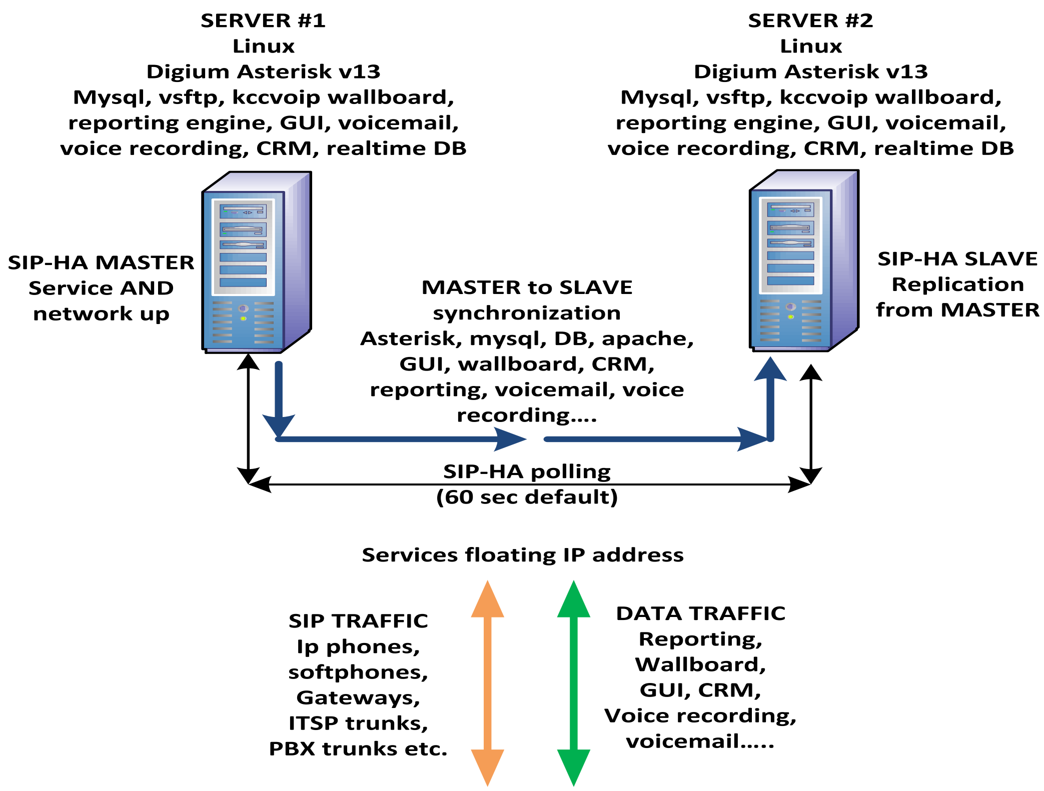
SUMMARY - KCCVoIP SIP-HA

SIP-HA is a low cost solution for sites with a requirement for high availability of SIP Linux Voice Servers (Digium Asterisk, Kamailio, ViciDial, FreePBX, AST-Now, TrixBox, SER, SIPs etc..). This solution provides secure file synchronization and failover for non-clustered pair of servers and primary/secondary controllers for clustered groups.

In non-clustered mode the Linux servers run active/active to ensure backup services are available when required and that all configuration and support files are synchronised and up to date. The key VoIP services are kept running on the backup server, but are controlled to ensure only the 'SIP-ACTIVE' server (MASTER = default) is able to communicate with the endpoints (ip phones, softphones, PRI, ISDN, FXO gateways, PBX switches and ITSP trunks etc.)

Default – MASTER is SIP-ACTIVE SERVER

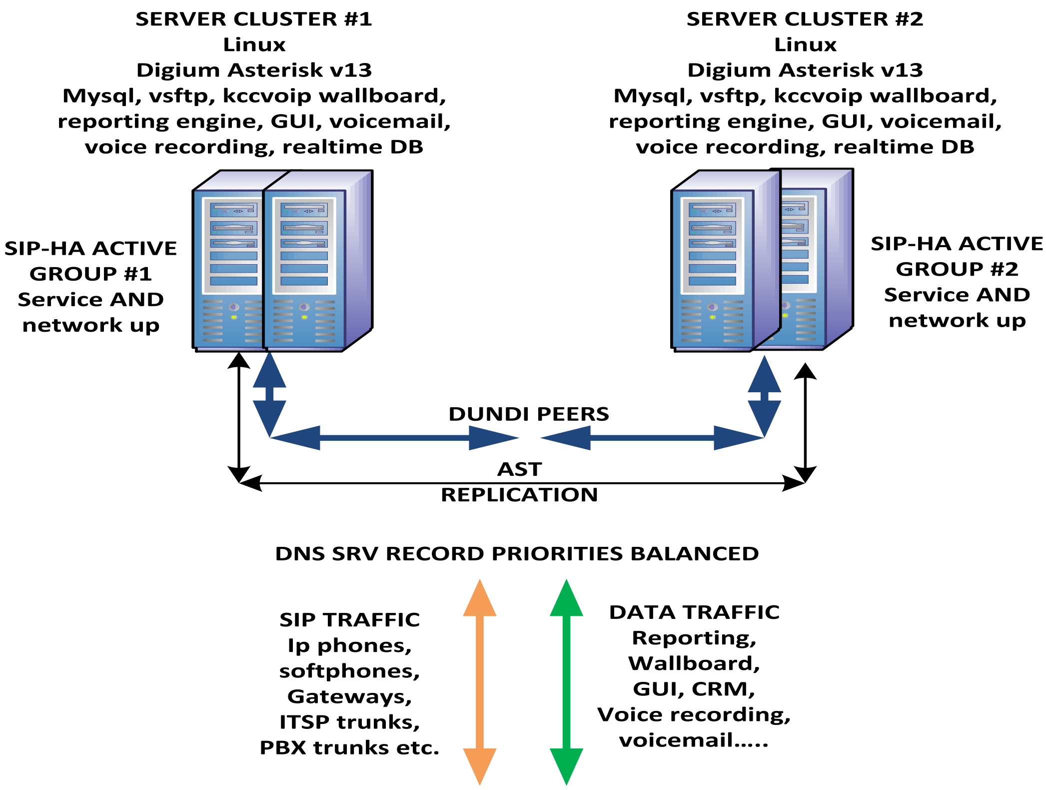


Features Provided ;

- Failover for all voice services and support applications
    - SIP Trunks from providers (dedicated and sub interfaces)
    - SIP/SCCP/H323 etc for telephony endpoints
    - HTTP/HTTPS/SFTP for management and voice applications
    - KCCVoIP Reporting-Engine, VoiceRecording and database

- Auto synchronization of configuration and system files
    - Configurable directories and files required for synchronization
    - System files, configuration, data and programs updated on the SIP-ACTIVE server are automatically sent to synchronize the other server (every 60 seconds – update checks are made)
    - Files can be synchronised to NAS and SAN (or NFS etc anywhere)
    - MASTER and SLAVE can share NAS or SAN storage areas if required
    - Database replication maintains CDR, reporting and realtime data
- Auto failover of MASTER to SLAVE in less than 60 seconds
    - Failover can be done instantly by manual intervention
    - Failover tracks the status of the SIP applications and network reachability
- Notification of failover
    - Failover and sync status eMails sent if required
    - Full log history and timestamps on both MASTER and SLAVE
    - SNMP & SYSLOG supported

example – cluster HA



Simple command line functionality

```
        KCCVoIP High Availability Routines for Asterisk
        ~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~


help:   sip-ha script options

         Options:
         -l       enable logging
         -v       verify - show on command line
         -c       copy the current sip-ha to the other server
         -d       disable active services on this server
         -f       manual failover - stop this active server
         -?       help display
```

Example – non-cluster Master server with Asterisk not running

```
       KCCVoIP High Availability Routines for Asterisk
       ~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
-v verfiy information:
master value      : 1 - this server is MASTER
master server is  : 192.168.0.110 management
slave server is   : 192.168.0.100 management
interface         : eth0  physical
timestamp         : 02/17/2014-11:36:56
service address   : 192.168.0.120  controlled by kccvoip SIP-HA

   not active for services on this server for 192.168.0.120

      one of the services on this server needs attention

         S E R V I C E   D O W N   L O C A L L Y
         ****************************************
```

Example – non-cluster Master server with Asterisk running and network up/up

```
       KCCVoIP High Availability Routines for Asterisk
       ~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
-v verfiy information:
master value      : 1 - this server is MASTER
master server is  : 192.168.0.110 management
slave server is   : 192.168.0.100 management
interface         : eth0  physical
timestamp         : 02/17/2014-11:39:38
service address   : 192.168.0.120  controlled by kccvoip SIP-HA

 ACTIVE SERVICES ON THIS SERVER - this server is MASTER

    - NOW IN NORMAL STATE -
```

Example – Slave server with Asterisk running and network up/up

```
       KCCVoIP High Availability Routines for Asterisk
       ~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~

-v verfiy information:
master value      : 0 - this server is SLAVE
master server is  : 192.168.0.110 management
slave server is   : 192.168.0.100 management
interface         : eth1  physical
timestamp         : 02/17/2014-11:46:31
service address   : 192.168.0.120  controlled by kccvoip SIP-HA

    ACTIVE SERVICES ON OTHER SERVER - this server is SLAVE
```

Example – Slave server after Master server failure detected

```
       KCCVoIP High Availability Routines for Asterisk
       ~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~

 -v verify information:

 this server is SLAVE  in  active/standby HA pair
master server is : 192.168.0.216 service primary
slave server is  : 192.168.0.215 service secondary
local interface  : eth0  physical
timestamp        : 02/05/2017-09:08:20

VIP service address #1 : 192.168.0.218 controlled by kccvoip SIP-HA

 Source IP 192.168.0.215 reachable
 Replication IP 192.168.0.216 NOT reachable
 BT SIP Trunk NOT configured - NOT available

Privilege escalation protection disabled!
See https://wiki.asterisk.org/wiki/x/1gKfAQ for more details.
 **** activating services for 192.168.0.218 ****
ARPING 192.168.0.218 from 192.168.0.218 eth0
Sent 1 probes (1 broadcast(s))
Received 0 response(s)
 services active on this server for 192.168.0.218
services restart
Stopping httpd: [  OK  ]
Starting httpd: [  OK  ]

 ACTIVE SERVICES NOW ON THIS SERVER - this server is SLAVE  in  active/standby HA pair
```

Notice the above example shows that the slave server will try to update the master server with all changes that are made to the active slave server during this failover condition so that both servers are kept in full synchronization and the master server will be up to date and ready to take over services again once normal state has been confirmed

KCCVoIP sip-ha will wait for the operators to check that the master server is fully functional again BEFORE it fails back over to the 'normal state' with the master server being the active server once again.

```
       KCCVoIP High Availability Routines for Asterisk
       ~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
-v verfiy information:
master value      : 1 - this server is MASTER
master server is  : 192.168.0.110 management
slave server is   : 192.168.0.100 management
interface         : eth0  physical
timestamp         : 02/17/2014-11:54:43
service address   : 192.168.0.120  controlled by kccvoip SIP-HA

    ACTIVE SERVICES ON OTHER SERVER - this server is MASTER

   N O W   I N   F A I L O V E R   S T A T E   -   SLAVE HAS ACTIVE SERVICES
   *****************************************       *************************
```

Then the operator issues the manual command on the slave server to force the service back in to the 'normal state' from the slave server (sip-ha –f) and replication will then continue in the master to slave direction

```
       KCCVoIP High Availability Routines for Asterisk
       ~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
 active service stopped on this server for 192.168.0.120
services stop
```

```
       KCCVoIP High Availability Routines for Asterisk
       ~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
-v verfiy information:
master value      : 0 - this server is SLAVE
master server is  : 192.168.0.110 management
slave server is   : 192.168.0.100 management
interface         : eth1  physical
timestamp         : 02/17/2014-11:59:52
service address   : 192.168.0.120  controlled by kccvoip SIP-HA

    ACTIVE SERVICES ON OTHER SERVER - this server is SLAVE
```

```
       KCCVoIP High Availability Routines for Asterisk
       ~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
-v verfiy information:
master value      : 1 - this server is MASTER
master server is  : 192.168.0.110 management
slave server is   : 192.168.0.100 management
interface         : eth0  physical
timestamp         : 02/17/2014-12:02:23
service address   : 192.168.0.120  controlled by kccvoip SIP-HA

 ACTIVE SERVICES ON THIS SERVER - this server is MASTER

    - NOW IN NORMAL STATE -


 SIP-HA will replicate local files to 192.168.0.100
```

CLIENTS COMMENTS - THE TRUTH

- **GOOD POINTS**
  - simple, low maintenance and low cost
  - servers remain active/active
  - all features work as described
  - works with DUNDI
  - works with PBX trunks through gateways
  - agents remained logged-in through failover
  - handles ITSP trunks AND softphones without intervention
  - servers remain active/active (no surprises during an outage)
  - works with Cisco CUCM and CME trunks
  - maintains CDR and reporting data
  - maintains database and other software details on servers
  - ip phones and most softphones failover automatically
  - can be customized for SCCP and support of Cisco phones etc
  - works on VM also (and Oracle virtual box etc)
  - server connectivity uses ssh and encrypted keys

- **NOT SO GOOD POINTS**
  - can not be used on clusters as the site requirements may grow
  - only supports load balancing if done in 'server MASTER/SLAVE pairs'
  - good Linux and SIP server knowledge required for any detailed customization of scripts (it is possible to overwrite files if incorrectly configured)
  - wallboard and reporting web sessions need to be restarted on failover
  - some of the free softphones need to be restarted on failover
  - FXS FAX lines need careful design & planning
  - PRI and ISDN connections have to be moved to gateways in order to fully automate failover of TDM telco circuits

ALTERNATIVES FOR SIP SERVER HIGH AVAILABILITY

DNS SRV can be used for load-balancing and/or failover for SIP servers - BUT does not provide any failover for ITSP trunks and any OUTBOUND failovers.

MULTIPLE SERVER – CM GROUP can support ip phones that are able to have multiple servers configured to enable the ip phones to failover to the next active server – BUT does not provide any failover for ITSP trunks and any OUTBOUND failovers unless using a high-end call server such as Cisco CUCM etc = BIG MONEY

CLUSTER and SBC – multiple SBC and/or SIP proxy servers connecting to multiple SIP servers and full load balancing will provide all the features required – BUT requires much more hardware and maintenance = BIG MONEY

LINUX VRRP or keepalived etc . - will provide a failover of the virtual ip address – BUT does not monitor the SIP server applications and synchronise configurations etc. Several additional scripts would be needed to implement keepalived – will not failover OUTBOUND trunks and ITSP links………

MANUAL INTEVENTION – wrongly described as 'the cheapest solution of all', until there is an outage – but what are the costs to maintain the servers and keep them synchronized, how do you failover for the ip phones, softphones, telco lines AND the ITSP trunks…. how do you failover the wallboard, reporting and GUI sessions….. how do you failover the database servers and realtime data…. how do you know that 'standby server' will work when you need it…… do you have operators able to respond quickly to do the manual tasks involved

TECHNICAL SPECIFICATIONS

Most Linux/Unix servers supported (RedHat, CentOS, Ubunto, Debian, Sun/Oracle, SuSE, Fedora, AIX, BSD, SCO…)

Dependencies - scripts require rsync, nmap, ssh, vsftp, shell running from cron. **Asterisk based PBX must have been installed to Digium specifications (directories and user rights etc.).** Mysql should be running as master/master. sshd configured for asterisk user using public key.

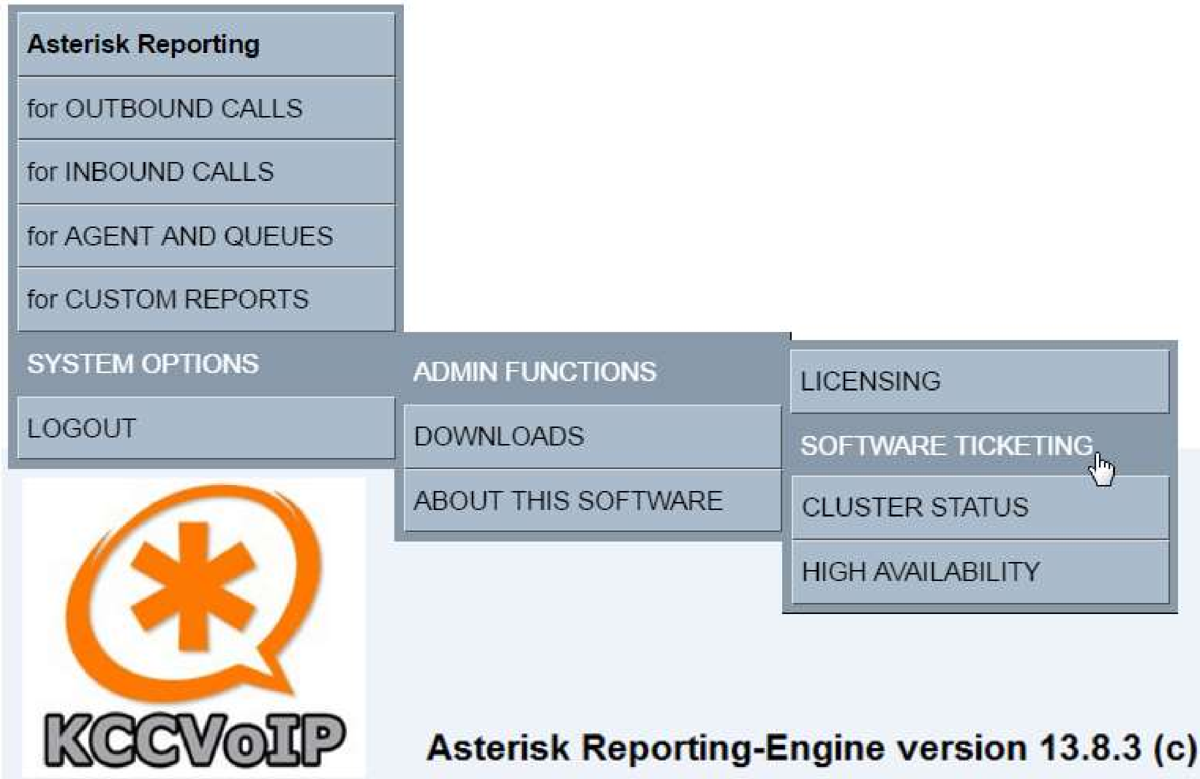
File Syncronization – provides for configuration files, PBX files, programs, web, CDR, crontab, mysql, tftp and ftp files etc. 'Normal flow' = MASTER to SLAVE - ONLY updated files are synchronized with the other server using secure ssh.

Basic static customization – user configurable – directories and files for synchronization, crontab, timeouts and failover reporting/logging. Manual testing from command line and manual failover from command line.

{this solution is NOT to be used with any other form of load balancing or failover systems and is customizable for the site requirements}

THE LOW COST SOLUTION **=** KCCVoIP SIP-HA is customized for your site nvironment with ongoing support – **contact us for a quotation** sip-ha@kccvoip.com

SEE ALSO - KCCVoIP Reporting, Custom Report Generator, VoiceRecordingManager, Wallboard Software ;



Also available from KCCVoIP – Wallboard Software, Reporting Engine and Report Generator, Digium GUI, Voice Recording Manager, Screen Capture Manager etc. All customized for your site. KCCVoIP specialise in custom design and support for Asterisk, Cisco, Avaya, Mitel, Siemens…. Integration of legacy voice systems, SIP trunk broker/unbiased advice for PRI ISDN replacement, bit level diagnostics and call center design www.kccvoip.com