone lines (ISDN, PSTN), traditional extentions (analogue, E&M) and VoIP lines from SIP carriers, offering versatile connection options.
- **Comprehensive Telephony Services:** Offers a wide range of services, including Voice Mail, Automatic Operator, Enhanced Call Routing, Conference Server, and fixed-mobile convergence*, addressing diverse communication needs.
- **Simplified Management:** Administrators can manage the system via web interfaces or dedicated clients, streamlining configuration and maintenance tasks.
- **Secure Communications:** Ensures secure communications via TLS and SRTP protocols, safeguarding data integrity and confidentiality.

The SAM4000E's modular design and extensive feature set make it a robust solution for enterprises seeking a reliable and adaptable communication platform.

*with BRAVO integration.

SAM 4000E is the integrated communication solution of the SAMIP® family for the Enterprise world and Critical Infrastructure market, designed to offer high reliability in the most critical conditions.
SAM 4000E is designed to provide high-performance communication services to a huge number of both IP and TDM users, managing up to a maximum of 2560 "line / trunk" ports. The hot redundancy of the main parts makes the **SAM 4000E** apposite for "mission critical" applications, particularly in environments sensitive to the high availability requirements of the communication system.
SAM 4000E is the communication server for large companies, in which business processes cannot ignore the integration between voice and data, which guarantees each user the same service independently of the device and applications used. Thanks to its flexibility,
SAM 4000E adapts perfectly to multi-site or campus-type scenarios in combination with the other systems of the SAMIP® family.
SAM 4000E can be used in hosted/managed mode also thanks to the centralized management system designed to easily and transparently connect a large number of users spread over different locations.
The integration of **SAM 4000E** with vertical applications completes the system by tying it to the activities, processes and specific needs of the company and the market.

Main Features

SAM 4000E is designed to connect to any type of traditional telephone line (ISDN or PSTN) as well as to VoIP trunk lines provided by ISPs and SIP carriers. The system modularity permits you to configure the **SAM 4000E** according to the business needs of the moment while ensuring the ability to respond to the future operational needs of the organization.

VoIP resources (up to 320 simultaneous communication channels) can be used for networking in multisite scenarios and to use all the services available on IP terminals. The integrated SIP Proxy makes it possible to connect SIP standard telephones and clients installed on PCs, smartphones or tablets, which can also be reached via the Wi-Fi network.

The large number of features offered by **SAM 4000E**, such as Voice mail, Automated Attendant, Conference Server service, Attendant console with accessibility options for blind or low-vision users, TAPI-based services, etc. allow you to meet all the communication needs of a modern organization.SAMIP® systems can perform advanced gateway functions in combination with the BRAVO® solution, capable of enabling Virtual-PBX and Unified Communication scenarios, as well as Hybrid Working applications.

	BASE	MAX
IP USERS	16	512 / 10000 (WITH HP-UC SERVER)
TDM USERS	320	2560
AVAILABLE SLOTS	15	126
EXPANSION MODULES	-	7

Maximum equipment not reachable at the same time

SYSTEM CAPABILITY

External lines

- up to 60 BRI T0/S0 ISDN Ports
- up to 120 FX0 Ports (analog trunk lines)
- up to 2016 FXS Ports (analog extensions)
- up to 8 PRI E1/T1 ISDN Ports.

Telephone extensions

- up to 512 IP 51xx Series extensions
- up to 512 IP NETfon Bluelight extensions
- up to 512 IP SIP extensions
- up to 2016 analogue extensions.

DSP resources

up to 320 VoIP channels with DSP Farm units.

SYSTEM DATA

BHCC

Up to 30.000 Busy Hour Call Completion.

Central Unit (redundant)

Intel Xscale main processor, 4096-channel, 32-wire PCM TDM switching matrix, ARM auxiliary processor.

Operating system

Linux.

TECHNICAL SPECIFICATIONS

Dimensions and weight

Subrack (mm) : H.225 x L.490 x P.240; 9Kg.

Mounting

on 19 ‘’ data rack (5U for each subrack).

Connectors

- Connection with the LAN network: 10/100 Base-T Ethernet port switch, IEEE 802.3 / 802.3u, AutoSense, Automatic Cable crossover (MDI / DI-X), PoE 802.3af
- AUX port for inputs and outputs to the range (2 DIN, relay output)
- COM-RS232 port
- 2 LVDS interfaces for Extended modules (expandable up to 8 and redundable).

Power Supply

- Redundant DC/DC converter
- DC IN: Input of the supply voltage -48Vdc (-38Vdc: -60Vdc) max current for single subrack 9A
- Integrated AC/DC power supply (opt.)

Consumption

- Estimated data for the base system:
- SAM4000E Full IP configuration: 97 W
- SAM4000E TDM configuration: 154 W
- Peak traffic consumption: 220 W.

CERTIFICATIONS

- CE marking (European Community directive 1999/05 / EC: ETSI EN 300 386; EN 55022; EN 55024; EN 61000-3-2; EN 61000-3-3; EN 60950)
- Interoperability on TIM’s Hyperway network (network of SAMIP exchanges)
- PSTN interface (ETSI TBR-21)
- ISDN-PRA public network interface (ETSI TBR-4; ETS 300102; ETS 300125)
- ISDN-BRA Public Network Interface (ETSI TBR-3; ETS 300102; ETS 300125)
- Caller ID (ETS 300 778-1)
- Environmental tests (ETS EN 300 019 class 3.2 in use) performed in the internal laboratory.

VoIP FEATURES

- Signalling
  - SIP (RFC 3261)
  - H.323 (ITU-T)
- Voice encodings
  - g.711 (ITU-T)
  - g.729a/b (ITU-T)
  - g.723 (6.3) (ITU-T)
  - g.723 (5.3) (ITU-T)
- Network Echo Canceller (ITU-T G.168)
- Voice Activity Detection (VAD)
- Comfort Noise Generation (CNG)
- Configurable Jitter Buffer
- QoS via DiffServ (RFC 2474/2597/2598)
- QoS via TOS (RFC791)
- H.323 gatekeeper support (ITU-T)
- Trunk lines backup
- Music on hold.

APPLICATIONS AND CTI

- TAPI 1st party (Microsoft
- TAPI 2.2)
- TAPI 3rd party (Microsoft
- TAPI 2.2)
- Personal Telephony CTI
- Supervisor Telephony CTI
- Contact Center:
  - Multichannel ACD
  - IVR
  - SMS services
  - WEB services
- Click to chat:
  - Click to talk (VoIP)
  - Call me back
  - Co-browsing
- CRM-oriented
- Metrics reports
- Call Flow Editor:
  - Integrated VM/AA
  - Facility UM
- Unified Messaging:
  - Voice2Email
  - FAX2Email
  - SMS2Email
- Integrated Hotel Services:
  - On the phone and P.O.
  - Web Hotel in browser
  - External PMS integration.

VoIP AND ToIP SERVICES

- SIP gateway
  - analog / digital terminals
  - analog / ISDN lines (BRI/PRI)
  - QSIG networking
  - S0 Data
- H.323 gateway (ITU-T H.225.0)
- SIP networking with SIP-T protocol (RFC3204)
- Fax T38 (ITU-T)
- Standard SIP terminal support:
  - hardphone
  - softphone
- Wi-Fi terminal support (IEEE 802.11)
- Support hybrid SIP terminals
- Wi-Fi - 2/3 g (IEEE 802.11 /ITU-T IMT-2000)
- Standard H.323 terminal support (called base)(U-T)
- Proprietary IP terminals for KTS services:
  - hardphone (NETfon)
- Internet connection to the company LAN with a NETfon telephone:
  - telework
  - peripheral site
- Power-over-Ethernet (PoE)(IEEE 802.3af).

LEGACY PBX SERVICES

- User termination management:
  - analogue (CEI 103, TBR21)
  - proprietary digital (SAEfon CL family) (CEI 103)
  - ISDN on S0 interface (ETSI TBR-3)
- Operator console (CEI 103):
  - supervisor Console
  - desktop Attendant console (Microsoft TAPI 2.2)
  - desktop Attendant console with accessibility options for blind or low-vision users
- Management of urban line terminations:
  - analogue (ETSI TBR-21)
  - ISDN PRA (ETSI TBR-4, 300 102, 300 125)
  - ISDN (BRA) (ETSI TBR-3, 300 102, 300 125)
- Management of private network terminations:
  - analogue E&M
  - digital QSig (2Mbit/s) (ETS 300 011/ 170/ 171/ 172/ 173/ 237/ 238/ 260/ 261)
- Legacy Class 5 Caller ID telephone services (ETS 300 778-1)
- ISDN supplementary services:
  - malicious call subscriber identification (ETS 300 128 / 129/130)
  - calling / connected line identification (ETS 300 089/ 091/ 092 / ETS 300 094/ 096/ 097)
  - Calling / Connected Line Identification Restriction (ETS 300 090/ 091/ 093/ 095/ 096/ 098)
  - hold and retrive (ETS 300196-1)
  - three-party conference (ETS 300 188-1)
  - taxation according to the “SIP pilot” and ETSI-DSS1 standards
  - remote reading of costs
- QSIG services:
  - basic call (ISO / IEC 11572)
  - name identification (ISO IEC 13868)
  - diversion (ISO / IEC 13673)
  - call transfer (ISO / IEC13869)
  - call offer (ISO / IEC 14843)
  - call completion on busy subscriber (ISO / IEC13870)
  - call completion on no reply (ISO / IEC 13870)
  - path replacement (ISO/ IEC 13874)
  - advice of charge (ISO/IEC DIS15050)
  - message waiting indication.

TELEPHONE SERVICES FOR SIP USERS

- Calling Line Identification Presentation/Restriction (CLIP/CLIR)
- Connected Line Identification Presentation/Restriction (COLP/COLR)
- Call holding and retrieval (RFC 3261)
- Call forwarding (RFC 3261)
- Call transfer (RFC 3515/3891)
- Conference (DRAFT 00)
- Instant messaging (partial RFC 3428)
- Video call (RFC2327)
- Multilinearity
- DTMF transmission / reception (RFC3261 / 2833).

UTILITY FOR SIP USERS

- Registration controlled by management system
- Hierarchical management of SIP users through service classes
- Controlled access to system resources:
  - outband output
  - abbreviated numbering
  - prohibited numbers
- Easy access to “trusted-host” SIP remote users
- Voice mail
- Group management in “SIP forking” mode (RFC 3261)
- Call forwarding / Call transfer.

MANAGEMENT AND PROVISIONING

- IP Management:
  - online and offline configuration
  - FW upload of central units and peripherals via web
  - uploading welcome messages via web
  - configuration release management facility
  - online and offline maintenance
  - online alarm collection
  - accounting and CDR
- Web Personal Provisioning
- Multi-sites Management: alarm accounting
- SNMP-TRAP Service Support (RFC 1157).

SAMIP® systems can perform advanced gateway functions in combination with the BRAVO solution, capable of enabling Virtual-PBX and Unified Communications scenarios, as well as Smart Working applications

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DigitalPlatforms

SAMIP® Communication Server

SAMoffice 4

The SAMoffice 4, part of SAMIP® family, is a versatile communication server tailored for small to medium-sized enterprises. Its key features include:

- **Scalability:** Supports up to 128 line/trunk ports or VoIP channels, accommodating organizational growth.
- **Flexible Connectivity:** Compatible with traditional telephone lines (ISDN, PSTN), traditional extentions (analogue, E&M) and VoIP lines from SIP carriers, offering versatile connection options.
- **Integrated Services:** Provides Enhanced Call Routing, IP LAN Telephony, personal mobility, and multimedia contact center applications*.
- **Comprehensive Telephony Services:** Offers a wide range of services, including Voice Mail, Automatic Operator, Unified Messaging, Conference Server, and fixed-mobile convergence*, addressing diverse communication needs.
- **Simplified Management:** Administrators can manage the system via web interfaces, streamlining configuration and maintenance tasks.
- **Secure Communications:** Ensures secure communications via TLS and SRTP protocols, safeguarding data integrity and confidentiality.
- **Compact Design:** Designed for mounting in a 19'' data rack, occupying 2U per subrack, optimizing space utilization in server rooms.

The SAMoffice 4 is engineered to meet the communication needs of small businesses, providing a robust and scalable solution that supports both traditional and modern telephony services.

*with BRAVO integration.

SAMoffice 4 is the integrated solution of the SAMIP® family developed to meet the communication needs of medium and large companies.

The system allows widespread use of all the most advanced telephone functions available on the market today and is designed to guarantee a high-performance communication service for any type of telephone user, both IP and traditional.

The flexibility and modularity of the elements make **SAMoffice 4** suitable for companies that want to integrate the data network with the voice network and that pay great attention to cost optimization and the improvement of internal processes.

SAMoffice 4 is a communication server that offers each user the same service independently of the device and applications used by the user. The integration of SAMoffice 4 with vertical applications completes the system by linking it to the activities, processes and specific needs of the company and the market.

Main Features

SAMoffice 4 has been designed to connect to any type of traditional telephone line (ISDN or PSTN) as well as to VoIP trunk lines provided by ISPs and SIP carriers.

The modularity of the system allows you to configure **SAMoffice 4** according to the business needs of the moment while ensuring the ability to respond to the future operational needs of the organization.

VoIP resources (up to 192 simultaneous communication channels) can be used for networking in multi-site scenarios and to use all the services available on IP terminals. The integrated SIP Proxy makes it possible to connect SIP standard telephones and clients installed on PCs, smartphones or tablets, which can also be reached via the Wi-Fi network.

The large number of features offered by **SAMoffice 4,** such as Voice Mail, Automated Attendant, Conference Server service, Attendant console with accessibility options for blind or low-vision users, TAPI-based services, etc. allow you to meet all the communication needs of a modern organization.

SAMIP® systems can perform advanced gateway functions in combination with the BRAVO® solution, capable of enabling Virtual-PBX and Unified Communication scenarios, as well as Hybrid Working applications.

	BASE	MAX
IP USERS	16	128
TDM USERS	16	160
AVAILABLE SLOTS	4	8
EXPANSION MODULES	-	1

* Maximum equipment not reachable at the same time

SYSTEM CAPABILITY

External lines
- up to 160 BRI TDM Ports
- up to 16 BRI T0/S0 ISDN Ports
- up to 32 FX0 Ports (analog trunk lines)
- up to 128 FXS Ports (analog extensions)
- up to 2 PRI E1/T1 ISDN Ports.

Telephone extensions
- up to 128 IP 51xx Series extensions
- up to 128 IP Netfon Bluelight extensions
- up to 128 IP SIP extensions
- up to 128 analog extensions.

DSP resources
- up to 128 VoIP channels with DSP Farm units.

SYSTEM DATA

BHCC
- up to 30.000 Busy Hour Call Completion.

Central Unit (redundant)
- Intel Xscale main processor, matrix
- 4096 channel TDM switching and 32 PCM wires, ARM auxiliary processor.

Operating system
- Linux.

TECHNICAL SPECIFICATIONS

Dimensions and weight
- subrack (mm): H.90 x W.490 x D.410; 7 Kg.

Mounting
- 19'' data rack (2U for each subrack).

Connectors
- connection to the LAN network: 10/100 BASE-T Ethernet port switch, IEEE 802.3 / 802.3u, AutoSense, Automatic cable crossover (MDI / DI-X), PoE 802.3af.
- AUX port for inputs and outputs to the field (2 DIN, relay output)
- COM-RS232 port
- 2 LVDS interfaces for Extended (expandable up to 8).

Power Supply
- integrated AC / DC power supply.
- AC IN: 90-264 Vac 50/60 Hz 2.4 / 0.8A.
- integrated batteries (optional).
- DC IN (opt.): Input of power supply voltage -48Vdc (-38Vdc ÷ -60Vdc), max current for single subrack 4.5A.

Consumption

Estimated data for the base system:
- SAMoffice 4 Full IP configuration: 40 W.
- SAMoffice 4 TDM configuration: 63 W.

Peak traffic consumption: 110 W

CERTIFICATIONS
- CE marking (European Community directive 1999/05 /EC: ETSI EN 300 386; EN 55022; EN 55024; EN 61000-3-2; EN 61000-3-3; EN 60950)
- interoperability on TIM's Hyperway network (network of SAMIP exchanges)
- PSTN interface (ETSI TBR-21)
- ISDN-PRA public network interface (ETSI TBR-4; ETS 300102; ETS 300125)
- ISDN-BRA Public Network Interface (ETSI TBR-3; ETS 300102; ETS 300125)
- Caller ID (ETS 300 778-1)
- Environmental tests (ETS EN 300 019 class 3.2 in use) performed in the internal laboratory.

VoIP FEATURES
- Signalling
  - SIP (RFC 3261)
  - H.323 (ITU-T)
- Voice encodings
  - g.711 (ITU-T)
  - g.729a/b (ITU-T)
  - g.723 (6.3) (ITU-T)
  - g.723 (5.3) (ITU-T)
- Network Echo Canceller (ITU-T G.168)
- Voice Activity Detection (VAD)
- Comfort Noise Generation (CNG)
- Jitter Buffer configurable
- QoS via DiffServ (RFC 2474/2597/2598)
- QoS via TOS (RFC791)
- H.323 gatekeeper support (ITU-T)
- Trunks lines backup
- Music on hold.

APPLICATIONS AND CTI
- TAPI 1st party (Microsoft TAPI2.2)
- TAPI 3rd party (Microsoft TAPI2.2)
- Personal Telephony CTI
- Supervisor Telephony CTI
- Contact Center:
  - Multichannel ACD
  - IVR
  - SMS services
  - WEB services
- Click to chat:
  - Click to talk (VoIP)
  - Call me back
  - Co-browsing
- CRM-oriented
- Metrics reports
- Call Flow Editor:
  - Integrated VM/AA
  - Facility UM
- Unified Messaging:
  - Voice2Email
  - FAX2Email
  - SMS2Email
- Integrated Hotel Services:
  - On the phone and P.O.
  - Web Hotel in browser
  - External PMS integration.

VoIP AND ToIP SERVICES
- SIP gateway
  - analog / digital terminals
  - analog / ISDN lines (BRI / PRI)
  - QSIG network
  - S0 data
- H.323 gateway (ITU-T H.225.0)
- SIP network with protocol (RFC 3204)
- Fax T38 (ITU-T)
- Standard SIP terminal support
  - hardphone
  - softphone
- Support standard SIP Wi-Fi terminals (IEEE 802.11)
- Support hybrid SIP terminals Wi-Fi - 2/3 g (IEEE 802.11 / ITU-T IMT-2000)
- Standard H.323 terminal support
- Proprietary IP terminals for KTS services:
  - hardphone (NETfon)
- Internet connection to the company LAN with a NETfon phone:
  - telework
  - peripheral site
- Power-over- Ethernet (PoE) (IEEE 802.3af).

LEGACY PBX SERVICES
- User termination management:
  - analog (CEI 103, TBR 21)
  - proprietary digital (SAEfon CL family) (CEI 103)
  - ISDN on S0 interface (ETSI TBR-3)
- Operator console (CEI 103):
  - supervisor Console
  - desktop Attendant console (Microsoft TAPI 2.2)
  - desktop Attendant console with Accessibility options for blind or low-vision users
- Management of urban line terminations:
  - analog (ETSI TBR-21)
  - ISDN PRA (ETSI TBR-4, 300 102, 300 125)
  - ISDN (BRA) (ETSI TBR-3, 300 102, 300 125)
- Management of private network terminations:
  - analog E&M
  - digital QSig (2Mbit/s) (ETS 300 011/ 170/ 171/ 172/ 173/ 237/ 238/ 260/ 261)
  - Legacy Class 5 Caller ID telephone services (ETS 300 778-1)
- ISDN supplementary services:
  - malicious call identification (ETS 300 128 / 129/130)
  - calling / Connected Line Identification (ETS 300 089/ 091/ 092 / ETS 300 094/ 096/ 097)
  - calling / Connected Line Identification Restriction (ETS 300 090/ 091/ 093/ 095/ 096/ 098)
  - hold and retrive (ETS 300196-1)
  - three-party conference (ETS 300 188-1)
  - taxation according to the "SIP pilot" and ETSI-DSS1 standards
  - remote reading of costs
- QSIG Services:
  - basic call (ISO / IEC 11572)
  - name identification (ISO IEC 13868)
  - diversion (ISO / IEC 13673)
  - call transfer (ISO / IEC13869)
  - call offer (ISO / IEC 14843)
  - call completion on busy subscriber (ISO/ IEC13870)
  - call completion on no reply (ISO / IEC 13870)
  - path replacement (ISO / IEC 13874)
  - advice of charge (ISO / IEC DIS15050)
  - message waiting indication.

TELEPHONE SERVICES FOR SIP USERS
- Calling Line Identification Presentation/Restriction (CLIP/CLIR)
- Connected Line Identification Presentation/Restriction (COLP / COLR)
- Call holding and retrieval (RFC 3261)
- Call forwarding (RFC 3261)
- Call transfer (RFC 3515/3891)
- Conference (DRAFT 00)
- Instant messaging (RFC 3428 partial)
- Video call (RFC2327)
- Multilinearity
- DTMF transmission / reception (RFC3261 / 2833).

UTILITY FOR SIP USERS
- Registration controlled by management system
- Hierarchical management of SIP users through service classes
- Controlled access to system resources:
  - outband output
  - abbreviated numbering
  - prohibited numbers
- Easy access to "trusted-host" SIP remote users
- Voice mail
- Group management in "SIP forking" mode (RFC 3261)
- Call forwarding
- Call transfer.

MANAGEMENT AND PROVISIONING
- IP Management:
  - online and offline configuration
  - FW loading of units
  - control panels and peripherals via the web
  - uploading welcome messages via web
  - configuration release management facility
  - online and offline maintenance
  - online alarm collection
  - accounting and CDR
- Web Personal Provisioning
- Multi-site Management:
  - alarm
  - accounting
- SNMP-TRAP Service Support (RFC 1157).

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DigitalPlatforms

SAMIP® Communication Server

SAMoffice 8

The SAMoffice 8 is an integrated communication solution from the SAMIP® family, designed to meet the needs of medium to large enterprises requiring advanced telephony services. Key features include:

- **Scalability:** Supports up to 640 line/trunk ports, accommodating organizational growth.
- **VoIP Integration:** Enables up to 192 simultaneous VoIP channels, facilitating seamless networking across multiple sites and reducing communication costs.
- **Flexible Connectivity:** Compatible with traditional telephone lines (ISDN, PSTN), traditional extentions (analogue, E&M) and VoIP lines from SIP carriers, offering versatile connection options.
- **Integrated Services:** Provides Enhanced Call Routing, IP LAN Telephony, personal mobility, and multimedia contact center applications*.
- **Comprehensive Telephony Services:** Offers a wide range of services, including Voice Mail, Automatic Operator, Unified Messaging, Conference Server, and fixed-mobile convergence*, addressing diverse communication needs.
- **Simplified Management:** Administrators can manage the system via web interfaces, streamlining configuration and maintenance tasks.
- **Secure Communications:** Ensures secure communications via TLS and SRTP protocols, safeguarding data integrity and confidentiality.

The SAMoffice 8's modular design and extensive feature set make it a robust solution for enterprises seeking a reliable and adaptable communication platform.

*with BRAVO integration.

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The system allows widespread use of all the most advanced telephone functions available on the market today and is designed to guarantee a high-performance communication service for any type of telephone user, both IP and traditional.

The flexibility and modularity of the elements make **SAMoffice 8** suitable for companies that want to integrate the data network with the voice network and that pay great attention to cost optimization and the improvement of internal processes.

SAMoffice 8 is a communication server that offers each user the same service independently of the device and applications used by the user.

The integration of **SAMoffice 8** with vertical applications completes the system by linking it to the activities, processes and specific needs of the company and the market.

Main Features

SAMoffice 8 has been designed to connect to any type of traditional telephone line (ISDN or PSTN) as well as to VoIP trunk lines provided by ISPs and SIP carriers.

The modularity of the system allows you to configure **SAMoffice 8** according to the business needs of the moment while ensuring the ability to respond to the future operational needs of the organization.

VoIP resources (up to 192 simultaneous communication channels) can be used for networking in multi-site scenarios and to use all the services available on IP terminals. The integrated SIP Proxy makes it possible to connect SIP standard telephones and clients installed on PCs, smartphones or tablets, which can also be reached via the Wi-Fi network.

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SAMIP® systems can perform advanced gateway functions in combination with the BRAVO® solution, capable of enabling Virtual-PBX and Unified Communication scenarios, as well as Hybrid Working applications.

	BASE	MAX
IP USERS	16	384
TDM USERS	16	496
AVAILABLE SLOTS	8	31
EXPANSION MODULES	-	3

* Maximum equipment not reachable at the same time

SYSTEM CAPABILITY

External lines

- up to 32 BRI T0/S0 ISDN Ports
- up to 64 FX0 Ports (analog trunk lines)
- up to 496 FXS Ports (analog extensions)
- up to 4 PRI E1/T1 ISDN Ports

Telephone extensions

- up to 384 IP 51xx Series extensions
- up to 384 IP Netfon Bluelight extensions
- up to 384 IP SIP extensions
- up to 496 analog extensions

DSP resources

- up to 192 VoIP channels with DSP Farm units.

SYSTEM DATA

BHCC

- up to 30.000 Busy Hour Call Completion

Central Unit (redundant)

- Intel Xscale main processor, matrix
- 4096 channel TDM switching and 32 PCM wires, ARM auxiliary processor

Operating system

- Linux.

TECHNICAL SPECIFICATIONS

Dimensions and weight

- subrack (mm): H.135 x W.490 x D.410; 7 Kg

Mounting

- 19'' data rack (3U for each subrack)

Connectors

- connection to the LAN network: 10/100 BASE-T Ethernet port switch, IEEE 802.3 / 802.3u, AutoSense, Automatic cable crossover (MDI / DI-X), PoE 802.3af
- AUX port for inputs and outputs to the field (2 DIN, relay output)
- COM-RS232 port
- 2 LVDS interfaces for Extended (expandable up to 8)

Power Supply

- integrated AC / DC power supply
- AC IN: 90-264 Vac 50/60 Hz 2.4 / 0.8A
- integrated batteries (optional)
- DC IN (opt.): Input of power supply voltage -48Vdc (-38Vdc ÷ -60Vdc), max current for single subrack 4.5A

Consumption

Estimated data for the base system:

- SAMoffice8 Full IP configuration: 40 W
- SAMoffice8 TDM configuration: 86 W

Peak traffic consumption: 165 W.

CERTIFICATIONS

- CE marking (European Community directive 1999/05 / EC: ETSI EN 300 386; EN 55022; EN 55024; EN 61000-3-2; EN 61000-3-3; EN 60950)
- Interoperability on TIM's Hyperway network (network of SAMIP exchanges)
- PSTN interface (ETSI TBR-21)
- ISDN-PRA public network interface (ETSI TBR-4; ETS 300102; ETS 300125)
- ISDN-BRA Public Network Interface (ETSI TBR-3; ETS 300102; ETS 300125)
- Caller ID (ETS 300 778-1)
- Environmental tests (ETS EN 300 019 class 3.2 in use) performed in the internal laboratory.

VoIP FEATURES

- Signalling
  - SIP (RFC 3261)
  - H.323 (ITU-T)
- Voice encodings
  - g.711 (ITU-T)
  - g.729a/b (ITU-T)
  - g.723 (6.3) (ITU-T)
  - g.723 (5.3) (ITU-T)
- Network Echo Canceller (ITU-T G.168)
- Voice Activity Detection (VAD)
- Comfort Noise Generation (CNG)
- Jitter Buffer configurable
- QoS via DiffServ (RFC 2474/2597/2598)
- QoS via TOS (RFC791)
- H.323 gatekeeper support (ITU-T)
- Trunks lines backup
- Music on hold.

APPLICATIONS AND CTI

- TAPI 1st party (Microsoft TAPI2.2)
- TAPI 3rd party (Microsoft TAPI2.2)
- Personal Telephony CTI
- Supervisor Telephony CTI
- Contact Center:
  - Multichannel ACD
  - IVR
  - SMS services
  - WEB services
- Click to chat:
  - Click to talk (VoIP)
  - Call me back
  - Co-browsing
- CRM-oriented
- Metrics reports
  - Call Flow Editor:
  - Integrated VM/AA
  - Facility UM
- Unified Messaging:
  - Voice2Email
  - FAX2Email
  - SMS2Email
- Integrated Hotel Services:
  - On the phone and P.O.
  - Web Hotel in browser
  - External PMS integration.

VoIP AND ToIP SERVICES

- SIP gateway
  - analog / digital terminals
  - analog / ISDN lines (BRI / PRI)
  - QSIG network
  - S0 data
- H.323 gateway (ITU-T H.225.0)
- SIP network with protocol (RFC 3204)
- Fax T38 (ITU-T)
- Standard SIP terminal support
  - hardphone
  - softphone
- Support standard SIP Wi-Fi terminals (IEEE 802.11)
- Support hybrid SIP terminals Wi-Fi - 2/3 g (IEEE 802.11 / ITU-T IMT-2000)
- Standard H.323 terminal support
- Proprietary IP terminals for KTS services:
  - hardphone (NETfon)
- Internet connection to the company LAN with a NETfon phone:
  - telework
  - peripheral site
- Power-over- Ethernet (PoE) (IEEE 802.3af).

LEGACY PBX SERVICES

- User termination management:
  - analog (CEI 103, TBR 21)
  - proprietary digital (SAEfon CL family) (CEI 103)
  - ISDN on S0 interface (ETSI TBR-3)
- Operator console (CEI 103):
  - supervisor Console
  - desktop Attendant console (Microsoft TAPI 2.2)
  - desktop Attendant console with Accessibility options for blind or low-vision users
- Management of urban line terminations:
  - analog (ETSI TBR-21)
  - ISDN PRA (ETSI TBR-4, 300 102, 300 125)
  - ISDN (BRA) (ETSI TBR-3, 300 102, 300 125)
- Management of private network terminations:
  - analog E&M
  - digital QSig (2Mbit/s) (ETS 300 011/ 170/ 171/ 172/ 173/ 237/ 238/ 260/ 261)
  - Legacy Class 5 Caller ID telephone services (ETS 300 778-1)
- ISDN supplementary services:
  - malicious call identification (ETS 300 128 / 129/130)
  - calling / Connected Line Identification (ETS 300 089/ 091/ 092 / ETS 300 094/ 096/ 097)
  - calling / Connected Line Identification Restriction (ETS 300 090/ 091/ 093/ 095/ 096/ 098)
  - hold and retrive (ETS 300196-1)
  - three-party conference (ETS 300 188-1)
  - taxation according to the "SIP pilot" and ETSI-DSS1 standards
  - remote reading of costs
- QSIG Services:
  - basic call (ISO / IEC 11572)
  - name identification (ISO IEC 13868)
  - diversion (ISO / IEC 13673)
  - call transfer (ISO / IEC13869)
  - call offer (ISO / IEC 14843)
  - call completion on busy subscriber (ISO/ IEC13870)
  - call completion on no reply (ISO / IEC 13870)
  - path replacement (ISO / IEC 13874)
  - advice of charge (ISO / IEC DIS15050)
  - message waiting indication.

TELEPHONE SERVICES FOR SIP USERS

- Calling Line Identification Presentation/Restriction (CLIP/CLIR)
- Connected Line Identification Presentation/Restriction (COLP / COLR)
- Call holding and retrieval (RFC 3261)
- Call forwarding (RFC 3261)
- Call transfer (RFC 3515/3891)
- Conference (DRAFT 00)
- Instant messaging (RFC 3428 partial)
- Video call (RFC2327)
- Multilinearity
- DTMF transmission / reception (RFC3261 / 2833.

UTILITY FOR SIP USERS

- Registration controlled by management system
- Hierarchical management of SIP users through service classes
- Controlled access to system resources:
  - outband output
  - abbreviated numbering
  - prohibited numbers
- Easy access to "trusted-host" SIP remote users
- Voice mail
- Group management in "SIP forking" mode (RFC 3261)
- Call forwarding
- Call transfer.

MANAGEMENT AND PROVISIONING

- IP Management:
  - online and offline configuration
  - FW loading of units
  - control panels and peripherals via the web
  - uploading welcome messages via web
  - configuration release management facility
  - online and offline maintenance
  - online alarm collection
  - accounting and CDR
- Web Personal Provisioning
- Multi-site Management:
  - alarm
  - accounting
- SNMP-TRAP Service Support (RFC 1157).

SAMIP® systems can perform advanced gateway functions in combination with the BRAVO solution, capable of enabling Virtual-PBX and Unified Communications scenarios, as well as Smart Working applications.

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