



SARK UCS/MVP VoIP/TDM PBX

Administrators Guide V3.1

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Introduction

The SARK UCS/MVP PBX server provides a resilient SIP/TDM PBX solution for system integrators and PBX resellers. It is based upon Digium's Asterisk switching platform running on either

- SME Server Linux
- PIKA Technologies Warp embedded Linux platform
- CentOS/Redhat EL5 Linux

The product has established a reputation for being easy-to-use with a very small footprint while being functionally able to compete with traditional proprietary PBX offerings at a much lower overall cost-of-ownership. There is also a high availability add-on which can give SARK users close to 100% uptime for their PBX (at the cost of deploying a second server).

SARK V3.1.0 is delivered as a series of Redhat RPMs ready to be installed onto an existing SME server 8.x or vanilla EL5; It is also available as a ready to install .iso image or as a complete ready-to-run system on regular Intel hardware or on the Warp PPC embedded platform.

One rather unique feature of SARK is its ability to automatically recognize and provision PnP SIP phones (principally from Snom, Yealink and Gigaset) with no human intervention and no special network setup. This means that, unlike competitive VoIP offerings, which can take several hours or even days to set up, SARK can be deployed into an unknown network (provided only that the network has a functioning DHCP server) with brand new factory set phones and require no initial set-up. The system will come up; provision the phones as they come on-line. This makes the system as easy and quick to install as a traditional TDM PBX, an important consideration when installing into small businesses where cost is often an overriding issue.

In addition, the on board SARK UCS/MVP software provides a feature-rich graphical workbench and decision engine designed to generate and run efficient PBX images. All adds moves and changes can be done remotely as soon as the system is installed.

Features Overview

SARK provides an easy to use graphical workbench and decision engine that controls and manages the PBX.

SARK UCS/MVP features

- Zero touch setup for PnP capable SIP phones
- Fast, automated installation
- Resilient High-Availability (option)
- Multi tenant support
- Add or change extension and voicemail accounts in seconds
- Integrated, extensible automatic provisioning service for all major SIP handsets
- Supports all Asterisk-supported trunk technologies
- Reduce long distance costs with LCR and powerful pattern-based outbound routing
- Route incoming calls based on time-of-day, DID, Caller ID or caller ID pattern
- Supports BLF, call parks and directed call pickup
- Create infinite level interactive Digital Receptionist (IVR) menus
- Design sophisticated recursive call groups
- True hot-desk support (option)
- Advanced on-board voice recording feature (option)
- Manage callers using ACD and Queues
- Detect and direct incoming faxes
- Automatic Backup and one-click Restore
- Powerful on-board firewall
- easy to use network management

SARK hardware platforms

There are 4 models in the SARK range. The range in capability from 8 channels and 20 endpoints up to 120 channels and several hundred end-points per node.

| MODEL | DESCRIPTION | Expansion Slots | MEMORY | Storage |
|--------------|--------------------|------------------------|---------------|----------------|
| 500 | Embedded | 2 | 256Mb | 2Gb Flash |
| 850 | 1U rack mount | 1 | 1Gb | 80Gb SATA |
| 1000 | 1U rack mount | 1 | 1Gb | 80Gb SATA |
| 1200 | 2U rack mount | 3 | 2Gb | 160Gb SATA |

Fore more information on the SARK500 entry level system please see the *SARK500 admin guide*.

All SARK units are capable of running their own inboard telephony gear. Depending on the model, they can be fitted with up to 8xISDN2(BRI) channels or up to 120xISDN30 (PRI) channels (lines) per module. For more channels, the modules can be stacked into clusters.

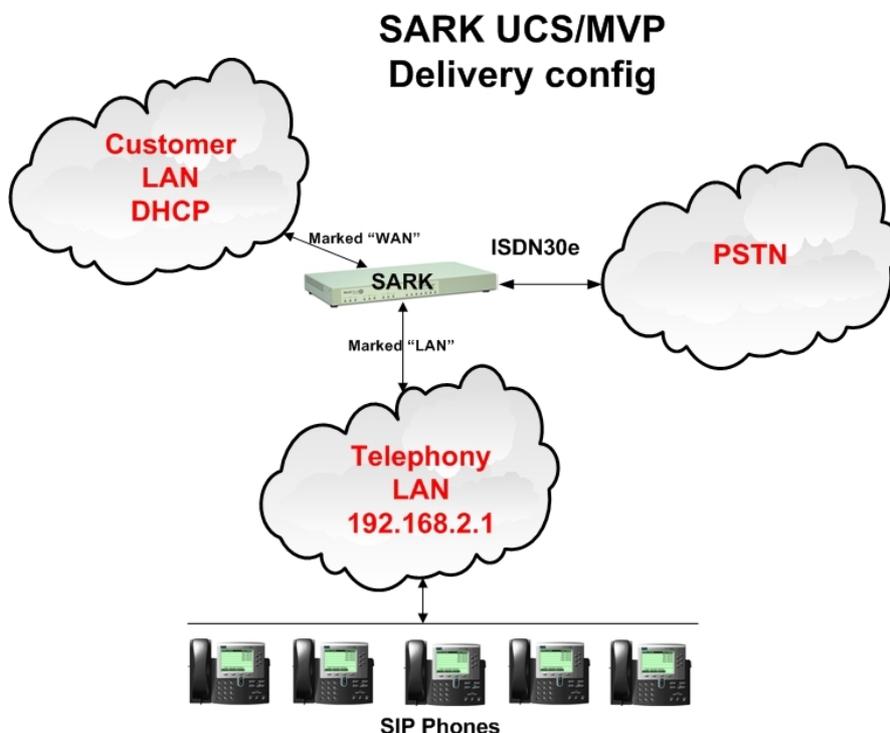
Modes of operation

Depending upon the model, SARK can run in one of two modes; either as a discrete network server as part of an existing network setup or as a server/gateway.

The S500 and S850 (which only have one NIC) always run as discrete network servers. The S1000 and S1200 can be delivered with twin Ethernet circuitry terminating in RJ45 sockets to the rear of the enclosure. These units can run in server/gateway mode. The sockets are marked "WAN" and "LAN". The WAN socket connects upstream to one of the following

- Customer LAN
- cable or ADSL router

The LAN socket connects downstream to the phones. Unless you specify otherwise, single S1000/S1200 systems will be delivered pre-configured to run in Server/Gateway mode and set to request DHCP from the upstream network. The downstream LAN will be set to 192.168.2.0/255.255.255.0 with the SARK unit itself running at 192.168.2.1.



Changing the low-level system defaults

If you wish to change the factory default settings you can attach a screen and keyboard directly to the PBX server and log in as admin (with the Aelintra supplied password). This will run a wizard type application which you can use to change underlying settings such as the system mode, domain information and IP addresses.

Server Only

In server only mode, SARK is attached to an existing network and provides PBX, services to the network. Perimeter protection, internet sharing, caching, port forwarding and other services will be provided from an existing firewall/perimeter protection device or server. To run in this mode, the RJ45 socket marked LAN, should be connected to an available socket on your network LAN hub/switch using a standard RJ45 Cat5 cable. The RJ45 socket marked WLAN should not be connected. You should also forward the following ports to SARK from your firewall

| PORT | NAME | TYPE |
|--------------|-------|------|
| 5060 | SIP | UDP |
| 10000->20000 | RTP | UDP |
| 4569 | IAX2 | UDP |
| 443 | HTTPS | TCP |
| 45678 | SSH | TCP |

Server/Gateway

In server/gateway mode SARK UCS/MVP provides full perimeter protection (firewall) to the downstream network and manages all interactions to and from the upstream network. To run in this mode you should connect the WAN socket to your upstream network or Internet modem (either cable or ADSL). You should connect your LAN socket to an available socket on a network switch using a standard RJ45 Cat5 cable. If the upstream network has its own router then you should forward the following ports to SARK

| PORT | NAME | TYPE |
|--------------|-------|------|
| 5060 | SIP | UDP |
| 10000->20000 | RTP | UDP |
| 4569 | IAX2 | UDP |
| 443 | HTTPS | TCP |
| 45678 | SSH | TCP |

You can run your upstream router either in bridged or “no NAT” mode. In Bridged mode you will need a single public IP Address whereas in “no NAT” mode, you will need at least two public IP Addresses; one for the router and one for SARK.

Equipment Deployment

Unpacking

Your SARK UCS/MVP platform and ancillary components are carefully packed to your order at the factory. Enclosed with your shipment is a bill-of-material detailing each component in the shipment. Before doing anything else, check each component against the bill to ensure that all components are present and none have been damaged. Whilst every reasonable care is taken to ensure that the system reaches you in the same condition it left the factory, it is an unfortunate fact that occasionally a component may be damaged in transit. If you find a damaged item then you should immediately contact your Distributor/Reseller to obtain an rma number which will allow you to order a replacement.

Equipment Placement

Before beginning the installation proper, it is good idea to give some thought as to where you are going to site the system components, in particular, the server itself, the gateways (if any), the network switch and the IP telephones. Generally speaking, the server should be sited in a well ventilated location, away from high traffic areas in the workspace and ideally, above the floor. You should avoid stacking any other items on top of, or around, the casing to avoid overheating. Should the unit overheat as a result of restriction of the vents then this will automatically invalidate your warranty.

WARNING! *NONE of the rack capable units may be mounted solely by the rack ears. You must ensure that there is adequate support at the rear of the unit. This can be done either by ordering the optional rack-mount rail kit from your supplier or by using a suitable rack tray.*

Maximum Cable lengths

- Unless your network switch manufacturer’s instructions state otherwise, the PBX server may be sited up to 100m (328 feet) from the network switch when using CAT5 or CAT5e UTP cabling, or 10m (33 feet) for stranded patch cable.
- When operating in Server/Gateway mode to an upstream router, the PBX server should be no more than 5m (16 feet) from the modem (cable or DSL).

- Analogue Telephone Adapters (ATA's) and SIP/IAX Gateways may be sited up to a maximum of 100m (328 feet) from the network switch when using CAT5 or CAT5e UTP cabling, or 10m (33 feet) for stranded patch cable. They may be up to 10m (33 feet) from any attached analog telephony equipment or carrier termination points (i.e the telephone company network terminators (NT)).
- IP telephones can be sited up to 100m from the network switch when using CAT5 or CAT5e UTP cabling, or 10m for stranded patch cable.
- If you have ordered any digital telephony cards as part of your system (ISDN2e or ISDN30e), they will have been predefined to your system with a line build out of up to 40M (133 feet).

Network Port Assignment

The Sark500 and SARK850 units are fitted with a single RJ45 Ethernet port.

SME 7.x (CentOS 4) units of the Sark650xp (discontinued), SARK850 and SARK1000 each have four (4) RJ45 Ethernet ports assigned as follows (when viewed, left to right, from the rear of the enclosure);

- Port1 - unused
- Port2 - LAN
- Port3 - WAN
- Port4 - unused

SME 8.x (CentOS 5) units Sark650xp SARK850 and SARK1000 each have four (4) RJ45 Ethernet port apertures assigned as follows (when viewed, left to right, from the rear of the enclosure);

- Port1 - LAN
- Port2 - unused
- Port3 - WAN
- Port4 - unused



The SARK1200 has two (2) RJ45 Ethernet ports assigned as follows (when viewed, left to right, from the rear of the enclosure);

Port1 - LAN

Port2 - WAN

ISDN Port Assignment

The S500 and The S850 can each be fitted with up to four (4) ISDN2e ports, while the larger machines can be fitted with up to eight (8) ISDN2e ports or up to four (4) ISDN30e ports. Ports are assigned as follows (when viewed, left to right, from the rear of the enclosure);

ISDN2e (S500) - 1, 2 and 3,4 (2 per daughter-board)

ISDN2e (S850, S1000, S1200) - 1,2,3,4,5,6,7,8

ISDN30e (S850, S1000, S1200) - 1,2,3,4

Notes ISDN QSIG and ISDN DASS

- In UK, only ISDN30e (QSIG) is supported. If you have older ISDN30 or DASS equipment then it will need to be upgraded.
- All SARK ISDN terminations are to RJ45.
- Eight-way (16 channel) ISDN2e cards use Siamese ports. They require made-up or pre-ordered split Cat5e patch cables.

Network Switch

The network switch should have enough ports to handle all of your IP connected telephones plus room for growth. It should be sited away from high traffic areas in the workspace and it should have sufficient front clearance to accommodate the RJ45 connectors from the UTP cables. You should also ensure that there is adequate access to enable easy addition/removal and rerouting of cables from time to time. UTP cables should be uniquely identified and marked at both termination points in order to facilitate easy tracing from device to switch.

If you intend to deploy a POE capable switch then you should ensure that its total power output is adequate for the overall load that the POE phones will impose. This information is available from the phone and POE switch manufacturers.

Power Up

Once you have decided upon a satisfactory location, install the network switch. Everything else will connect to this device so it's the best place to start. If your workplace is already cabled for a computer network you may wish to assess whether you have sufficient access points to handle an integrated computer/telephone network. This may necessitate new cable runs and/or switch upgrades and it is best to do these now before any further equipment is positioned.

Next, install the SARK UCS/MVP server platform. Attach the LAN socket on the rear of the SARK UCS/MVP enclosure to the network switch using a Cat5e (RJ45) UTP or braided patch

cable (depending upon distance). Attach the WAN Socket to your upstream network switch or router. Bring up the network switches and the SARK UCS/MVP server. The server will take about 3 minutes to boot and run through its startup checks. Your SARK UCS/MVP PBX is now ready to be tailored to your requirements. Depending upon the support subscription you have with your supplier this will either be carried out locally by yourself, by your reseller or remotely by your distributor.

Logging into the system

The SARK UCS/PBX GUI manager is browser based. You can connect using any modern browser (I.E., Firefox, MSIE, or Chrome) with the following URLs

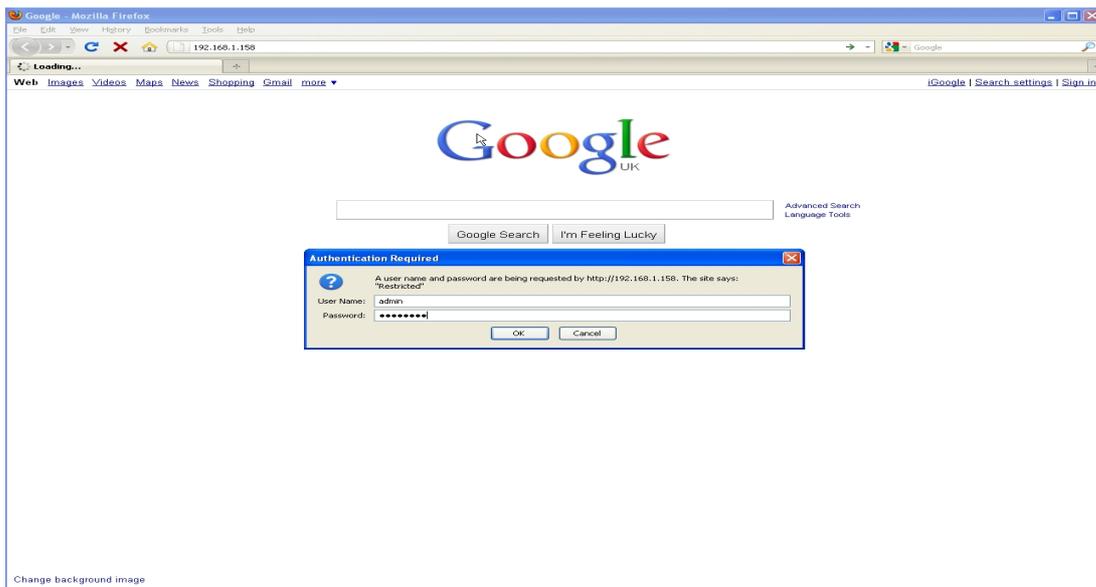
<https://local ip-address/sark>

The SME Server “*server-manager*” component is used to manage the underlying Linux platform. You can connect to it using

<https://local ip-address/server-manager>

The user-id is *admin*. The system password will either be found on a label on the PBX packaging or will be communicated to you by your reseller. Entering the user-id and password will display the SARK UCS/MVP home panel.

To log on to SARK you will need to know the IP address of the appliance. Type the IP address of the appliance in the address bar of any Web browser on the same subnet. The SARK login appears.



Type the user name and password in the appropriate text box. Click the OK button. The SARK ‘globals’ page appears.

TDM/VoIP Softswitch - Globals.

- Home
- Extensions
- Trunks
- Routes
- Call Groups
- IVR Menus
- Custom Apps
- Call Back
- Day/night Timers
- Class of Service
- Agents
- Queues
- Greetings
- IP Devices
- Asterisk File Edit
- Sylogs
- Firewall
- Change Password
- Backup/Regress
- Global Settings

General Services Call-Control Admin

External IP Address

Voicemail Instructions YES

Late Termination NO

Conference Type simple

Extension Length 3

Extension Start Number 401

Agent Start 1001

System Operator(default 0) 0

Call Recording System Default None

Operator Real Extension 400

Sysinfo
Serial #: 568558
Release #: 3.1.0-81
Web Server: lighttpd
Browser: Firefox

Network
hostname: sark
domain: aelintra.com
LAN IP: 192.168.1.140
Netmask: 255.255.255.0

Hardware
System Media: flash
Disk Usage: 88%
RAM Size: 256208
RAM Free: 72680

Status
PBX State: RUNNING



The System Statistics section (to the right) shows general status information.

Using SARK

The SARK browser pages follow a standard layout. The figure below shows a typical SARK Web page.

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Navigation is handled by the column to the left of the screen. Modifiable data appears in the data window in the centre and information and action buttons appear to the right. All data fields have instant help (see below). To save your changes, press the save (disk icon) button. To commit your changes (i.e. bring them into service), click the tick, or checkmark, button; this will cause SARK to regenerate the underlying Asterisk configuration files and issue a soft reload to the PBX.

System Help

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Every name field in the system has context sensitive help associated with it. All you need do is hover the cursor over any name field for an explanation of the field's function.

System Buttons

The system buttons are clustered to the right of each data panel. The available buttons are as follows;



NEW button. Used to create a new object instance



SAVE button. Used to save a change



ACTION button. Used to perform an action.



DELETE button. Used to delete an object instance.



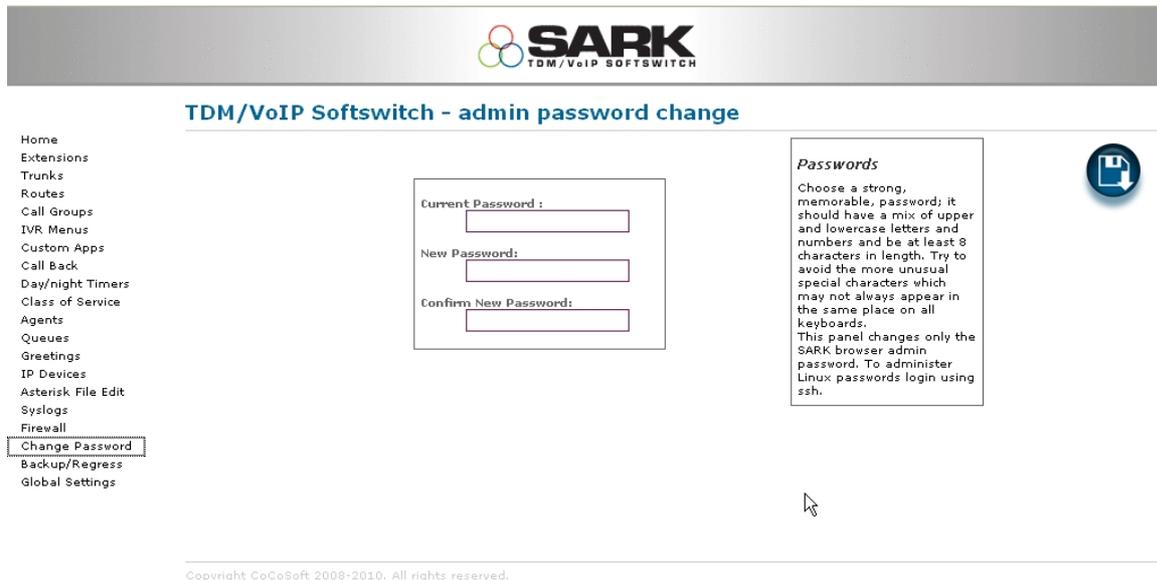
COMMIT button. Used to “enliven” your changes by updating the PBX running instance



CANCEL button. Used to cancel the current operation and return to the parent screen.

Changing the Browser Password

For security reasons, we recommend that you change the default browser password immediately using the change password panel.



The screenshot shows the SARK TDM/VoIP Softswitch admin interface. At the top, there is a header with the SARK logo and the text "TDM/VoIP Softswitch - admin password change". On the left side, there is a navigation menu with the following items: Home, Extensions, Trunks, Routes, Call Groups, IVR Menus, Custom Apps, Call Back, Day/night Timers, Class of Service, Agents, Queues, Greetings, IP Devices, Asterisk File Edit, Syslogs, Firewall, Change Password (highlighted), Backup/Regress, and Global Settings. The main content area is titled "Passwords" and contains three input fields: "Current Password:", "New Password:", and "Confirm New Password:". To the right of the input fields, there is a text box with the following text: "Choose a strong, memorable, password; it should have a mix of upper and lowercase letters and numbers and be at least 8 characters in length. Try to avoid the more unusual special characters which may not always appear in the same place on all keyboards. This panel changes only the SARK browser admin password. To administer Linux passwords login using ssh." There is also a save icon (a floppy disk) in the top right corner of the main content area. At the bottom of the page, there is a copyright notice: "Copyright CoCoSoft 2008-2010. All rights reserved."

The system requires a password of at least 8 characters and you should try to use a mixture of upper and lower case and at least one numeric character. This panel is only responsible for the maintenance of the browser password. It does NOT change the root password of the box.

Changing the root user Password for SARK.

The root password can be changed by visiting the underlying SME Server Linux platform. You can connect to it using

<https://local ip-address/server-manager>

Documentation for SME server can be found on the SME server Wiki here

http://wiki.contribs.org/SME_Server:Documentation

The default set-up (PnP install)

SARK is initially set-up to allow you to install it and be up and running within just a few minutes of powering it up. Provided you deploy SIP PnP capable phones (such as Snom, Yealink or Gigaset), they will be automatically configured and registered on-the-fly when they are powered-up.

Once connected to the ISDN lines, the SARK is ready to receive calls. As phones are powered up and configured, they are automatically added to a special ring-group called the 'RINGALL' group. Any calls arriving over ISDN will be automatically sent to the RINGALL group, which will ring all of the phones, and 'first to pick-up' will get the call. The RINGALL group can be removed as and when you move to more sophisticated routing and ACD configurations but initially it just allows you to get up and running very quickly. Also by default, outbound calls will initially be routed via the ISDN hunt group.

Of course, you will probably want to progress to more sophisticated call distribution and routing and you may wish to deploy VoIP trunks and perhaps remote phones. All of these things can be accomplished using the SARK browser-based interface. The following sections we will cover some of these topics in more detail.

Adding Trunks

Outbound trunks for your installed hardware and a general inbound trunk class are automatically installed on the platform so there is nothing to do to activate them. You will probably wish/need to create additional inbound and outbound routing (see Routes below) to handle DDI's, ACD and IVR requirements.



TDM/VoIP Softswitch - Trunklines

- Extensions
- Trunks**
- Routes
- Call Groups
- IVR Menus
- Custom Apps
- Call Back
- Day/night Timers
- Class of Service
- Agents
- Queues
- Greetings
- IP Devices
- Asterisk File Edit
- Syslogs
- Firewall
- Network
- Change Password
- Backup/Regress
- Global Settings

| Line | Trunkname | Type | Tenant | IP Address | Latency | Open | Closed | | | | |
|------------|------------|-------|---------|------------|---------|----------|----------|-----|---|--|--|
| GSM1 | GSM1 | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| GSM2 | GSM2 | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| PIKABRI_G1 | BRIPORT1 | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| PIKABRI_G2 | BRIPORT2 | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| PIKAFXO_G0 | FXOGROUP | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| _XXXX. | Inbound... | Class | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |




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If you are running FXO channels, (or BRI without DNID presentation) then inbound calls will automatically be delivered to the 'operator' entry specified in the globals panel. The system default for the operator is to point to a ring group called RINGALL which simply rings all phones; you can, of course update this to point to another call-group, queue or IVR.

The routing of inbound calls falls into one of two categories; you can route by the Dialed Number ID (DNID) if you have digital lines with DNID enabled or you can route by channel name/number in the case of FXO lines.

To route inbound Digital calls with DNID enabled, simply create one or more DiD (DDI) trunks to route the inbound calls. (see the next section). You may need to confirm with your telephone line supplier (PTT) how many digits will be presented in the DNID (it may be 3 or more digits depending upon your carrier's practice).

To route individual FXO channels you can set up a custom trunks and name them as follows:-

- for FXO, use `fxo{n}`; where {n} is the port number on the tdm card

The following shot shows the creation of a trunk to handle calls to and from channel 1 on the FXO card.

SARK
TDM/VoIP SOFTSWITCH

TDM/VoIP Softswitch - Trunk Create: Carriertype is group

Extensions
Trunks
Routes
Call Groups
IVR Menus
Custom Apps
Call Back
Day/night Timers
Class of Service
Agents
Queues
Greetings
IP Devices
Asterisk File Edit
Syslogs
Firewall
Network
Change Password
Backup/Regress
Global Settings

Asterisk peer:

Dial String Lead-in:

Dial String Lead-out:

Routeable?:

Trunkname:

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N.B. It is important that you set the *routeable?* Variable to 'YES'. This will allow you to route calls arriving on this channel. This will create a trunk like the one highlighted below.

SARK
TDM/VoIP SOFTSWITCH

TDM/VoIP Softswitch - Entry Created!

Extensions
Trunks
Routes
Call Groups
IVR Menus
Custom Apps
Call Back
Day/night Timers
Class of Service
Agents
Queues
Greetings
IP Devices
Asterisk File Edit
Syslogs
Firewall
Network
Change Password
Backup/Regress
Global Settings

| Line | Trunkname | Type | Tenant | IP Address | Latency | Open | Closed | | | | |
|------------|-----------|-------|---------|------------|---------|----------|----------|-----|---|--|--|
| GSM1 | GSM1 | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| GSM2 | GSM2 | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| PIKABRI_G1 | BRIPORT1 | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| PIKABRI_G2 | BRIPORT2 | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| PIKAFXO_G0 | FXOGROUP | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| ._XXXX. | Inbound | Class | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| fxo1 | FXO1 | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |

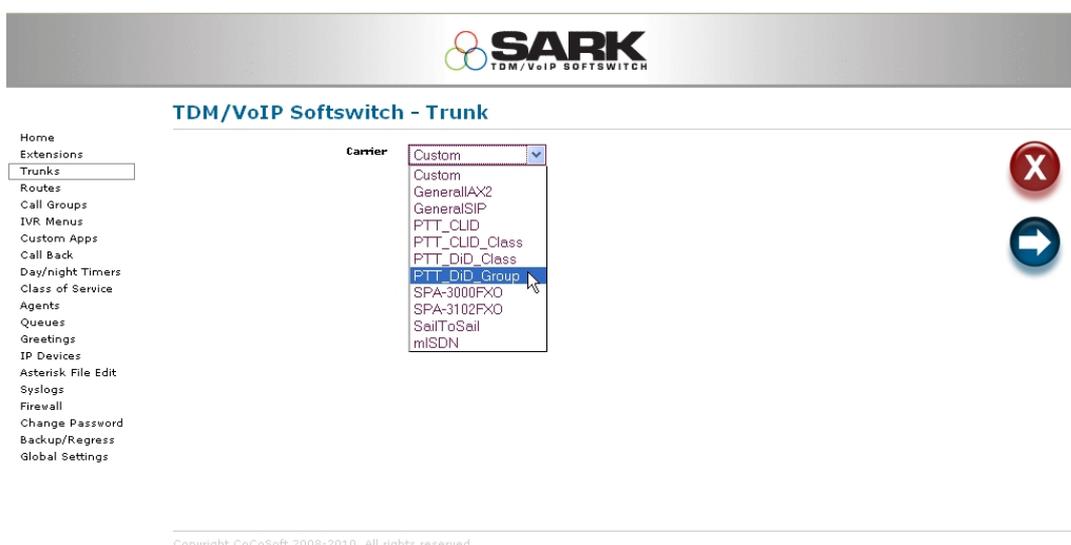
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The trunk can be routed by clicking the edit button and filling out the 'open' and 'closed' route destinations in the normal way.

Adding a DDI (or DiD) Trunk

Digital (ISDN) and VoIP (SIP/IAX2) inbound calls are usually routed by testing the 'Dialed Number ID' (DNID). To confuse matters further; in digital systems these numbers are often allocated in blocks by the telephone company and known as DiD numbers (Direct Inward Dial) or DDI numbers (Direct Dial In). SARK has a special trunk construct to manage and route these numbers. It is called a *PTT_DiD_Group* trunk and you can define a set of one or more contiguous numbers using this trunk type.

Let's add an example DiD group with the range 446640 to 446649. To do that we can create a trunk object called a *PTT_DiD_Group*. Click the new button on the trunks panel and then choose "PTT_DiD_Group" from the drop down.



The screenshot shows the SARK TDM/VoIP Softswitch - Trunk configuration interface. The 'Carrier' dropdown menu is open, displaying a list of options: Custom, GeneralIAX2, GeneralSIP, PTT_CLID, PTT_CLID_Class, PTT_DiD_Class, PTT_DiD_Group (highlighted), SPA-3000FXO, SPA-3102FXO, SailToSail, and miSDN. The left sidebar contains a navigation menu with items like Home, Extensions, Trunks, Routes, Call Groups, IVR Menus, Custom Apps, Call Back, Day/night Timers, Class of Service, Agents, Queues, Greetings, IP Devices, Asterisk File Edit, Syslogs, Firewall, Change Password, Backup/Regress, and Global Settings. The top right corner features a red 'X' button and a blue right-pointing arrow button. The footer text reads: Copyright CoCoSoft 2008-2010. All rights reserved.

Now fill out the number range...



The screenshot shows the SARK TDM/VoIP Softswitch - Trunk Create: Carriertype is DiD configuration interface. The 'DiD Start' field is filled with '446640' and the 'DiD End' field is filled with '446649'. Other fields like 'Add CLI Prefix', 'Alert-info string', 'Smartlink', and 'Display name' are empty. The left sidebar is the same as in the previous screenshot. The top right corner features a red 'X' button, a blue floppy disk icon, and a red button with a white checkmark. The footer text reads: Copyright CoCoSoft 2008-2010. All rights reserved.

Click Commit to create the Trunks. Of course, the set of numbers in your Group can quite legally consist of a single number, in which case you would leave the end box blank. N.B. Even though the update screen requests positive confirmation of a DiD

Group create, it is not unknown for a digit to be entered incorrectly causing hundreds or even thousands of DiDs to be created. Not to worry, if this happens you can simply regress SARK to the commit BEFORE you made a mistake.(see the section on Backup/Regress).

SARK
TDM/VoIP SOFTSWITCH

TDM/VoIP Softswitch - Entries Created!

| Line | Peer | Tenant | IP Address | Latency | Open | Closed | S | A | | |
|-----------|---------|---------|------------|---------|----------|----------|-----|---|---|----|
| 446640 | | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | ✎ | 🗑️ |
| 446641 | | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | ✎ | 🗑️ |
| 446642 | | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | ✎ | 🗑️ |
| 446643 | | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | ✎ | 🗑️ |
| 446644 | | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | ✎ | 🗑️ |
| 446645 | | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | ✎ | 🗑️ |
| 446646 | | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | ✎ | 🗑️ |
| 446647 | | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | ✎ | 🗑️ |
| 446648 | | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | ✎ | 🗑️ |
| 446649 | | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | ✎ | 🗑️ |
| Analog-In | | default | LOCAL | N/A | 401 | 401 | N/A | ✓ | ✎ | 🗑️ |
| PIKABRI | PIKABRI | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | ✎ | 🗑️ |

Line:

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Our trunks have been created and we can now route them to their destinations by clicking the edit button on each trunk.

SARK
TDM/VoIP SOFTSWITCH

PTT_DiD_Group/446640 Settings

Routing - Line

Active? YES

Tenant: default

Fax detect: NO

Switch on CLIP: NO

Open: Operator

Closed: Operator

Description: *IVRs*
testivr
Default IVR
QUEUES
EXTENSIONS
401
401
CALL GROUPS
DISA
DISA
CALLBACK
GET VOICEMAIL
Retrieve Voicemail
LEAVE VOICEMAIL
*400
*401
Siblings
Trunks

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Route the inbound calls by clicking the open and closed inbound route drop downs and choosing a destination from the available system endpoints.

Smartlink

SARK has a flag called "smartlink" which you can set on when you create a DiD group. This is useful when you first create a set of extensions and then create your DiD trunks later. Smartlink will match the numbers of each extension to the right hand side numbers in the DiD group. For example, if you had an extension range of 400 to 420 and a DiD range of 668400 to 668420 then SARK will automatically route inbound calls on 668400 to extension 400; calls on 668401 to 401 and so on. This can save a lot of work when there are lots of extensions to be routed during system setup.

Adding an IAX2 “Sibling” Trunk

Sibling trunks provide a convenient way of routing calls between SARK instances. In this way, you can easily and quickly build a network of interconnected PBX’s which route calls to one another depending upon the number dialed.

To create a Sibling trunk choose SailToSail from the trunk type dropdown in Trunk create

TDM/VoIP Softswitch - Trunk

Carrier: Custom

- Custom
- GeneralIAX2
- GeneralSIP
- PTT_CLID
- PTT_CLID_Class
- PTT_DiD_Class
- PTT_DiD_Group
- SailToSail
- miSDN

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Now fill out the link information. In the Sibling name box enter the hostname of the PBX you wish to link to (just the short name not the FQDN). Fill out the IP address and include a password (this will be the same on both sides of the link). If the Sibling is remote (i.e. not in the same subnet) then you may want to have SARK create a firewall rule for it. In the example below, the other machine is in the same subnet so there is no need to create a rule.

TDM/VoIP Softswitch - Trunk Create: Carriertype is Sibling

Sibling or Trunk name: switch

URI/IP address: 192.168.1.210

Create Firewall Rule?: NO

Password: 1234

Priv: YES

Trunkname: to_switch

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Now click on save or commit to create your new trunk....



TDM/VoIP Softswitch - Entry Created!

- Home
- Extensions
- Trunks
- Routes
- Call Groups
- IVR Menus
- Custom Apps
- Call Back
- Day/night Timers
- Class of Service
- Agents
- Queues
- Greetings
- IP Devices
- Asterisk File Edit
- Syslogs
- Firewall
- Network
- Change Password
- Backup/Regress
- Global Settings

| Line | Trunkname | Type | Tenant | IP Address | Latency | Open | Closed | | | | |
|--------------|------------|---------|---------|---------------|---------|----------|----------|-----|---|--|--|
| Analog-In | INBOUND... | PSTN | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| PIKABRI_G1 | BRIPORT1 | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| PIKABRI_G2 | BRIPORT2 | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| PIKAFXO_G0 | FXOGROUP | group | default | LOCAL | N/A | Operator | Operator | N/A | ✓ | | |
| switchmysark | to_switch | Sibling | default | 192.168.1.210 | 123 ms | Operator | Operator | | ✗ | | |



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SARK has created a trunk for you with a peername which is the concatenation of your hostname and the remote hostname. You must now login to the remote PBX and create a mirror-image Sibling trunk to receive calls and send calls back. Once done, you can create a route to send calls up to the other PBX. In this example, we want to use the remote PBX as our outbound media server (perhaps for PSTN line consolidation) so we will send all of our outbound calls up to it. Here is a route that will do that...



TDM/VoIP Softswitch - Route MAINTRIB

- Home
- Extensions
- Trunks
- Routes
- Call Groups
- IVR Menus
- Custom Apps
- Call Back
- Day/night Timers
- Class of Service
- Agents
- Queues
- Greetings
- IP Devices
- Asterisk File Edit
- Syslogs
- Firewall
- Network
- Change Password
- Backup/Regress
- Global Settings

Active? YES

Path 1

Path 2

Path 3

Path 4

Dialplan

Alternate Route to remote Host

Auth NO

Description

Tenant



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Now, whenever we dial a PSTN number on the Sark500 it will be routed up to the other SARK PBX (perhaps a larger model) for termination onto the PSTN.

Perhaps we also want to allow extension-to-extension calling between the two PBXs. We can extend our route dial plan to include that also. Let's say, for example, that the extensions on the remote PBX are 4 digits in length and always begin with "41". We can modify our route as follows...

SARK
TDM/VoIP SOFTSWITCH

TDM/VoIP Softswitch - Route MAINTRIB

- Home
- Extensions
- Trunks
- Routes**
- Call Groups
- IVR Menus
- Custom Apps
- Call Back
- Day/night Timers
- Class of Service
- Agents
- Queues
- Greetings
- IP Devices
- Asterisk File Edit
- Syslogs
- Firewall
- Network
- Change Password
- Backup/Regress
- Global Settings

Active? YES

Path 1 switchmysark

Path 2 None

Path 3 None

Path 4 None

Dialplan _0XXXXXXXXXX_00XXXXXXXXXX_41XX

Alternate Route to remote Host

Auth NO

Description UP_TO_SWITCH

Tenant default

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We've added the dialplan **41XX** which will route any 4 digit dials beginning 41 up to the other PBX for termination. We would probably also create a dialplan on the remote PBX to route 3 digit numbers beginning 4 back to us (because throughout this guide, our examples use 3 digit extensions on the local PBX, beginning at 401).

Adding Extensions

Extensions for the FXS channels are already defined in the system when it leaves the factory. There is nothing further to do. However, you will almost certainly want to define SIP extensions to the system. If you have SIP Multicast (sometimes called PnP) capable handsets you can use SARK500's ZTP auto-configure feature to set up your phones (see the ZTP/PnP section below). For non-PnP-capable handsets you can add new extensions using the extensions panel ...

To add an extension; click extensions at the top of the navigation menu. Then click the "new" button to create an extension.

The screenshot shows the SARK TDM/VoIP Softswitch - Add an Extension web interface. The page has a header with the SARK logo and the title "TDM/VoIP Softswitch - Add an Extension". On the left is a navigation menu with options: Home, Extensions (selected), Trunks, Routes, Call Groups, IVR Menus, Custom Apps, Call Back, Day/night Timers, Class of Service, Agents, Queues, Greetings, IP Devices, Asterisk File Edit, Syslogs, Firewall, Change Password, Backup/Regress, and Global Settings. The main form contains the following fields:

- Extension Number: 402
- Device: Aastra 480i (dropdown menu)
- RecOpts: SPA-942, SPA-962, SPA-PAP2T, Siemens C460IP, Siemens C470IP, Siemens S450IP, SipStack
- User to receive email notifications: (empty)
- MAC address: (empty)
- Display name: (empty)
- Local/Remote: (empty)
- Tenant: Snom 300, Snom 300 XML, Snom 320, Snom 320 XML, Snom 360, Snom 360 XML, Snom 370, Snom 370 XML
- Alert-info string: (empty)
- Caller ID: (empty)
- Custom Dial String: Snom 820 XML, Snom 870 XML, Snom VXT, Yealink T2x, General SIP

On the right side, there is an "Extension Details" box with the following text: "Fill out the details for the new extension... N.B. If you wish to use SARK's autoprovisioning features then you must provide the MAC address of the phone and choose the correct phone type from the Device drop down. If you are unsure of the phone type or you do not wish to use provisioning then you can simply choose General SIP or General IAX." Below the text are three buttons: a red 'X' button, a blue floppy disk icon button, and a red checkmark button. At the bottom of the page, there is a small copyright notice: "Copyright CoCoSoft 2008-2010. All rights reserved."

Choose the phone type from the device drop-down. Fill out the details for your phone and (optionally) enter the MAC address if you want SARK to automatically generate a TFTP provisioning file for your phone type. SARK has a fairly extensive phone database and it can provision most commercially available phone types. Click the Save button if you have several entries to create or simply click the Commit button to create this entry and regenerate the Asterisk files immediately.

Ext: SIP/402(Snom 820 XML) Settings

- Home
- Extensions**
- Trunks
- Routes
- Call Groups
- IVR Menus
- Custom Apps
- Call Back
- Day/night Timers
- Class of Service
- Agents
- Queues
- Greetings
- IP Devices
- Asterisk File Edit
- Syslogs
- Firewall
- Change Password
- Backup/Regress
- Global Settings

General
Asterisk
Provisioning
CoS
CFWD
XREF

Extension Number

MAC address

User to receive email notifications

Tenant

RecOpts

Local/Remote

Deliver Vmail to

Caller ID

Alert-info string

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SARK will create the necessary Asterisk entries with automatic randomly generated passwords and the correct ACL checking for your subnet...

Ext: SIP/402(Snom 820 XML) Settings

- Home
- Extensions**
- Trunks
- Routes
- Call Groups
- IVR Menus
- Custom Apps
- Call Back
- Day/night Timers
- Class of Service
- Agents
- Queues
- Greetings
- IP Devices
- Asterisk File Edit
- Syslogs
- Firewall
- Change Password
- Backup/Regress
- Global Settings

General
Asterisk
Provisioning
CoS
CFWD
XREF

```

type=friend
username=Ext402
secret=M9cFb2g6
mailbox=402
host=dynamic
qualify=3000
canreinvite=no
context=internal
callerid="Ext402" <402>
pickupgroup=1
callgroup=1
call-limit=3
subscribecontext=extensions
deny=0.0.0.0/0.0.0.0
permit=192.168.1.140/255.255.255.0
disallow=all

```

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...And it will also generate the correct provisioning data for the phone type you chose...

Ext: SIP/402(Snom 820 XML) Settings

- Home
- Extensions
- Trunks
- Routes
- Call Groups
- IVR Menus
- Custom Apps
- Call Back
- Day/night Timers
- Class of Service
- Agents
- Queues
- Greetings
- IP Devices
- Asterisk File Edit
- Syslogs
- Firewall
- Change Password
- Backup/Regress
- Global Settings

General Asterisk Provisioning CoS CFWD XREF

```
[!snom820-$MAC.htm"
<?xml version="1.0" encoding="utf-8"?>
<settings>
<phone-settings e="2">
<dnd_on_code perm="RW">*41*</dnd_on_code>
<dnd_off_code perm="RW">*42*</dnd_off_code>
<ntp_server perm="RW">$localip</ntp_server>
<timezone perm="RW">GBR-0</timezone>
<challenge_response perm="RW">off</challenge_response>
<call_join_xfer perm="RW">on</call_join_xfer>
<tone_scheme perm="RW">GBR</tone_scheme>
<alert_internal_ring_sound perm="RW">Ringer6</alert_internal_ring_sound>
<alert_external_ring_sound perm="RW">Ringer2</alert_external_ring_sound>
<edit_alpha_mode perm="R">123</edit_alpha_mode>
<call_waiting perm="RW">visual</call_waiting>
<update_policy perm="R">auto_update</update_policy>
```



You may freely modify any of these entries SARK has generated for you and save them back to the database. You can also freely create your own provisioning templates and profiles allowing you to tailor the SARK provisioning platform to suit your individual needs.

Automatically adding extensions with ZTP/PnP

The SARK500 can automatically create extensions and provisioning data on-the-fly for phones which support SIP Multi-cast provisioning. This has a couple of major benefits for users of SIP phones that are SIP/PnP aware:-

- There is no need to run DHCP option 66, and no need to run an on-board DHCP server.
- The provisioning stream is synthesized by the listener on-demand
- No tftp or ftp files are stored on the server.
- New phones can optionally be provisioned and defined to asterisk on-the-fly by the listener/builder.
-

Turn on ZTP/PnP in globals panel.

The screenshot displays the SARK TDM/VoIP Softswitch - Globals configuration interface. The 'Services' tab is selected, showing various provisioning options. A red oval highlights the 'PnP Provisioning' and 'Zero Touch Provisioning' settings, both of which are set to 'enabled'. Other visible settings include 'TFTP Server' (enabled), 'XMPP' (disabled), 'XMPP server' (localhost), 'Class Of Service ON/OFF' (ON), 'Tenant support ON/OFF' (OFF), 'Alert-Info for Blind Xfer Bounce' (empty), 'Bounce busy destination' (Operator), 'Campon mini-queue ON/OFF' (OFF), and 'Campon mini-queue options' (1,30). The right sidebar provides system information: Sysinfo (Serial #: 157070, Release #: 3.1.0-115, Web Server: lighttpd, Browser: Firefox), Network (hostname: sark500, domain: cable.virginmedia.net, LAN IP: 192.168.1.107, Netmask: 255.255.255.0), Hardware (System Media: flash, RAM Size: 256208, RAM Free: 44288), and Status (PBX State: RUNNING, System Uptime: 3 days).

'PnP Provisioning' (or SIP Plug 'n Play) turns on the SIP Multicast listener. This allows SARK to asynchronously tell a SIP phone where its provisioning file is located *without* the need to modify the existing DHCP server or to run an on-board DHCP server or even to run the SARK unit at a static IP address.

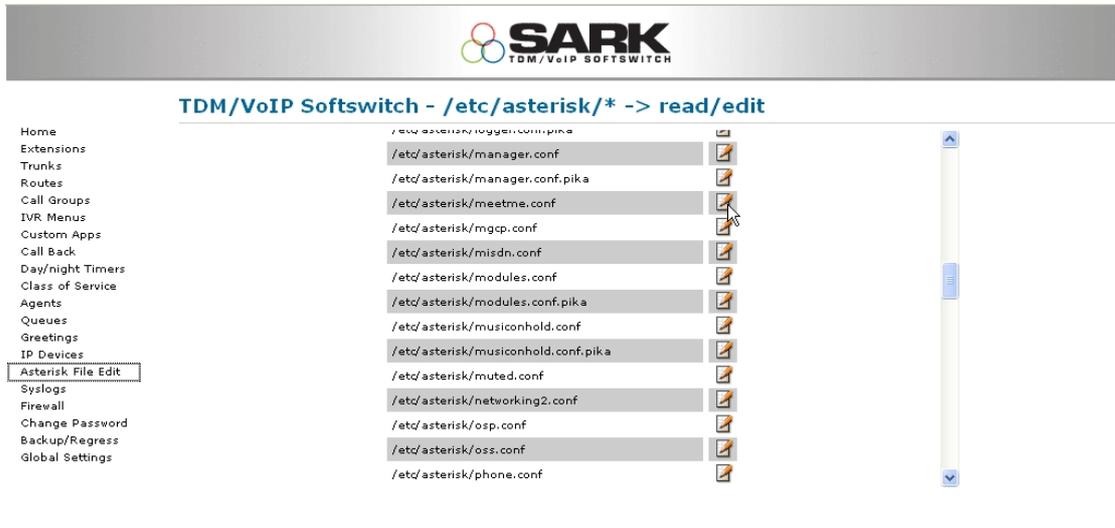
'Zero Touch Provisioning' (ZTP) works in tandem with PnP. When ZTP is enabled SARK will not only inform a phone where its provisioning data is but it will also build the data on-the-fly and update Asterisk at the time the phone first requests it. What this means is that you can simply power up a PnP aware phone and SARK will automatically create an extension for it in the SARK database, inform the phone where the provisioning data is and finally, regenerate Asterisk so that it is aware of the new extension. Thus the phone will come up, be automatically provisioned, register with Asterisk and be immediately ready to make and receive calls. This is very useful for medium to large roll-outs because it needs no prior set-up.

For security reasons, you should only run ZTP during initial system installation or when you have new phones to roll-out. The remainder of the time it should be switched off.

In order to be able to run multiple PBX servers in the same subnet, the Multicast listener must be able to selectively reply to only those PnP aware phones which it "owns" (i.e. are defined to it). For this reason, unless ZTP is enabled, SARK will only respond to requests from phones for which it already has an entry in its database. It recognizes individual phones by their MAC addresses. This mode of operation is much the same as regular DHCP 66 type provisioning; i.e. only phones which have been predefined to the system will be provisioned. In this way, multiple servers may all be listening for Multi-casts and each can respond to its own group of phones without disrupting other listeners.

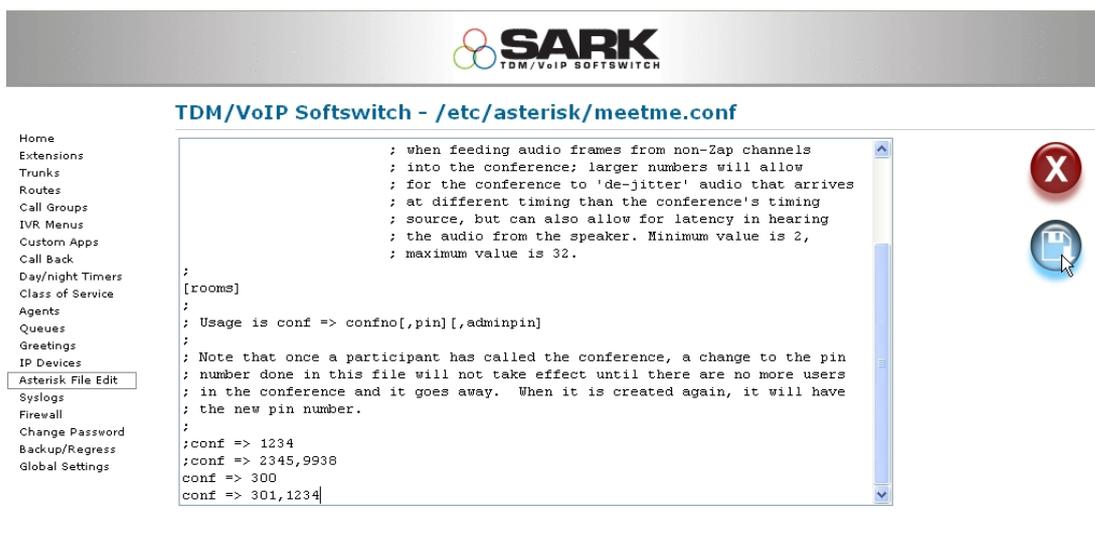
Conferencing

Conference rooms 300-303 are predefined to the system at the factory. If you want to add other rooms or change the conference room number you can modify the underlying Asterisk ".conf" file directly (it is called *meetme.conf*). As a general rule, SARK keeps its management of Asterisk files to a minimum. You can directly modify most asterisk files just using the Asterisk Edit panel...



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To modify a conference room simply choose the *meetme.conf* file and make your changes directly. In the example below we've just added conference room 301 and now we are about to save it back.



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Once the conference room has been created in this way we can either dial it directly from any extension or we can select it in a route (it will auto magically appear in all end-point drop downs)... Below we route a DiD trunk to our new conference room so we can host a conference with an external callers.



PTT_DiD_Group/446641 Settings

Home

Extensions

Trunks

Routes

Call Groups

IVR Menus

Custom Apps

Call Back

Day/night Timers

Class of Service

Agents

Queues

Greetings

IP Devices

Asterisk File Edit

Syslogs

Firewall

Change Password

Backup/Regress

Global Settings

Routing Line

Active? YES

Tenant default

Fax detect NO

Switch on CLIP NO

Open Operator

Closed Default IVR

Description **QUEUES**

EXTENSIONS

400

CALL GROUPS

DISA

DISA

CALLBACK

GET VOICEMAIL

Retrieve Voicemail

LEAVE VOICEMAIL

*400

Siblings

Trunks

PIKABRI

PIKAFXD

Custom Apps

Conferences

300

301






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Interactive Voice Response

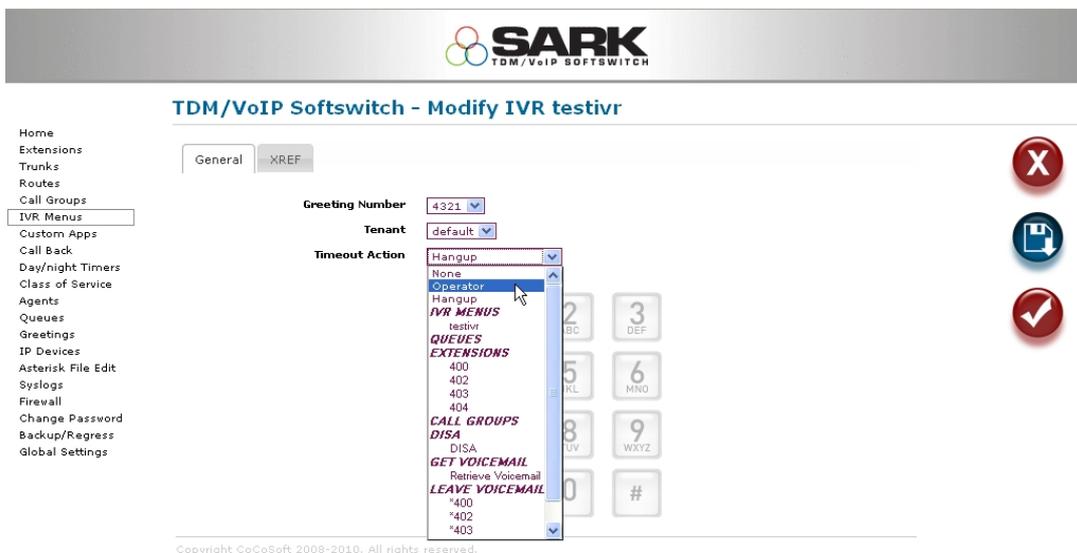
To create an IVR, you must first create a system greeting.. Simply go to any phone and dial *60*+nnnn, where nnnn is a 4 digit number that we will associate with our recording. -

***60*4321**

The auto attendant will lead you through the process of creating a recording. Let's work an example and record a message like this...

'Welcome to ABC Widget Corporation. Press 1 for sales, press 2 for accounts or please hold for an operator'

We can now go ahead and create our IVR. Choose the IVR panel from the left hand menu and then click "New" to create a new IVR.



We choose our greeting (4321) from the drop down and then decide what to do if the caller does not press a key. Our greeting said "...or please hold for an operator", so we can choose "Operator" from the timeout drop down.

We can now choose our key-press operations by clicking on the keypad graphic and choosing an endpoint from the drop down.

SARK
TDM/VoIP SOFTSWITCH

TDM/VoIP Softswitch - Modify IVR testivr

Home
Extensions
Trunks
Routes
Call Groups
IVR Menus
Custom Apps
Call Back
Day/night Timers
Class of Service
Agents
Queues
Greetings
IP Devices
Asterisk File Edit
Syslogs
Firewall
Change Password
Backup/Regress
Global Settings

General XREF

Greeting Number: 4321
Tenant: default
Timeout Action: Operator

KEY1
Action on Keypress
None
Tag entry
Alert Info

4 GHI
7 PQRS
* 0 #

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And....

SARK
TDM/VoIP SOFTSWITCH

TDM/VoIP Softswitch - Modify IVR testivr

Home
Extensions
Trunks
Routes
Call Groups
IVR Menus
Custom Apps
Call Back
Day/night Timers
Class of Service
Agents
Queues
Greetings
IP Devices
Asterisk File Edit
Syslogs
Firewall
Change Password
Backup/Regress
Global Settings

General XREF

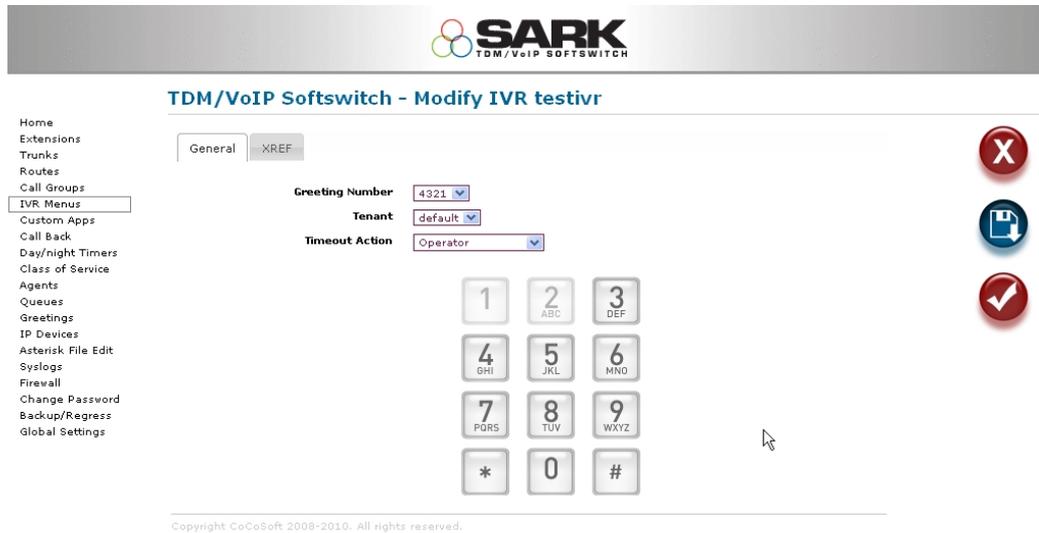
Greeting Number: 4321
Tenant: default
Timeout Action: Operator

KEY1
Action on Keypress
None
Operator
Hangup
IVR MENU
testivr
QUEUES
EXTENSIONS
400
402
403
404
CALL GROUPS
DISA
DISA
GET VOICEMAIL
Retrieve Voicemail
LEAVE VOICEMAIL
*400
*402
*403

1
4 GHI
7 PQRS
*

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Finally, after committing the IVR, if we return to the screen we see .that SARK has updated the graphic to show at a glance which keys are in use...



SARK
TDM/VoIP SOFTSWITCH

TDM/VoIP Softswitch - Modify IVR testivr

General XREF

Greeting Number 4321
Tenant default
Timeout Action Operator

| | | |
|-----------|----------|-----------|
| 1 | 2 ABC | 3 DEF |
| 4 GHI | 5 JKL | 6 MNO |
| 7 PQRS | 8 TUV | 9 WXYZ |
| * | 0 | # |

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The IVR is now ready to use.

Adding Routes

Within SARK, an outbound route consists of a set of dialplans and up to four outbound pathways (Trunks). The SARK HPE processing engine will choose the first available route which matches the dialed number ID (DNID) .

To add a route, click the link Routes in the left navigation menu and then click the “new” button.

The screenshot shows the SARK TDM/VoIP Softswitch - Route AnalogueOut configuration page. The page has a header with the SARK logo and the text 'TDM/VoIP SOFTSWITCH'. Below the header is a navigation menu on the left with the following items: Home, Extensions, Trunks, Routes (highlighted), Call Groups, IVR Menus, Custom Apps, Call Back, Day/night Timers, Class of Service, Agents, Queues, Greetings, IP Devices, Asterisk File Edit, Syslogs, Firewall, Change Password, Backup/Regress, and Global Settings. The main content area is titled 'TDM/VoIP Softswitch - Route AnalogueOut' and contains the following configuration fields: Active? (YES), Path 1 (PIKAFXO), Path 2 (None), Path 3 (None), Path 4 (None), Dialplan (XXXX_1XX), Alternate Route to remote Host (empty), Auth (NO), Description (Main trunk), and Tenant (default). On the right side of the form, there are three buttons: a red 'X' button, a blue 'Save' button, and a red checkmark button. At the bottom of the page, there is a copyright notice: 'Copyright CoCoSoft 2008-2010. All rights reserved.'

You can configure up to four trunks for each outbound route. They will be successively tried in the event that a trunk fails or refuses to honor the Dial. The dialplans are conventional Asterisk dialplans and you can code as many as you wish using white space as a separator.

Click Save or Commit to save your new route into the database and/or bring it into service.

Firewall

IN SME based system, the firewall is set automatically. In its default configuration, only the following ports are opened:-

| | |
|-----------------------|-----|
| HTTP (TCP 80) | WAN |
| HTTPS (TCP 443) | LAN |
| NTP (TCP 123) | WAN |
| SSH (TCP 45678) | WAN |
| TFTP (UDP 69) | LAN |
| SIP (UDP 5060) | WAN |
| RTP (UDP 10000:20000) | WAN |
| IAX2 (UDP 4569) | WAN |

However, you can change/restrict these settings; either in your external firewall or by modifying the SME firewall settings.

Firewall guidelines

- Close port 5060 to the WAN unless you intend to conduct remote SIP operations (using a SIP carrier or remote SIP phone(s)). With many SIP carriers it isn't necessary to open 5060 at all because they will keep the port alive from the time you register. Check with your carrier.
- Close port 4569 to the WAN unless you intend to conduct remote IAX2 operations (using an IAX2 carrier or a remote IAX2 device).
- Close the RTP ports (Usually 10000-20000) to the WAN unless you intend to conduct remote SIP operations.
- ONLY open the firewall to SIP endpoints you know (unless the far end is using a Dynamic IP which changes often, in which case try to limit access to just that range, even if it is quite large, or consider using a VPN - see below).

SME Server Firewall

In SARK-3.1 we define the objects "sailSIP" and "sailIAX" in the SME server database in order to manage the ports. To modify the default settings you must log in using ssh. The default ssh port is TCP 45678 but you can change this in the remote-access section of the server-manager.

Closing SIP and IAX2 ports in the SME server firewall

```
[]# config setprop sailSIP status disabled  
[]# signal-event remoteaccess-update
```

```
[]# config setprop sailIAX status disabled  
[]# signal-event remoteaccess-update
```

Restricting SIP and IAX2 port access in the SME server firewall

You can also add an *AllowHosts* statement to control external access;

```
[ ]# config setprop sailSIP AllowHosts 111.111.111.111/0,222.222.222.222/24  
[ ]# signal-event remoteaccess-update
```

In this way you can limit access to just those remote extensions and trunks you want to allow in. The other alternative is to deploy your SAIL/SARK box behind a good quality firewall which can control access to the ports on behalf of the PBX. For frequently changing IP addresses such as home workers or road warriors we would recommend bringing them in over VPN if you can. Adding openvpn to SME server itself or to your firewall server is straightforward and you can then use softphones on VPN connected PC's or hardphones which can run openVPN (Snom 370,820,870 can all run openvpn).

Finally, you should make sure that you put strong passwords and ACLs onto any IAX to IAX trunks you have that span the internet.

Remote phones

If you can, use a phone type which supports VPN tunneling. Snom 370, 820, 870 all support openvpn and you should also be able to run softphones over a VPN. If you can't, then implement strong passwords and ACL checking in the SIP definition for the remote IP address.

Backup/Regress & Save to USB

SARK will automatically spawn backups every night. It will keep up to 7 on-board backups and you can “regress” to a backup simply by clicking on it. It will also take an “Instant Backup” (called a snapshot) every time you issue a commit. You can take a commit (or issue an explicit backup) before you begin a piece of work or make a change and then perhaps regress to the start version if you are not satisfied with the end result or if you make a mistake during configuration. You can view your current snapshots & backups by choosing Backup/Regress from the Navigation menu.

SARK
TDM/VoIP SOFTSWITCH

TDM/VoIP Softswitch - Database Backups

Home
Extensions
Trunks
Routes
Call Groups
IVR Menus
Custom Apps
Call Back
Day/night Timers
Class of Service
Agents
Queues
Greetings
IP Devices
Asterisk File Edit
Syslogs
Firewall
Network
Change Password
Backup/Regress
Global Settings

| Filename | create date | |
|-----------------------------------|--------------------------|-------------------------|
| /opt/sark/bkup/sark.db.1298732704 | Sat Feb 26 15:05:04 2011 | Regress |

Database Snapshots

| Filename | create date | |
|-----------------------------------|--------------------------|-------------------------|
| /opt/sark/snap/sark.db.1298732737 | Sat Feb 26 15:05:37 2011 | Regress |

Backups/Snapshots

You can regress the database to any of your backups or snapshots by clicking the regress link. By default, the system will take a backup every day and keep it for 7 days. It will also take a snapshot every time you issue a commit and keep the last 6 snapshots. You can also take an instant backup at any time by clicking the blue backup button to the right. Backups and Snapshots highlighted in yellow are identical to the current working copy(except perhaps for the SQLite update count).

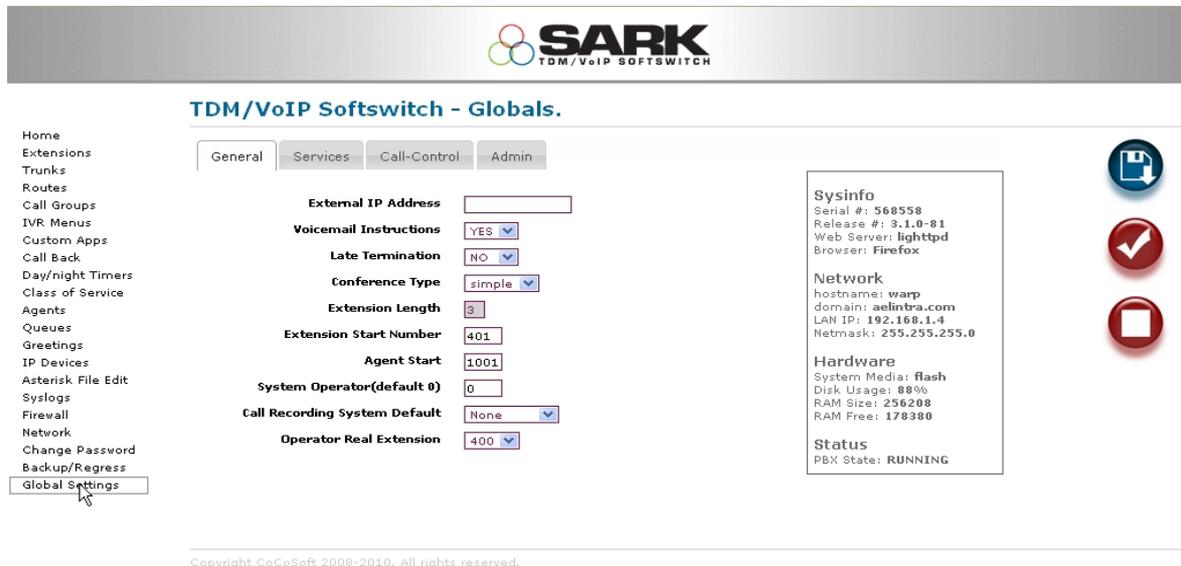
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To regress to an earlier database version simply click the regress link next to the backup or snapshot you wish to use. Regression is non-destructive because SARK will automatically take a backup of the current database immediately before the regression. Thus you can freely “regress the regression” if required. Regressions and snapshots which are identical to the current working copy are highlighted in yellow.

You can save all of your backups and snapshots to an external USB drive. Insert the USB drive into the USB slot on the rear of the appliance and click Backup/Regress. An additional “save to disk” button will appear (see the image above). Clicking on the button will save all of your snaps and backups into a pair of time-stamped tar files on the USB drive. You can now remove your backup and store it in a safe place.

Global Settings

The Global Settings Web page allows you to configure global PBX settings. In the left navigation menu, click Global Settings. The following Web page appears.



SARK
TDM/VoIP SOFTSWITCH

TDM/VoIP Softswitch - Globals.

General Services Call-Control Admin

External IP Address

Voicemail Instructions YES

Late Termination NO

Conference Type simple

Extension Length 3

Extension Start Number 401

Agent Start 1001

System Operator(default 0) 0

Call Recording System Default None

Operator Real Extension 400

Sysinfo
Serial #: 568558
Release #: 3.1.0-81
Web Server: lighttpd
Browser: Firefox

Network
hostname: warp
domain: aelintra.com
LAN IP: 192.168.1.4
Netmask: 255.255.255.0

Hardware
System Media: flash
Disk Usage: 88%
RAM Size: 256208
RAM Free: 178380

Status
PBX State: RUNNING

Home
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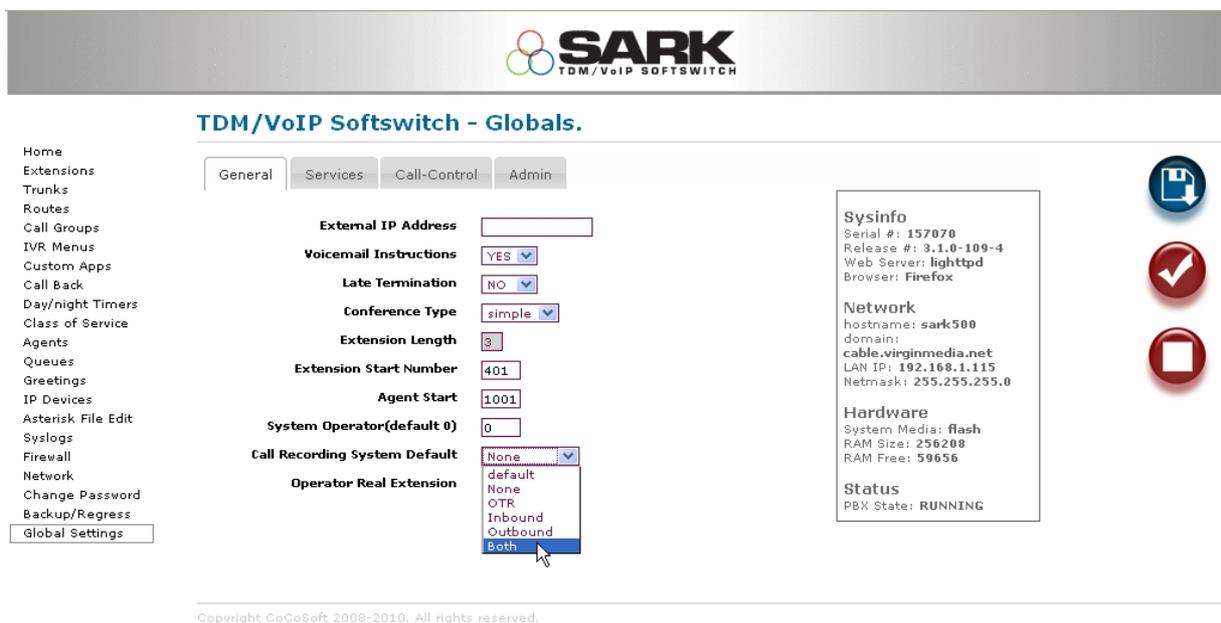
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You can click through the various tabs to set the system up to your requirements.

Call Recording

SARK has several different settings for call recording which you can choose to suit your requirements. However, before you begin call recording, you should be aware that in most countries and jurisdictions there are legal implications whenever you record a call and that neither Aelintra Telecom nor its distributors or resellers will be responsible in the event that you use the recording features in an unlawful manner. If you are in any doubt regarding the legality of your proposed recordings then you should seek legal advice *before* you begin.

You can specify the default type of call recording you want to make at the system level and you can override this setting for each individual endpoint. The system setting is in the *Globals* panel under the *General* tab.



Whatever you choose in this section will be inherited by any endpoint which has its call recording preferences set to *default*. You can also override this directive by visiting the edit panel for an endpoint and setting its preference to some value other than *default*.

Recording process

All voice capture is done directly to `/var/spool/asterisk/monitor`. Immediately after a recorded call ends, a task is called to off-load the recording from *monitor* to a staging dataset (MONITOROUT). The default is `/var/spool/asterisk/monout` and you can use a cron updater to move the recordings over the network if required. Recordings are named as follows:-

{Linux Epoch}-{DNID}-{CLID}.wav

An example calling from extension 5099 to a test number; 01924 566170, looks like this

1263728106-01924566170-5099.wav

Asterisk File Edit

You can freely modify most of the Asterisk “.conf” files to suit your needs. There are a some files which SARK maintains and it won't allow you to modify them directly but even those files have user include files that you can use to create and maintain your own settings. To change an Asterisk file simply select it using the Asterisk file edit panel.



SARK
TDM/VoIP SOFTSWITCH

TDM/VoIP Softswitch - /etc/asterisk/* -> read/edit

- Home
- Extensions
- Trunks
- Routes
- Call Groups
- IVR Menus
- Custom Apps
- Call Back
- Day/night Timers
- Class of Service
- Agents
- Queues
- Greetings
- IP Devices
- Asterisk File Edit**
- Syslogs
- Firewall
- Network
- Change Password
- Backup/Regress
- Global Settings

| | |
|----------------------------------|--|
| /etc/asterisk/ads_i.conf | |
| /etc/asterisk/adtranvofr.conf | |
| /etc/asterisk/agents.conf | |
| /etc/asterisk/alarmreceiver.conf | |
| /etc/asterisk/alsa.conf | |
| /etc/asterisk/amd.conf | |
| /etc/asterisk/asterisk.ads_i | |
| /etc/asterisk/asterisk.conf | |
| /etc/asterisk/cdr.conf | |
| /etc/asterisk/cdr_custom.conf | |
| /etc/asterisk/cdr_manager.conf | |
| /etc/asterisk/cdr_mysql.conf | |
| /etc/asterisk/cdr_odb_c.conf | |
| /etc/asterisk/cdr_pgsql.conf | |
| /etc/asterisk/cdr_tds.conf | |

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In the above example we have chosen to edit the Asterisk cdr.conf control file...

TDM/VoIP Softswitch - /etc/asterisk/cdr.conf

- Home
- Extensions
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- Asterisk File Edit**
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- Global Settings

```
; Asterisk Call Detail Record engine configuration
;
; CDR is Call Detail Record, which provides logging services via a variety of
; pluggable backend modules. Detailed call information can be recorded to
; databases, files, etc. Useful for billing, fraud prevention, compliance with
; Sarbanes-Oxley aka The Enron Act, QOS evaluations, and more.
;
[general]
; Define whether or not to use CDR logging. Setting this to "no" will override
; any loading of backend CDR modules. Default is "yes".
enable=no
; Define whether or not to log unanswered calls. Setting this to "yes" will
; report every attempt to ring a phone in dialing attempts, when it was not
; answered. For example, if you try to dial 3 extensions, and this option is
; "yes",
; you will get 3 CDR's, one for each phone that was rung. Default is "no". Some
```



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We can now freely edit the file and save our changes by clicking the save button. If it is a change which requires an Asterisk restart then Asterisk can be stopped and started from the Global Settings panel.

If you choose a file that SARK maintains, then the content will still be viewable but it will be “grayed out” and you won't be able to save or update it. One of the files which SARK manages is called extensions.conf...

TDM/VoIP Softswitch - /etc/asterisk/extensions.conf

- Home
- Extensions
- Trunks
- Routes
- Call Groups
- IVR Menus
- Custom Apps
- Call Back
- Day/night Timers
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- Asterisk File Edit**
- Syslogs
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- Global Settings

```
; DO NOT MODIFY THIS FILE
; It is generated by the SARK/SAIL Asterisk generator code and
; it will be overwritten each time you issue a COMMIT in the
; SARK/SAIL GUI workbench.
;
[general]
static=yes
writeprotect=yes
[globals]
LOCALIP=192.168.1.4
ABSTIMEOUT=3600
ALLOWHASHXFER=disabled
BLINDBUSY=Operator
BOUNCEALERT=
CALLRECORD1=None
EXTLEN=3
FAX=401
FAXDETECT=4
INTRINGDELAY=20
LTERM=NO
```



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As you can see from the above, when you display a system managed file, the “Save” button is not presented and the content is set to read-only.

SARK Service Codes

| TRANSFERS, PARKING & GROUP PICKUP | | | | |
|-------------------------------------|-----------|--|--|--|
| Key | data | function | Availability/Terminal Type | Comment |
| # # | extension | Blind Transfer | internal | Immediately transfer a call - use with analogue phones - usually turned off |
| *1 | none | One Touch Record(OTR) or One Touch Retrospective Record(OTRR)* | internal | Activates call recording This is an in-band request. Actual key sequence may vary - refer to site administrator |
| *2# | extension | Attended Transfer | internal | Dial callee then transfer - use with analogue phones - usually turned off |
| <i>Transfer Key</i> | extension | Transfer | All SIP phones | Either blind or attended depending upon the phone type |
| ##*5# | none | Call Park | internal | Parks the call into the first free park - usually used with BLF parks |
| *8 | none | Call Pickup | internal | Pick up ringing call on same group |
| *8 | extension | Directed Call Pickup | internal | Pick up specific ringing call - usually used with BLF and Visual Call Pickup on Snom or Aastra |
| Hold | none | Hold | SIP phones which have a hold function or key | Hold call and play MOH |
| <i>Message Light/Mail Icon *50*</i> | none | Pick up voicemail | internal | Connect to Voicemail |
| 300 - 307 | none | Conference Rooms | internal | Actual key sequence may vary depending upon extension number choice and other factors |

| | | | | |
|-------------------------|------|---|----------|---|
| 499 | none | Broadcast directly to the overhead page amps using line out | internal | Check with your administrator to see if this feature is activated - Actual key sequence may vary depending upon extension number choice |
| 700 - 702 | none | Parked call pickup | internal | Actual key range may vary depending upon extension number choice- refer to site administrator |
| * + <i>extension</i> | none | dial or transfer to an extension's Voicemail | internal | Useful to transfer a call direct to an extension's voicemail box. |
| **+ <i>extension</i> | none | dial or transfer to an extension's MiniQueue (camp-on) | internal | Used to transfer a call to a busy extensions "camp-on" queue. |

*OTRR Requires Advanced recording feature to be installed.

Diverts

| DIVERTS | | | | |
|----------|-------------------|--|----------------------------|---|
| Key | data | function | Availability/Terminal Type | Comment |
| *20* | none | Call Forward Immediate to Voicemail Toggle (DND) | internal | sets feature on/off |
| *21*+ext | none or extension | Call Forward Immediate (CFIM) Toggle | internal | Use with no extension to turn off |
| *22*+ext | none or extension | Call Forward on Busy/No Answer (CFBS) Toggle | internal | Use with no extension to turn off |
| *23* | none | Clear All Call Forward Instructions (CLACF) | internal | clear all call forward instructions for this extension |
| *27*+ext | extension | Follow Me (RCFIM) | internal | Direct a remote extension's calls to the local station - requests your vmail passwd |
| *12*+ext | extension | Temporarily forward operator calls to this extension | master user | Useful in small office environments during operator rest/break periods |

Permissions

| PERMISSIONS | | | | |
|-------------|------|--|---------------------------------|--|
| Key | data | function | Availability/Terminal Type | Comment |
| *24* | none | Allow Call Forward to External Numbers | master user - requires password | |
| *25* | none | Deny Call Forward to External Numbers | master user - requires password | |
| *30* | none | Set Master Timers to Automatic (default) | master user - requires password | Incoming calls will be checked against time of day |
| *31* | none | Set Master Timers to global "CLOSED" | master user - requires password | Incoming calls will be routed to closed handler |
| *32* | none | Set Master Timers to global "OPEN" | master user - requires password | Incoming calls will be routed to open handler |

Voicemail

| VOICEMAIL | | | | |
|-----------|----------------------|------------------|----------------------------|---|
| Key | data | function | Availability/Terminal Type | Comment |
| *50* | password | Local Voicemail | internal | connect to voicemail from own station |
| *51* | Extension & password | Remote Voicemail | internal/external | Connect to voicemail from another station |

Miscellaneous operations

| Miscellaneous | | | | |
|---------------|---|---------------------------|----------------------------|---|
| Key | data | function | Availability/Terminal Type | Comment |
| *26*+n | none or ring delay | Set ring delay (seconds) | internal | if no ring delay is specified the ring will not timeout |
| *40*+group | none, pagegroup (call group) or extension | Page | internal | Use with no extension to page all phones |
| *52* | none | Echo test | internal/external | |
| *55* | none | Time and date | internal | |
| *56* | none | Check my extension number | internal | |

System Greetings

| System Greetings | | | | |
|------------------|-------------------------|-----------------------------|----------------------------|--------------------------|
| Key | data | function | Availability/Terminal Type | Comment |
| *60*+greet | 4-digit greeting number | Record a system greeting | internal | Record a system greeting |
| *61*+greet | 4-digit greeting number | Listen to a system greeting | internal | |

Agents and Queues

| Agents/Queues | | | | |
|---------------|-------------------------------------|---------------------|---------------------------------|--|
| Key | data | function | Availability/Terminal Type | Comment |
| *65* | Agent Number/Password | Log on as an agent | internal | |
| *66* | Agent Number/Password followed by # | Log off as an agent | internal | |
| *67* | Agent Number | ChanSpy Function | master user - requires password | Allows a supervisor to anonymously listen to agents servicing Queues |
| *68* | extension | ChanSpy Function | master user - requires password | Allows a supervisor to anonymously listen to an extension |

NANP Supported Service Codes

| Key | data | function | Availability/Terminal Type | Comment |
|-----|--------------------|--|----------------------------|---|
| *60 | none | Time and date | internal | |
| *65 | none | Check my extension number | internal | |
| *72 | Extension | Call Forward Immediate (CFIM) | internal | |
| *73 | none | Cancel Call Forward Immediate (CFIM) | internal | |
| *78 | none | DND ON | internal | |
| *79 | None | DND OFF | | |
| *90 | extension | Call Forward on Busy/No Answer (CFBS) | internal | |
| *91 | extension | Cancel Call Forward on Busy/No Answer (CFBS) | internal | |
| *97 | password | Local Voicemail | internal | connect to voicemail from own station |
| *98 | extension/password | Remote Voicemail | internal/external | Connect to voicemail from another station |
| *99 | Greeting Number | Listen to a greeting | internal | |