ProTalk SARK UCS Phone systems

COMMUNICATIONS
INNOVATIVE VOIP DISTRIBUTION

ProTalk

ProTalk is the IP Telephone systems division of ProVu Communications, the specialist UK VoIP Distributor for Snom, Cisco, AVM and Siemens Gigaset Cordless IP Phones.

Having carefully and exhaustively evaluated the current software based PBX market, ProVu technical personnel have selected SARK UCS as the most advanced and reliable TDM/VoIP platform currently available in the mid-range market. With the addition of SARK UCS to ProVu's ProTalk range of VoIP systems, resellers now have access to a wide range of high quality, professional VoIP equipment and support services enabling them to offer a comprehensive bundled communications solution.

SARK - UCS Overview

SARK UCS is a new kind of PBX platform, built to cope with medium to high workloads on existing copper and fibre-based TDM networks. It is equally at home in the 21CN world of SIP and VoIP. Designed and developed for the UK market by UK telephony people, the platform is reliable, fast and has a low cost-of-ownership when compared with traditional TDM and proprietary IP offerings. SARK UCS also has many unique features which give it a substantial edge over its competitors, particularly in the areas of high-availability and remote platform support. Issues that are of particular importance to those users whose businesses depend upon high levels of customer interaction and care.

SARK
VOICE DATA AND INTERNET PLATFORM

SARK 650

Small office system - solid state unit

SARK 850

General office system - 1U rack mount server

SARK 1000

Large office / Call Centre system - 1U rack mount server

SARK 1200

Enterprise / Call Centre system - 2 U rack mount server

PBX	Guide to no. of extensions	Guide to no. of concurrent calls	PSTN Interface options	SIP/IAX trunks	Call queues	Ring groups	Conference Brisge	Call Centre Edition	HA (High Availability) Option	Advanced Call Recording
650	6-20	1-8 calls	up to 4 FX(O/S) or 1 to 4 ISDN2E	Unlimited	Unlimited	Unlimited	Unlimited	Not available	Not available	Not available
850	12-50	6-30 calls	up to 8-FX(O/S) or 1 to 4xISDN2E or 1xISDN30E	Unlimited	Unlimited	Unlimited	Unlimited	Available	Available	Available
1000	40-100	12-60 calls	1 or 2 ISDN30E	Unlimited	Unlimited	Unlimited	Unlimited	Available	Available	Available
1200	100-200	30-120 calls	1 to 4 ISDN30E	Unlimited	Unlimited	Unlimited	Unlimited	Available	Available	Available

ProVu Communications Ltd
Savile Mill, Savile Street, Milnsbridge, Huddersfield, HD3 4PG, United Kingdom
Tel: +44 (0)1484 840048 Fax: +44 (0)1484 651742 Web: www.provu.co.uk Email: contact@provu.co.uk

Company Registration No: 3748320 Registered in England and Wales

SARK UCS - Key Advantages

COMMUNICATIONS
INNOVATIVE VOIP DISTRIBUTION

Call Recording for SARK UCS

ProTalk's advanced call recording package has been designed to give customers flexibility and control of what they want to record and when. Within the easily managed web interface customers can specify Call Recording options at the individual phone, call group, queue or trunk level.

As standard, calls are offloaded to locally attached storage, for example a local USB2 disk drive. However a "high-capacity" option is also available to offload recordings to network attached devices. In this mode, recording is done directly into RAM and then asynchronously offloaded to the network device. We call this "lazy" offload because of its asynchronous nature.

Access to stored calls is via a web interface and can be searched by either the user name, extension number or date and time.

Recording Options (can be set at individual phone, call group, queue or system level)

All in-bound
All out-bound
Both
None

Additional advanced features include two "on-demand" options:-

- One Touch press and Record
  The call is recorded from the moment record button is pressed.
- One Touch Retrospective Recording
  The Operator presses the record button at any time and the whole of the call is recorded from the start of the call.

Real-time Pause and Resume for queued calls.

Sometimes called "call-ducking"; the whole call is recorded but the agent can stop recording at any time by pressing the "pause" button and resume recording by pressing the "play" button. The resulting parts are married together as one recording in the final mix. This feature is useful for preventing sensitive customer information such as credit card details from being recorded.

Remote Support / Management

ProTalk systems have been developed with the most advanced remote management, configuration and diagnostic capabilities of any PBX in their class. Tools like the unique SIP Dynamic Proxy capability plus automatic recognition and adoption of new SIP devices, enable effective system and terminal management, and make on-going adds, moves and changes much easier for both customer and supplier

Switching Performance

SARK UCS uses its own on-board high-speed logic and rules engine (the HSLE). The HSLE is very small and fast, being optimised to switch high volumes of in-bound and out-bound calls in the minimum number of cycles.
The end result is much faster switching decisions from SARK when compared to its competitors. Thanks to HSLE, SARK UCS can comfortably sustain high call arrival rates on relatively low CPU power, making it ideal for deployment into high volume or high "spike" environments.

'FlatPack' Turn-Key Solution for Fast and Easy Deployment

With the ProTalk developed configurator documentation, ProTalk resellers can forward order PBX systems with all key provisioning and routing information pre-loaded. This combined with the PBX's auto-provisioning software (available for most popular SIP phone types) means installation and deployment is painless, error free and normally completed in just a few hours.

Reliability – High Availability (HA)

Where high levels of reliability and uptime are of the utmost importance, as for example in call centre applications, the ProTalk SARK advanced High Availability option can be of huge business value.
The HA cluster comprises two PBX servers with identical configurations running side by side and connected by a 'heart beat' mechanism. In the event of a failure of the primary system, fail-over to the secondary takes just a few seconds and requires no reprogramming or restart of either the phones, the VoIP accounts, gateways or ISDN circuits. The total downtime in such situations is usually around 12 to 20 seconds from initial failure to resumption of operations. When compared to the minimum 4 hour callout, or next business day support terms available from our competitors, this one feature alone can make a huge difference to the bottom line in call-critical businesses.

VOICE DATA AND INTERNET PLATFORM

ProVu Communications Ltd
Savile Mill, Savile Street, Milnsbridge, Huddersfield, HD3 4PG, United Kingdom
Tel: +44 (0)1484 840048 Fax: +44 (0)1484 651742 Web: www.provu.co.uk Email: contact@provu.co.uk
Company Registration No: 3748320 Registered in England and Wales

Feature List

ProVu®
COMMUNICATIONS
INNOVATIVE VOIP DISTRIBUTION

Call Handling
- Call hold (with music)
- Blind/Attended Transfer with optional return on no answer
- Call "camp" on extension
- Call re-direct or shunt
- Call pickup groups
- Call parking
- Call Forwarding - (on busy, no answer or unconditional)
- Call waiting (multi line handsets)
- Do not disturb
- IVR/Automated attendant
- Multi-level automated attendants for routing of calls more efficiently (i.e. press 1 for etc...)
- Direct dial to extension
- Interactive directory name/number lookup
- Direct record from handset or upload messages
- Multilevel user access for security

Call queuing
- Call queuing with strict answer ordering
- Choice of call distribution methods including (ring all, round robin, round robin memory, least recently called first, fewest number of calls first or random)
- Static and dynamic agents to allow users to log-in/out of the call queues
- Caller announcements including queue position and estimated hold time.
- Link to IVR for in queue options
- Min. and Max. people in queue settings
- Max. wait time for queued calls
- Choice of hold music

SARK
VOICE DATA AND INTERNET PLATFORM

Voice mail
- Voicemail per extension (no port limits)
- Message retrieval by phone (local or remote) & web browser
- Email notification (with sound file if required)
- Visual and stutter dial tone message waiting indication (handset dependant)
- Group mailboxes
- Busy and Unavailable Messages
- Message forwarding and append options
- Multiple mailboxes for filing / storage

Advanced call routing
- Automatic fail over for outbound calls
- Provider preferencing for least cost routing
- Dialled number manipulation pattern match, add/subtract digits)
- Route password protection (class of service)
- Inbound call routing (either number called, or calling number)
- Direct SIP inbound/outbound calling support or ENUM/SIP URI calling
- Transcoding between codecs for maximum compatibility

Handset features (model dependant)
- Compatible with SIP compliant handsets
- 3 way conferencing
- Busy lamp field support
- Stutter dial tone for message waiting
- Missed/Dialled and Received calls logging
- Remote workers
- Caller ID name & number support
- Multiple codec support
- Personal phone book with distinctive ring
- Global address book linked to LDAP/XML existing database

Call distribution
- Flexible extension numbering
- Ring groups with timers and fail overs for call distribution including off site calling
- DDI inbound routing support
- Route based on Caller ID and/or DDI
- Time/Day/Date routing

Conferencing
- 3 way conferencing (handset dependent)
- Conference rooms for larger conferences
- Pin access
- Member announcements (join/leave)
- Mute/Un-mute per user

Reporting
- Detailed call logging with selection, search and CSV export
- Call comparison over time graphs
- Monthly traffic graphing
- Daily load graphing

Administration/usability
- Easy web based interface for administration local/remote
- Multi user/multi level access for delegated/departmental control
- Web based operator's console

Music on hold
- Multiple music on-hold tracks (mp3)
- Multiple music categories

Call Recording Option
- Selective recording at system, group or individual extension
- One-Touch on-demand recording
- One-Touch Retrospective on-demand recording option
- Real time pause and resume option
- High volume in-RAM record-and-offload option

High Availability Option
- Tandem system
- Fully automatic failover in the event of software or hardware failure
- ISDN BRI/PRI failover option (hardware card)
- Automatic failback when prime node recovers
- Failover in less than 20 seconds (on regular ISDN30 circuits)

System integration
- Tapi compliant (multiple drivers available)
- Inbuilt Manager API for custom integration/control
- Outlook click to call support (via Tapi)

Multi provider/trunk support
- Multiple inbound phone lines/trunks
- Multiple technologies including:
- Analogue, ISDN2e, ISDN30e, IP Trunks (SIP), ENUM (e.164.arpa & .org)

Expandable
- Custom programming language for system expansion and integration
- Media Link—connect multiple managers together across multiple sites providing one large integrated phone system across multiple sites/locations
- Say/Spell Engine (custom programming)
- Software upgradeable

Specifications may be subject to change without notice

ProVu Communications Ltd
Savile Mill, Savile Street, Milnsbridge, Huddersfield, HD3 4PG, United Kingdom
Tel: +44 (0)1484 840048 Fax: +44 (0)1484 651742 Web: www.provu.co.uk Email: contact@provu.co.uk

Company Registration No: 3748320 Registered in England and Wales