SAMoffice 2C

SAMIP® Communication Server

SAMoffice 2C is the compact communication system of the SAMIP family able to satisfy any needs of companies and offices with up to 96 users. **SAMoffice 2C** is designed to meet the needs of companies aware of technological innovation, such as VoIP trunk lines, fixed-mobile convergence, solutions with CTI integration between phone and PC and with the aim to speed up business processes (administrative, commercial and/or production). **SAMoffice 2C** is the ideal companion for any company to communicate with maximum productivity, efficiency and flexibility.

MAIN FEATURES

SAMoffice 2C can be used in Full-IP or hybrid IP-TDM mode to respond to specific needs, present and / or future, of each type of company. **SAMoffice 2C** has been designed to connect natively to any type of traditional telephone line (ISDN or PSTN) as well as to VoIP trunk lines provided by ISPs and SIP carriers. The integrated VoIP resources can be used to connect Selta IP terminals of the NETfon Bluelight and 51.xx series or for SIP standard telephones or SIP clients installed on fixed PCs, notebooks, smartphones or tablets connected via WI-FI network. The ability of **SAMoffice 2C** systems to connect to any network or telecommunication equipment both IP and TDM, even in gateway mode, as well as the availability of a large number of internal interfaces, make **SAMoffice 2C** the ideal companion for communicating with any instrument. Each **SAMoffice 2C** system is supplied equipped with:

- VoIP resources active for the use of IP terminals or VoIP trunk lines
- TAPI 1st Party license, for integration with MS Outlook address books and / or the use of external applications on a PC
- Personal Mobility licenses
- Inbound Routing licenses to route incoming calls
- Click-2-dial licenses to call from the directory via web browser
- Integrated responder (4 synchronized announcements up to 32 seconds each)

The Integrated Messaging feature makes the 2-port Voice Mail / Automated Attendant Station available for 32 voicemail boxes and 30 minutes of basic recording. Through the P.O. it is then possible to automatically recognize incoming faxes and make external calls transit on the local lines attested on **SAMoffice 2C**. Thanks to the "Voice-2-Mail Gateway" function, Voice Mail users can receive the recorded message directly in their mailbox. SAMIP® systems can perform advanced gateway functions in combination with the **SELTA BRAVO®** solution, capable of enabling Virtual-PBX and Unified Communication scenarios, as well as Hybrid Working applications.

MAIN FUNCTIONS

VoIP
Allows immediate use of dedicated resources for networking and IP telephony, or for cost containment thanks to the connection with SIP Carriers.

Reliable
Entirely designed and built in Italy according to the most advanced industrial standards.

Complete
It has a wide range of telephone services for any user, fixed or mobile.

Multi-sites
Enables geographic distribution of corporate communications via Q.Sig IP or TDM links with branch offices, agents, mobile personnel, and so on.

Open
The system seamlessly integrates with business applications and IT infrastructure to maximize the return on your technology investment.

Simplified management
Administrators can manage the system via web or through a dedicated client.

SYSTEM CAPACITY

External lines
- up to 16 BRI T0/S0 ISDN Ports
- up to 32 FX0 Ports (analog trunk lines)
- up to 64 FXS Ports (analog extensions)
- up to 2 PRI E1/T1 ISDN Ports

Telephone extensions
- up to 96 SELTA IP 51xx Series extensions
- up to 96 IP NETfon Bluelight extensions
- up to 96 IP SIP extensions
- up to 56 analogue extensions

DSP resources
- up to 72 VoIP channels with DSP Farm units

TECHNICAL SPECIFICATIONS

Dimensions and weight
Subrack (mm): H.45 x L.440 x P.400; 5 Kg
Mounting
19 " data rack (1U) or on wall

Connectors
- LAN network connection: 10/100 Base T Ethernet port switch, IEEE 802.3 / 802.3u, AutoSense, Automatic cable crossover (MDI / DIX), PoE 802.3af
- AUX port for inputs and outputs to the field (2 DIN, relay output) for connection alarms, buttons, repeaters call
- COM-RS232 RJ45 port for documentation charged
- COM-RS232 CONSOLE RJ45 port

Power Supply
- External AC / DC power supply
- AC INPUT: 100-250 Vca 47/63 Hz
- AC OUTPUT: - 48V DC: 1A

Consumption
Estimated data for the base system:
- SAMoffice2C max .: 60 W

SYSTEM DATA

BHCC
Up to 30.000 Busy Hour Call Completion
Central unit
Intel Xscale main processor, matrix 4096 channel TDM switching and 16 PCM wires, ARM auxiliary processor
Operating system
Linux

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 an internal laboratory

APPLICATIONS AND CTI

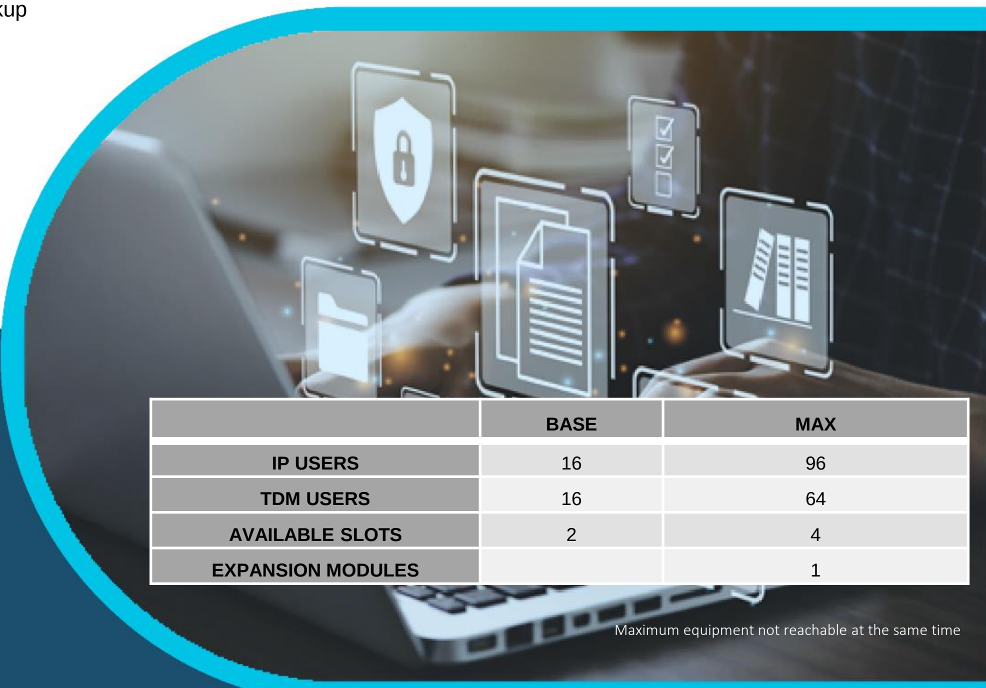
- TAPI 1st party (Microsoft TAPI 2.2)
- TAPI 3rd party (Microsoft TAPI 2.2)
- Personal Telephony CTI
- Supervisor Telephony CTI
- Contact Center:
  - multichannel ACD
  - IVR
  - SMS services
  - WEB services
- Click to chat:
  - click to talk (VoIP)
  - call me back
  - co-browsing
- CRM oriented
- Metrics reports
- Call Flow Editor:
  - Integrated VM/AA
  - facility UM
- Unified Messaging:
  - voice2Email
  - FAX2Email
  - SMS2Email
- Integrated Hotel Services:
  - on phone and P.O.
  - web Hotel in browser
  - external PMS integration

VoIP FEATURES

- Signalling:
  - SIP (RFC 3261)
  - H.323 (ITU-T)
- Voice codes:
  - 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

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 protocol SIP-T (RFC3204)
- 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 (called base) (ITU-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)



	BASE	MAX
IP USERS	16	96
TDM USERS	16	64
AVAILABLE SLOTS	2	4
EXPANSION MODULES		1

Maximum equipment not reachable at the same time

LEGACY PBX SERVICES

- **User termination management**
  - analogue (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
  - 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)
- Additional ISDN 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" standard ETSI-DSS1
  - Remote meter reading
- 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

PHONE 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)
- Videocall (RFC2327)
- Multilinearity
- DTMF transmission/reception (RFC3261/2833)

UTILITY FOR SIP USERS

- Controlled recording from 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-site Management
  - alarm
  - accounting
- SNMP-TRAP Service Support (RFC 1157)

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

DigitalPlatforms Spa Offices
29010 Cadeo (PC), Italy, Via Emilia 231 - ph. +39 0523.50161 – fax. +39 0523.5016333
64018 Tortoreto (TE), Italy, Via Nazionale km 404,500 - ph. +39 0861.772511 – fax. +39 0861.772555
00155 Roma (RM), Italy, Via Andrea Noale 351 - ph. +39 062291879 – fax. +39 0622709440
www.selta.com
marketing@selta.com

DigitalPlatforms

SAM 4000E

SAMIP® Communication Server

SAM4000E 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. **SAM4000E** 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 **SAM4000E** apposite for "mission critical" applications, particularly in environments sensitive to the high availability requirements of the communication system. **SAM4000E** 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, **SAM4000E** adapts perfectly to multi-site or campus-type scenarios in combination with the other systems of the SAMIP® family. SAM4000E 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 **SAM4000E** with vertical applications completes the system by tying it to the activities, processes and specific needs of the company and the market.

MAIN FEATURES

SAM4000E 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 **SAM4000E** 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 SELTA 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 SAM4000E, 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 SELTA BRAVO® solution, capable of enabling Virtual-PBX and Unified Communication scenarios, as well as Hybrid Working applications.

MAIN FUNCTIONS

VoIP
Allows immediate use of dedicated resources for networking and IP telephony, or for cost containment thanks to the connection with SIP Carriers. **Modular**return on your technology investment.

Simplified management
Administrators
Grows flexibly and scalable according to actual business needs.

Reliable
The system architecture includes the redundancy of the central unit, power supply and connections with the subracks to provide a high level of resiliency in critical situations.

Complete
It has a wide range of telephone services for any user, fixed or mobile.

Multi-sites
Allows geographic distribution of corporate communications via IP or TDM QSIG links with branches, agents, mobile personnel, etc.

Open
The system seamlessly integrates with business applications and IT infrastructure to maximize the can manage the system via web or through a dedicated client.

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 SELTA 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

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

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

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

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 FEATURES

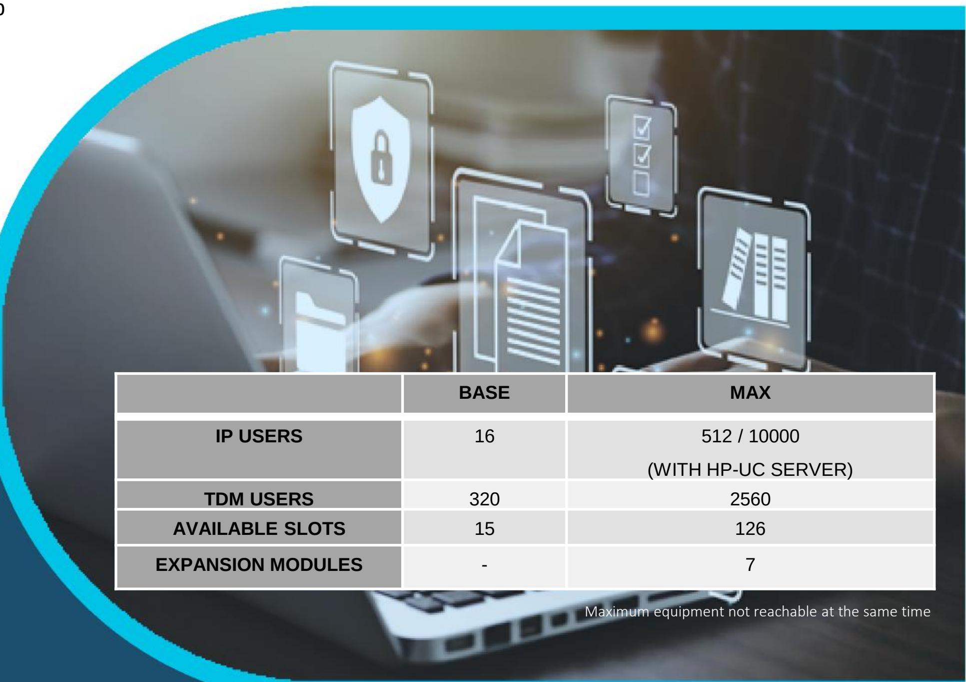
- Signalling
  - SIP (RFC 3261)
  - H.323 (ITU-T)
- Voice encodings
  - g.711 (ITU-T)
  - g.729a/b (ITU-T)
  - g.723 (6.3) (ITU-T)
  - g.723 (5.3) (ITU-T)
- Network Echo Canceller (ITU-T G.168)
- Voice Activity Detection (VAD)
- Comfort Noise Generation (CNG)
- Configurable Jitter Buffer
- QoS via DiffServ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

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) (ITU-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)



	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

LEGACY PBX SERVICES

- **User termination management:**
  - analogue (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:**
  - 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 SELTA BRAVO solution, capable of enabling Virtual-PBX and Unified Communications scenarios, as well as Smart Working applications

DigitalPlatforms Spa Offices
29010 Cadeo (PC), Italy, Via Emilia 231 - ph. +39 0523.50161 – fax. +39 0523.5016333
64018 Tortoreto (TE), Italy, Via Nazionale km 404,500 - ph. +39 0861.772511 – fax. +39 0861.772555
00155 Roma (RM), Italy, Via Andrea Noale 351 - ph. +39 062291879 – fax. +39 0622709440
www.selta.com
marketing@selta.com

DigitalPlatforms

SAMoffice 8

SAMIP® Communication Server

SAMoffice 8 is the integrated solution of the SAMIP® family developed by SELTA 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 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 SELTA 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 8**, 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 SELTA BRAVO® solution, capable of enabling Virtual-PBX and Unified Communication scenarios, as well as Hybrid Working applications.

MAIN FUNCTIONS

VoIP
Allows immediate use of dedicated resources for networking and IP telephony, or for cost containment thanks to the connection with SIP Carriers.

Modular
Grows flexibly based on actual business needs.

Complete
It has a wide range of telephone services for any user, fixed or mobile.
Multi-sites
Allows geographic distribution of corporate communications via IP or TDM QSIG links with branches, agents, mobile personnel, etc.

Open
The system seamlessly integrates with business applications and IT infrastructure to maximize the return on your technology investment.
Simplified management
Administrators can manage the system via web or dedicated client.

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 SELTA 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

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

SYSTEM DATA

- BHCC
  - up to 30.000 Busy Hour Call Completion
- Central unit
  - Intel Xscale main processor, matrix
  - 4096 channel TDM switching and 32 PCM wires, ARM auxiliary processor
- Operating system
  - Linux

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

APPLICATIONS AND CTI

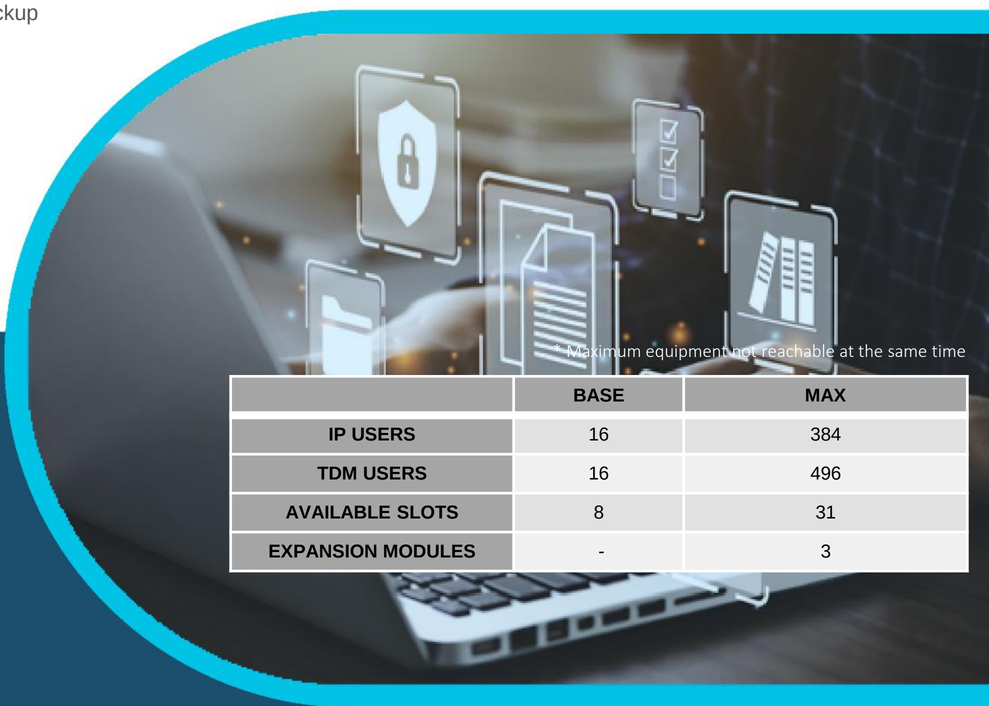
- TAPI 1st party (Microsoft TAPI2.2)
- TAPI 3rd party (Microsoft TAPI2.2)
- Personal Telephony CTI
- Supervisor Telephony CTI
- Contact Center:
  - Multichannel ACD
  - IVR
  - SMS services
  - WEB services
- Click to chat:
  - Click to talk (VoIP)
  - Call me back
  - Co-browsing
- CRM-oriented
- Metrics 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 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

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)



* Maximum equipment not reachable at the same time

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

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 SELTA BRAVO solution, capable of enabling Virtual-PBX and Unified Communications scenarios, as well as Smart Working applications

DigitalPlatforms Spa Offices
29010 Cadeo (PC), Italy, Via Emilia 231 - ph. +39 0523.50161 – fax. +39 0523.5016333
64018 Tortoreto (TE), Italy, Via Nazionale km 404,500 - ph. +39 0861.772511 – fax. +39 0861.772555
00155 Roma (RM), Italy, Via Andrea Noale 351 - ph. +39 062291879 – fax. +39 0622709440
www.selta.com
marketing@selta.com

SAMoffice 4

SAMIP® Communication Server

SAMoffice 4 is the integrated solution of the SAMIP® family developed by SELTA 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 SELTA 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 SELTA BRAVO® solution, capable of enabling Virtual-PBX and Unified Communication scenarios, as well as Hybrid Working applications.

MAIN FUNCTIONS

VoIP
Allows immediate use of dedicated resources for networking and IP telephony, or for cost containment thanks to the connection with SIP Carriers.

Modular
Grows flexibly based on actual business needs.

Complete
It has a wide range of telephone services for any user, fixed or mobile.
Multi-sites
Allows geographic distribution of corporate communications via IP or TDM QSIG links with branches, agents, mobile personnel, etc.

Open
The system seamlessly integrates with business applications and IT infrastructure to maximize the return on your technology investment.
Simplified management
Administrators can manage the system via web or dedicated client.

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 SELTA 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

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

SYSTEM DATA

- BHCC
  - up to 30.000 Busy Hour Call Completion
- Central unit
  - Intel Xscale main processor, matrix
  - 4096 channel TDM switching and 32 PCM wires, ARM auxiliary processor
- Operating system
  - Linux

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

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
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www.selta.com
marketing@selta.com

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