



ApplianX IP Gateway User Guide

APPLIANX IP GATEWAY USER GUIDE	1
1.0 GETTING STARTED.....	5
1.1 How to use this guide	5
1.2 Prerequisites	5
1.3 LEDs	5
1.4 Setting up the gateway	5
1.5 Logging in to the web interface	5
1.6 First time use.....	7
1.7 The Setup Wizard	10
1.8 The Main Menu	11
1.9 The Overview Page	12
1.10 Networking.....	13
1.10.1 Network settings via the web interface	13
1.10.2 Network settings via USB Flash Memory.....	15
2.0 CONFIGURING THE GATEWAY	16
2.1 Gateway Configuration.....	16
2.1.1 Gateway Configuration Page Descriptions	16
2.1.2 General Configuration Information.....	16
2.1.3 Editing Trunks	19
2.1.3.1 Editing a SIP Trunk.....	20
2.1.3.2 Editing a TDM Trunk.....	20
2.1.3.2.1 Editing a TDM Trunk Protocol.....	22
2.1.4 Endpoints.....	24
2.1.5 Groups	27
2.1.5.1 Adding or Editing a Group.....	27
2.1.6 Routes	29
2.1.7 Clocking	33
2.1.8 SIP	34
2.1.9 Codecs.....	39
2.1.10 Survivability.....	40
2.1.11 Test.....	41
2.2. Backing up and Restoring Configurations	43
Figure 2-17 Backup and Restore	43
2.2.1 System configurations.....	43

Figure 2-18 Saving the File	44
Figure 2-19 Restore configuration.....	44
Figure 2-20 Backup restored.....	44
2.2.1.1 Restoring system configuration via USB port	44
2.2.2 Gateway Configurations.....	45
2.3 Factory Reset	45
3.0 ADDITIONAL INFORMATION	46
3.1 Routing Overview.....	46
3.2 X.509 Certificates	47
3.3 Creating X.509 certificates using OpenSSL	48
3.5 Secure SIP over TLS	54
3.6 Software Updates.....	57
Getting update images.....	57
Getting ax-img-tool.....	57
Validating update images	57
To apply an update image using HTTP	58
To apply an update using USB	60
3.7 Local Survivability	62
3.7.1 Overview	62
3.7.2 Enabling Survivability mode.....	63
3.7.3 Central PBX responsiveness	63
3.7.4 Switching to active survivability.....	63
3.7.5 Registrar	63
3.7.6 Configuring SIP devices	65
3.7.7 Uploading a list of users.....	65
3.7.8 Downloading Aliases	66
3.7.9 View Aliases	66
3.7.10 Overview page.....	67
3.8 DDI Barring	69
3.9 DNS Caching	70
4.0 DIAGNOSTICS.....	71
4.1 Remote Logging.....	71
4.1 Diagnostic Log	72
5.0 TROUBLESHOOTING	73
5.1 Logging into the remote interface	73
5.1.1 I can't get access to the AplianX Gateway Web Interface	73
5.1.2 I log on but the overview screen has errors at the top	73
5.1.3 I get a warning saying that the gateway cannot.....	73

5.2 Making Calls through the Gateway	73
5.2.1 I can't make a call from the TDM side of the Gateway to an IP client.....	73
5.3 Configuring the Gateway.....	75
5.3.1 I have made changes to the configuration but they don't seem to have any effect.....	75
5.3.1 I used the wizard to create an initial configuration but I have an error saying that there is no active configuration.	75
6.0 GLOSSARY	76

1.0 Getting Started

1.1 How to use this guide

The ApplianX gateway interface has been designed to be intuitive. However we still recommend that new users read sections 1-3 of this guide before trying to set up a gateway for the first time. Sections 2 and 3 are a reference for those that have used the gateway before while sections 4 and 5, Diagnostics and Troubleshooting, should only be needed if problems have been encountered.

1.2 Prerequisites

The ApplianX gateway is configured via a Web Interface. Therefore a device with a web browser will be needed to connect to the ApplianX. Also any networking cables and switches needed to allow this connection will be needed. Note that the network ports of the IP Gateway need to be connected to Ethernet Switches and not Hubs.

1.3 LEDs

There are a number of LEDs on the front of the ApplianX that are there to help during the installation and running of the ApplianX.

- Halted – This red LED indicates a serious error. If this has occurred in any circumstance other than restarting or shutting down the ApplianX then a serious error has occurred and a restart of the unit will be required.
- Error – This red LED indicates that the ApplianX has an error condition that should be resolved. Log into the ApplianX via the web interface to identify the nature of the problem.
- Activity – This blue LED will flash when the ApplianX is starting up and also when the ApplianX is processing calls.
- Ready – This Green LED is lit when the ApplianX application is running.
- Startup/Initialising – This Yellow LED indicates that the ApplianX is starting. Note that user interaction may be needed via the web interface to complete startup.

1.4 Setting up the gateway

There are a number of steps that need to be carried out before the Gateway can be used to service calls. The **Setup Wizard** is designed to create a basic configuration.

1.5 Logging in to the web interface

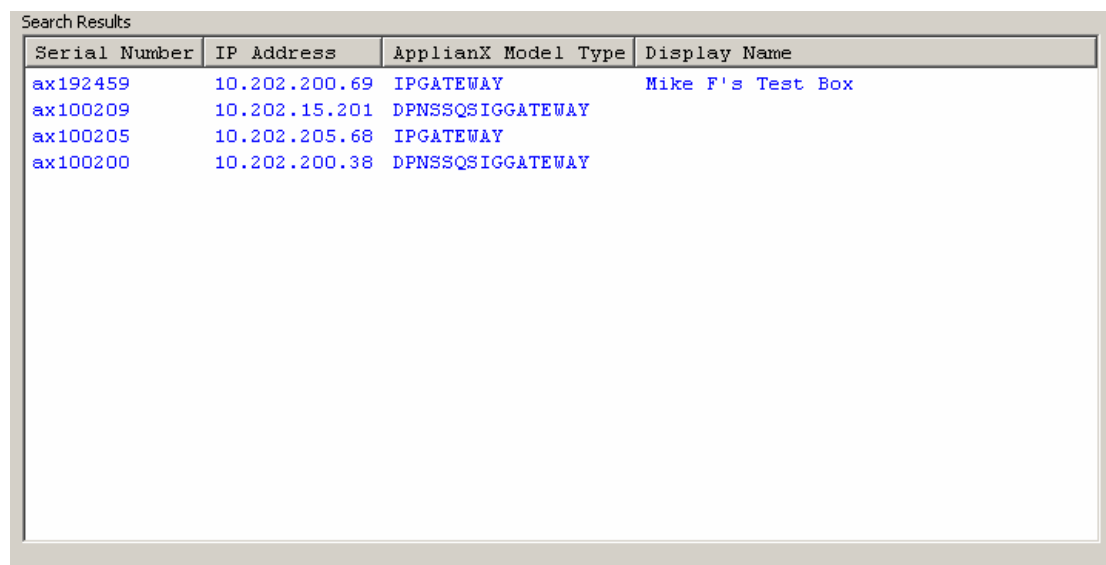
The ApplianX Gateway should be powered up with LAN cables connecting the VoIP traffic port and the Admin port to the network.

For versions of software 2.1.0 and later the ApplianX administrative interface will have the static IP address 192.168.1.100. For earlier versions please consult earlier versions of the documentation.

Connect a PC directly to the ApplianX admin port with an Ethernet cable. Set the PC to have the static IP address 192.168.1.1 with a net mask of 255.255.255.0. By typing 192.168.1.100 into the web browser the ApplianX administration interface should be accessible. Change the static IP address to something suitable for the network it will be used in. Once set up in a network the ApplianX will be accessible via the ApplianX Search Tool.

The *ApplianX Search Tool* is available from <http://www.aculab.com/downloads/>. Then navigating to the Applianx Tools & downloads section.

Once installed, start *ApplianX Search Tool* from the *start* menu. The *ApplianX Search Tool* will search the local network for ApplianX products and report the IP address of any products it finds (see Figure 1-0 below).



Serial Number	IP Address	ApplianX Model Type	Display Name
ax192459	10.202.200.69	IPGATEWAY	Mike F's Test Box
ax100209	10.202.15.201	DPNSSQSIGGATEWAY	
ax100205	10.202.205.68	IPGATEWAY	
ax100200	10.202.200.38	DPNSSQSIGGATEWAY	

Figure 1-0 The ApplianX Search Tool

A context menu is presented when right clicking on a listed Applianx device. From the context menu it is possible to spawn your default browser to automatically navigate to the Applianx administration web interface login page. Please see the troubleshooting section if you cannot gain access to the ApplianX web interface.

1.6 First time use

The Gateway Management Interface uses HTTPS to protect your session. The default certificate will trigger a security warning on modern browsers. Although the browsers will indicate that it doesn't trust the source of the certificate, the session will be encrypted. It is possible to replace the supplied certificate with your own.

NOTE: HTTPS is subject to export controls and may not be available in your territory.

In Internet Explorer 7 and 8:

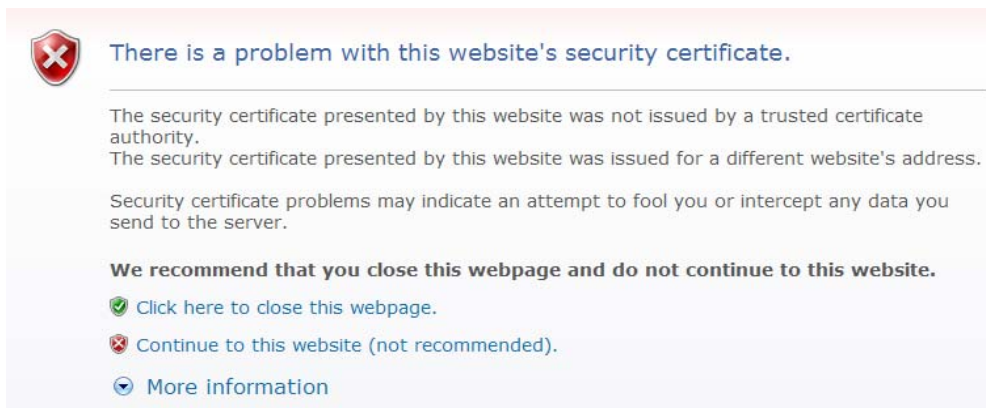


Figure 1-2 Internet Explorer 7 and 8 security warning

Click on "Continue to this website" to proceed.

In Firefox 3.5:

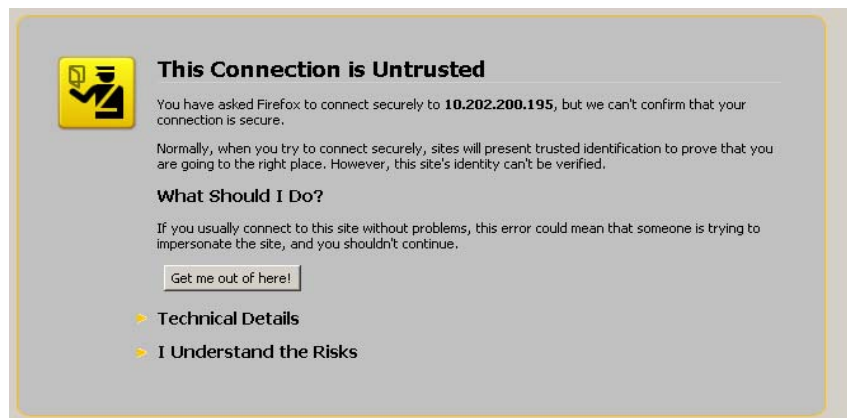


Figure 1-3 Firefox 3.5 security warning

Expand the "I Understand the Risks" section and click "Add Exception...".

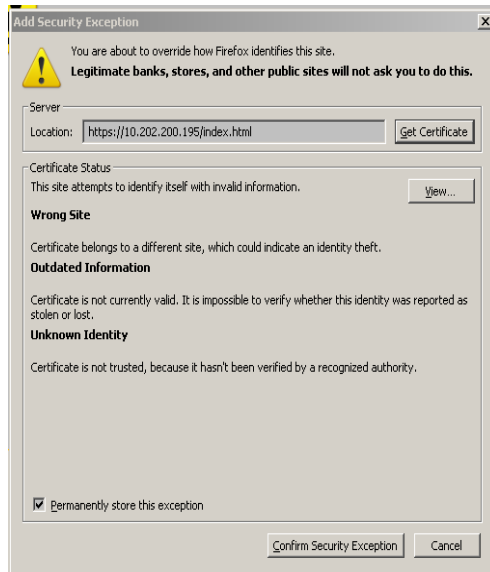


Figure 1-4 Firefox 3.5 security exception dialog

Click on “Confirm Security Exception” to proceed. You can uncheck the “Permanently store this exception” checkbox if you plan to replace the supplied certificate with your own.

In Chrome:

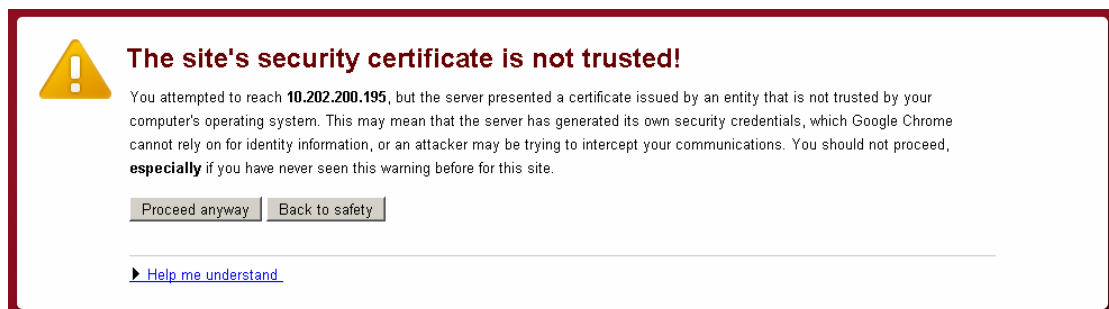


Figure 1-5 Chrome security warning

Click on “Proceed anyway” to continue.

In Safari:

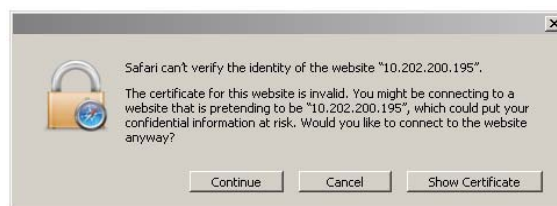


Figure 1-6 Safari security warning

Click on “Continue” to proceed.

On first use the Gateway Management Interface will display the page as in **Figure 1-7 Configuring initial administrative user**. The user is required to provide a user name and password for an administrative user for the Gateway.

Enter a user name, password and confirm the password. The user name and password cannot be left blank. Click **Submit** to create the account and login.

Note: It is important to remember the user name and password that you configure. If you forget, you will need to perform a factory reset to gain access to the ApplianX. See the Factory Reset section below.

The screenshot shows the ApplianX IP Gateway interface. At the top left is the ApplianX logo, and at the top right is the text "You are not logged in". Below the logo is the title "IP Gateway". On the left side, there is a "Log in" link and a link to "Register your product warranty". The main content area is titled "Configure administrative user" and contains a welcome message: "Welcome to your ApplianX system! In order to use the system you must configure an administrative user." Below this message are three input fields: "User Name", "Password", and "Confirm password". A red "Submit" button is located below the "Confirm password" field. At the bottom of the page, there is a footer with the text: "©2007 ApplianX | Terms & Conditions | Privacy Policy | Contact Details | www.aculab.com".

Figure 1-7 Configuring initial administrative user

1.7 The Setup Wizard

The Setup Wizard is accessed from the Gateway menu. It is also automatically invoked the first time the Gateway is used. The setup wizard allows the creation of a basic configuration, prompting for the most commonly required and important configuration details. Default values or reasonable values are used wherever possible.

At any time, **Cancel** can be selected to return to the main Gateway Overview page. No configuration is stored until the user selects **Apply** on the final wizard page.

A wizard-created new configuration will have:

- 3 Endpoints:
 - Default SIP Endpoint (Will have no associated IP address initially)
 - ApplianX IP Gateway Self (will match calls from the ApplianX to itself, e.g. as sometimes made during SIP transfer)
 - Proxy (Will have an address if given in the wizard)
- 3 Groups:
 - “TDM Trunks” containing all the TDM trunks
 - “Default Incoming SIP group”
 - “Proxy group”
- No routing rules defined and the “Use same rules for all groups” option turned on.
- The “Accept calls from unknown endpoints” option will be turned off.
- TDM clocking configured to use any good available TDM trunk or otherwise to fallback to local clocking
- SIP listening on UDP and TCP ports 5060
- using UDP for outgoing calls
- enabling DTMF as RFC2833
- G.711 a-law and G.711 mu-law codecs enabled
- TLS and SRTP disabled

At the end of the wizard your web browser will be redirected to the “Edit Configurations” page. Here you have a list of all configurations that have been setup on the ApplianX. Note that if this is the first time a configuration has been created then the new configuration will be listed in the “Available configurations” list. The configuration must be activated to bring it into use. This is done by selecting **Use** for the required configuration.

1.8 The Main Menu

On the left of the screen at all times, apart from when the wizard is running, you will be able to access all the configuration and status pages.

- **Status**
 - **Overview** – A page with some basic gateway call counts and a list of actions required of the gateway administrator.
 - **Alarms** – This page will display any Layer 1 or Layer 2 alarms on the TDM trunks. It will also allow the masking of these alarms.
 - **Calls** – A graphical display of all the call activity on the ApplianX Gateway.
 - **Call Log** - A recent history of calls that the gateway has attempted to route. This page can be very useful for diagnosing issues during the set up phase for the gateway.
 - **Trunk Status** – This has detailed information on the SIP and TDM trunks

- **System Configuration**
 - **Global Configuration** – This allows the box to be named
 - **System Time** – This allows the setting of the clock to local time and NTP configuration.
 - **HTTPS Configuration** – This allows you to view information about the HTTPS certificate currently in use and replace it if required.
 - **SIP TLS Configuration** – This allows you to configure TLS certificates.
 - **Software Update** – From this page a check can be made for software updates. See section 1.8 for more information.
 - **System Users** – This allows the addition of new administrators to the ApplianX and the setting of their privileges.
 - **Backup and Restore** – This allows configurations to be saved and restored to the ApplianX.
 - **Networking** – This allows the user to choose static IP addresses or DHCP mode
 - **Static DNS** – Manual input of static DNS address that avoids DNS request.
 - **DNS status** – All specified DNS server statuses. DNS cache contents and the ability to flush the DNS cache.
 - **SNMP** – This allows the configuration of the SNMP settings. From here you can enter the IP address of the host you wish to send traps to and enable them. Also here you can turn on the traps for the disconnecting of the Ethernet ports. Similar options are available for the TDM ports through the TDM configuration options.
 - **Setup Wizard** – This allows the setup wizard to be run to create a skeleton configuration
 - **SIP Credentials** – This allows the configuration of details to allow the gateway to respond appropriately when challenged for authorisation information.

- **Gateway Configuration**
 - **Alias Registrar** – Query and view SIP user aliases that are currently registered.

- **Manage Aliases** – Upload, backup and clear registered SIP users.
 - **DDI Barring** – Upload, backup and clear barred DDIs.
 - **Edit Configurations** – This takes you to the main configuration overview where different gateway configurations can be selected and edited. All aspects of the gateway from Codecs and SIP set up to routing rules and groups can be edited here.
 - **Interoperability** – Configuration of the system SIP stack. For use under Aculab Technical Support supervision.
 - **Cause Mappings** – Here the clearing causes between SIP, QSIG and DPNSS can be changed from their default values.
- **Diagnostics**
 - **Remote Logging** – This allows the administrator to point the syslog output from the ApplianX to an external syslog client or ApplianX Trace Tool. This is for advanced users and support teams.
 - **Network Diagnostics** – Lets you ping from the Admin or Signalling Ethernet ports to let you verify that the network interfaces can access other terminals on the network.
 - **Watchdog Status** – This reveals the status of the “watchdogs” running on the ApplianX. They are here to look for any elements that have failed or are reporting problems. This is for advanced users and support teams.
 - **Restart** – This is used to “reboot” the ApplianX. Note that rebooting will cause all contact to be lost with the ApplianX through the user interface.
 - **Diagnostic Log** – This provides a high level overview of gateway process and can be used for debugging purposes.
 - **Endpoint Status** – This page will list the status of those IP endpoints that have been configured for monitoring
 - **About** – This gives build information on the ApplianX.
 - **Hardware** – This displays the version and status of the hardware used in the ApplianX.
- **Account**
 - **Log Out** – This allows the current user to log out of the ApplianX administration screens.
 - **Change Password** – This allows the current administration user to change their password.

1.9 The Overview Page

The overview page gives some basic stats for the gateway such as total incoming and outgoing call counts. At the bottom of this page will be a list of actions that the gateway is flagging for the administrator.

Overview	
Status	Running
Incoming calls	0
Outgoing calls	0
Unroutable calls	0
Clock source	Local
Active configuration	My configuration

Required Actions	
Error	Trunk: (Trunk 1) - Layer 1 Errors (Possibly cabling, clocking, layer1 configuration (e.g. CRC on/off) or wrong firmware)
Error	Trunk: (Trunk 2) - Layer 1 Errors (Possibly cabling, clocking, layer1 configuration (e.g. CRC on/off) or wrong firmware)
Error	Trunk: (Trunk 3) - Layer 1 Errors (Possibly cabling, clocking, layer1 configuration (e.g. CRC on/off) or wrong firmware)
Error	Trunk: (Trunk 4) - Layer 1 Errors (Possibly cabling, clocking, layer1 configuration (e.g. CRC on/off) or wrong firmware)

Figure 1-8 The Overview page

As you can see in the above example the Gateway is telling us that we have Layer 1 errors on all trunks. In this case it is because we have not connected any TDM trunks to the gateway yet.

The overview page also tells you which one of your gateway configurations is currently in use.

1.10 Networking

The compact chassis requires 2 IP addresses. By default the admin port is set to a static IP address of 192.168.1.100, the signalling interface is set to 10.202.100.4. The gateway should only actually be deployed using static IP addresses. Note that if DHCP is selected and there is no DHCP server on the network the Appliance will use Zeroconf technologies to get IP addresses and to provide access to the unit.

There are two methods for changing the IP settings.

- 1) Via the web interface
- 2) Via a USB flash memory device.

1.10.1 Network settings via the web interface

The IP addresses can be manually be set to static addresses by selecting **Networking** from the menu on the left. Here the 2 interfaces can be selected to be set up via DHCP or can be set to be static IP addresses as shown below in figure1-11.

Networking

>> Administration interface

DHCP	<input type="checkbox"/>
IP address	10.202.100.92
Netmask	255.255.0.0
Gateway	

>> Signalling and media interface

DHCP	<input checked="" type="checkbox"/>
------	-------------------------------------

>> Name resolution settings

Name server source ?	<input type="radio"/> Enter settings manually <input checked="" type="radio"/> DHCP on the signalling interface
Name server ?	
Name server ?	
Search domains ?	

Save Configuration

Cancel Changes

Figure 1-11 Networking

Note that changing the IP addresses will affect the box and its internal and external communications. When changing the administration port and saving the configuration you will immediately lose connection between the web browser and the ApplianX. The browser should be manually redirected to the new IP address. Also when changing the signalling address the internal communications will need to be re-established. This should take around a couple of minutes to resolve. You may see the message below, in figure 1-12, on the Overview page.

Required Actions

Warning	The Gateway is waiting for hardware to become available. This is to be expected when the system has just been started or when the network configuration has changed. The hardware should be detected within a couple of minutes.
---------	--

Figure 1-12 Warning

Finally the options for name resolution can be setup on this page. Servers may be manually entered or DHCP on the signalling interface may be selected.

1.10.2 Network settings via USB Flash Memory

On a USB flash disk device, at root level, create a directory named `aplianx_net`. This directory must contain two text files. One file, named `admin`, will contain IP address information for the Admin port that allows web browser access to the Aplianx. A second file, named `signalling`, will contain the IP address information for the Signalling port that allows SIP calls to be made to and from an Aplianx. The files `admin` and `signalling` must not have file extensions.

To specify static IP address information the files may contain the following format;

```
[config]
ip = 10.202.165.169
netmask = 255.255.0.0
gateway = 10.202.100.254
```

To specify IP addresses to be set via DHCP the files may contain the following format;

```
[config]
dhcp = 1
```

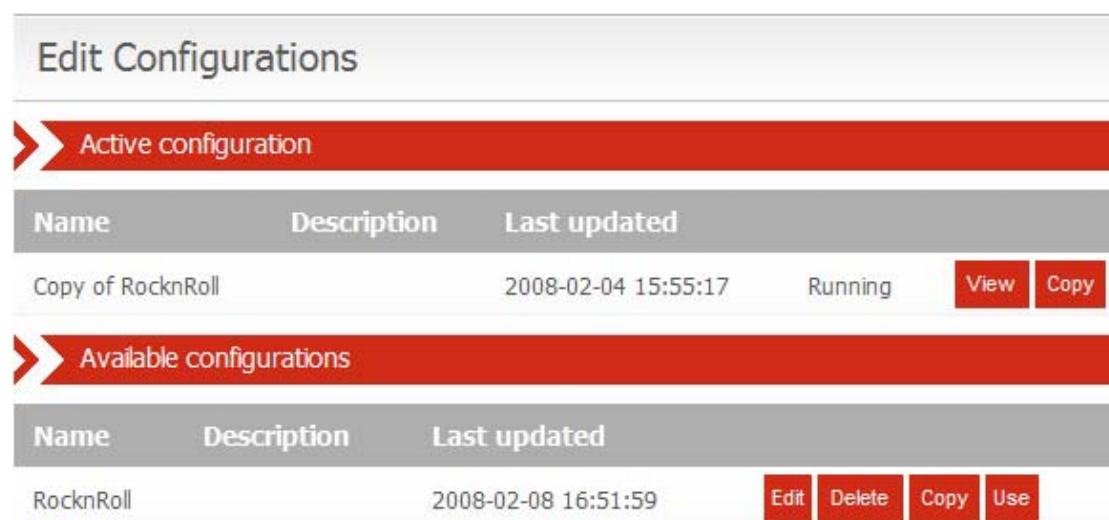
At boot up, if an Aplianx detects a USB flash disk device, then it will search the USB disk for the files mentioned above. If found the information inside them will be used to set the IP address information for the Aplianx.

Note: Using this method will mean it takes a few minutes longer for the Aplianx to come into service.

2.0 Configuring the Gateway

2.1 Gateway Configuration

All Gateway configurations are managed from the **Figure 2-1 Edit configurations page** (see Figure 2-1 below). The currently active configuration is listed first. This may not be directly edited, but may be examined by selecting **View**. To modify the active configuration, it is first necessary to click **Copy** next to the active configuration entry. When you are happy with edits made to a new or copied configuration you can select this to be the active configuration by selecting the **Use** button on the right of the configuration.



The screenshot shows a web interface titled "Edit Configurations". It is divided into two main sections: "Active configuration" and "Available configurations".

Active configuration section:

Name	Description	Last updated	Running	View	Copy
Copy of RocknRoll		2008-02-04 15:55:17	Running	View	Copy

Available configurations section:

Name	Description	Last updated	Edit	Delete	Copy	Use
RocknRoll		2008-02-08 16:51:59	Edit	Delete	Copy	Use

Figure 2-1 Edit configurations page

2.1.1 Gateway Configuration Page Descriptions

Configuration information is presented as a set of inter-related tabbed pages, some of which lead to further more detailed pages. At any time, selecting **Cancel Changes** will cause all changes to be discarded. Selecting **Save Configuration** will save the changes made. In either case, the main Edit Configurations page is redisplayed.

2.1.2 General Configuration Information

This page, shown in Figure 2-2, enables the setting of a configuration name, description, and other general options. A configuration may be renamed by changing the **Configuration name**. The **Configuration description** allows any notes or important information to be stored along with a configuration.

In the **Compatibility** section, when enabled the **Gateway Progress** option will cause the Appliance to gateway the Euro ISDN progress indicator message. On Appliance 2.3 this is by default turned on. Previous versions of Appliance did not gateway the progress indicator. So, this configuration check box has been provided to allow the 2.3 Appliance to work like the 2.2 Appliance with regards to this feature.

The **Network Unique ApplianX IP Gateway Call Identity** section relates to how this ApplianX behaves when proposing DPNSS Route Optimisation or QSIG Path Replacement. PBXs and other telecom network devices, such as the ApplianX, will insert a call id from a configured range in a proposal message. This is used to locate the associated call to replace when the far-end returns a new call. Hence, each such device must be able to determine whether an incoming call is a response to one of its own proposals.

The **Fax Configuration** section controls how the ApplianX handles incoming calls from a fax machine. If either of the two fax detection modes is enabled, after the call is connected the ApplianX will listen for fax CNG tone for the configured time period. The time is typically limited to avoid false positives during a call. If CNG tone is detected, any gateway active echo cancellation will be disabled. If either call leg is over SIP and “Detect fax and use SIP G.711 if necessary” option is selected, SDP renegotiation will also be performed to ensure only G.711 codecs are selected. If the “Detect fax and use SIP T.38 if necessary” option is selected, the SIP call leg will be renegotiated to use T.38. If the SIP endpoint rejects the T.38 renegotiation, then the gateway will attempt to negotiate G.711 as a fallback.

Fax is not supported over SIP to SIP calls.

The “Maximum simultaneous T.38 jobs” can be used to limit the maximum number of T.38 calls the gateway can process.

Incoming T.38 SIP calls, or SIP calls that are renegotiated to T.38 by the SIP endpoint, will always be accepted (as long as the maximum simultaneous T.38 jobs limit is not exceeded).

The fax call detection type maybe overridden on a per-routing rule basis.

Editing: My configuration

General Trunks Endpoints Groups Routes Clocking SIP Codecs Survivability Test

General Configuration Information

Configuration name: My configuration

Configuration description: [Empty text area]

Compatibility

Gateway PROGRESS:

Network Unique Appliance IP Gateway Call Identity

First callid for this Appliance IP Gateway: 8000

Range of callids for this Appliance IP Gateway: 100

Fax Configuration

Fax call detection:

- Fax detection disabled
- Detect fax and use SIP G.711 if necessary
- Detect fax and use SIP T.38 if necessary

Time in which to listen for an incoming fax: 10

Maximum simultaneous T.38 jobs: 8

Save Changes Save and Return Cancel Changes

Figure 2-2 General

2.1.3 Editing Trunks

All available trunks are listed on the Trunks page as in **Figure 2- 3 Trunks page**. SIP Trunks and TDM Trunks are listed separately. Settings for an individual trunk can be changed by selecting **Edit** next to the trunk.

Editing: My configuration

General Trunks Endpoints Groups Routes Clocking SIP Codecs Survivability Test

>> SIP trunks

Name	Description	Type	Group	
Trunk 5		SIP	No group	Edit

>> TDM trunks

Name	Description	Type	Group	
Trunk 1		TDM[DPNSS]	TDM trunks	Edit
Trunk 2		TDM[DASS]	TDM trunks	Edit
Trunk 3		TDM[QSIG]	TDM trunks	Edit
Trunk 4		TDM[ETS300]	TDM trunks	Edit

Save Changes Save and Return Cancel Changes

Figure 2- 3 Trunks page

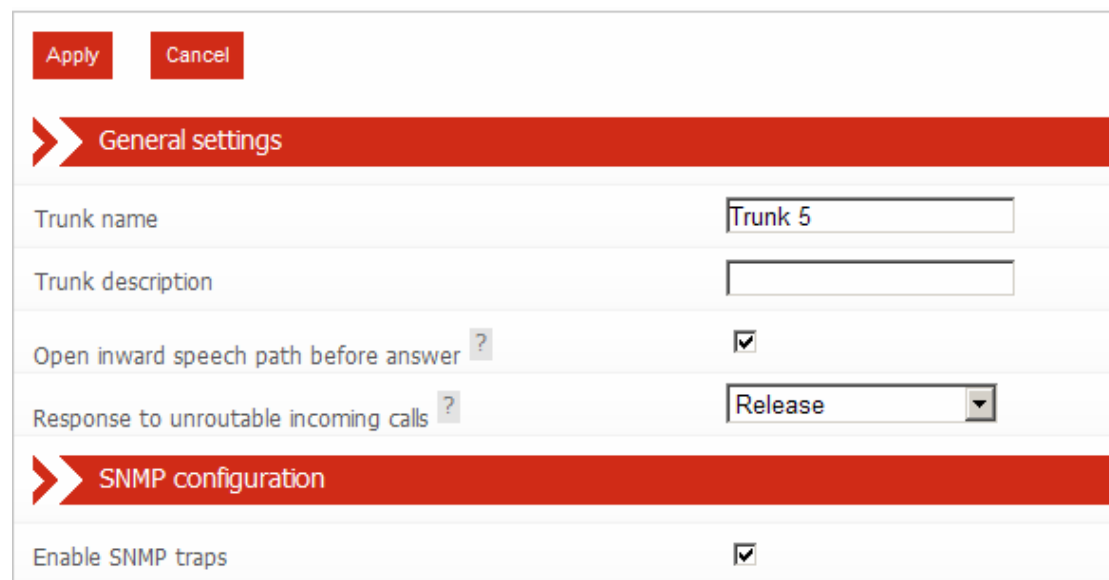
2.1.3.1 Editing a SIP Trunk

Each Trunk requires a name distinct from all other Trunks. Changing the name of a Trunk causes all references to the Trunk to also change. Trunk description should be something sensible that divulges more detail that could not be conveyed in the trunk name.

Often it is required to open up speech paths before a call is connected to allow the signalling of in band information. A tick box to enable this has been provided.

If a call comes in on a trunk that is not routable it is possible to select how the gateway deals with that call. Whether it is released, connected with a tone, send a SIP 180 with a tone, a SIP 183 with a tone or no response, as per the options in the drop down selection item.

To enable SNMP traps for the signalling network Interfaces, tick the appropriately labelled tick box. The SIP trunk SNMP traps sent at present relate to Local Office Survivability see [3.7 Local Survivability](#). SNMP traps can be expected when Local Office Survivability has been activated or deactivated.



The screenshot shows the 'Edit SIP Trunk Page' interface. At the top, there are 'Apply' and 'Cancel' buttons. Below them is a red header bar with a white arrow pointing right and the text 'General settings'. The main configuration area contains the following fields:

Trunk name	<input type="text" value="Trunk 5"/>
Trunk description	<input type="text"/>
Open inward speech path before answer ?	<input checked="" type="checkbox"/>
Response to unroutable incoming calls ?	<input type="text" value="Release"/>

Below this is another red header bar with a white arrow pointing right and the text 'SNMP configuration'. Underneath, there is a single row:

Enable SNMP traps	<input checked="" type="checkbox"/>
-------------------	-------------------------------------

Figure 2- 4 Edit SIP Trunk Page

2.1.3.2 Editing a TDM Trunk

Each Trunk requires a name distinct from all other Trunks. Changing the name of a Trunk causes all references to the Trunk to also change. Trunk description should be something sensible that divulges more detail that could not be conveyed in the trunk name.

The Group for this Trunk can be selected from the list of Trunk Groups.

NOTE: Mixing different types of Trunk in the same Trunk Group is not supported. All Trunks in a Group must be of the same type.

Often it is required to open up speech paths before a call is connected to allow the signalling of in band information. A tick box to enable this has been provided.

In contrast to a SIP trunk, it is possible to block a TDM trunk from participating in call activity.

The strategy for allocating outgoing timeslots can be selected from a list of options.

The minimum digit count allows the gateway to attempt routing when a certain number of digits have arrived.

The inter-digit timeout in milliseconds can be specified. This is the time that the gateway waits for another digit before deciding it has got them all. For CAS protocols this defaults to 5 seconds.

The strategy for dealing with calls that cannot be routed can be selected here also. For example, connect with a tone; alert with a tone; progress with a tone; setup with a tone or no response.

The currently configured protocol is displayed. This can be changed or configured by selecting **Edit**. In particular, supplementary features are enabled and disabled through this edit option. Finally the SNMP trap can be enabled for this trunk.

The screenshot shows the 'Edit TDM Trunk' configuration page. At the top, there are 'Apply' and 'Cancel' buttons. Below is a red header for 'General settings'. The configuration fields are as follows:

Trunk name	Trunk 1
Trunk description	
Open inward speech path before answer ?	<input checked="" type="checkbox"/>
Routing group	TDM trunks
Block trunk from call activity ?	No
Outgoing timeslot allocation strategy ?	Highest available
Minimum digit count ?	1
Interdigit timeout (milliseconds) ?	5000
Interdigit timeout for virtual calls (milliseconds) ?	1000
Send sending complete on outgoing calls ?	<input checked="" type="checkbox"/>
Send overlap digits on outgoing calls ?	<input checked="" type="checkbox"/>
Response to unroutable incoming calls ?	Release

Below the general settings is a red header for 'SNMP configuration' with one option:

Enable SNMP traps	<input checked="" type="checkbox"/>
-------------------	-------------------------------------

At the bottom is a red header for 'Protocol configuration' with one option:

Protocol	DPNSS	Edit Change
----------	-------	---------------------------

Figure 2- 5 Edit TDM Trunk page

2.1.3.2.1 Editing a TDM Trunk Protocol

In contrast to a SIP trunk, each TDM Trunk also requires a trunk protocol. The selected protocol must be chosen to be compatible with the remote equipment connected to the trunk. The current protocol can be set or modified by selecting **Change**. Protocol configuration options are also available. All settings and options for the trunk protocol are specific to the user's installation. You should seek the advice of your service provider or switch maintenance team for advice on the protocol selection and settings to be used.

DPNSS

General settings

Impedence	120 Ohms (default) ▾
CRC enabled [?]	<input type="checkbox"/>
Master/Slave configuration	AX ▾

Basic features

Display direction [?]	Send and receive ▾
Allow incoming data calls [?]	<input checked="" type="checkbox"/>
Loop avoidance mapping [?]	<input type="radio"/> Disabled <input checked="" type="radio"/> Transparent <input type="radio"/> Transit
Global transit limit [?]	25
Insert loop avoidance in outgoing calls [?]	<input type="checkbox"/>
Do-not-disturb mapping [?]	<input checked="" type="checkbox"/>
Method for generating CLC [?]	<input type="radio"/> Use a fixed value <input checked="" type="radio"/> Map from the other call leg (default) <input type="radio"/> Map from the calling name
CLC when map is not possible [?]	CLC-DEC ▾
Override CLC when OLI restricted [?]	No Override ▾
Insert Bearer Service Selection (BSS) [?]	<input checked="" type="radio"/> Disabled <input type="radio"/> Preferred <input type="radio"/> Mandatory
Call Offer Enabled [?]	<input checked="" type="checkbox"/>
Call Transfer Enabled [?]	<input checked="" type="checkbox"/>

Call Diversion Supplementary Service Support

Call Diversion Enabled [?]	<input checked="" type="checkbox"/>
Automatic Diversion Validation [?]	<input type="checkbox"/>

Figure 2-6 Editing TDM Trunk Protocol

2.1.4 Endpoints

This page lists known IP endpoints that are expected to work with this system and also provides a default endpoint definition for calls from unknown endpoints.

However, the default endpoint will only be utilised, in call routing, if the **Allow calls from unknown endpoints** option is enabled in the Routes configuration tab.

Endpoints, like trunks, can be grouped; from which calls can be routed to or from. See [2.1.5 Groups](#)

Figure 2-7 shows the default list of endpoints.

General	Trunks	Endpoints	Groups	Routes	Clocking	SIP	Codecs	Survivability	Test
Default SIP Endpoint		Default endpoint to match incoming SIP calls that don't match any other endpoint.						📄	
Appliance IP Gateway self		Certain supplementary services can result in the Appliance IP Gateway being asked to call itself. This endpoint matches those calls and allows them to be routed correctly.						📄	
Appliance IP Gateway registered users		Destination for calls to registered users.						📄	
Proxy		A SIP proxy						📄 ✖	
Add a new endpoint									
Save Configuration					Cancel Changes				

Figure 2- 7 Endpoints

If a SIP proxy IP address was provided during interaction with the Setup Wizard, then an endpoint named Proxy will be present

The Proxy endpoint is useful for inter-working with Proxies or soft PBX's.

User defined endpoints can be deleted by clicking the red cross or edited by clicking on the document icon. There is also the option to add further endpoints. Clicking **Add a new endpoint** will take you to the screen shown in figure 2.8a.

Editing: My configuration

Apply Cancel

General

Name ?

Description ?

Routing group ?

Endpoint Options

Endpoint address ?

UDP port ?

TCP port ?

Monitor this endpoint ?

Trust this endpoint ?

During call transfers, allow sending of 'INVITE with Replaces' ?

During call transfers, allow sending of 'REFER with Replaces' ?

During call transfers, allow sending of 'REFER' ?

This endpoint is an Aculab ApplianX IP Gateway ?

This endpoint is the central PBX ?

Figure 2- 8a Configuring an endpoint

Name field should be a unique name used to identify the endpoint. The **Description** field is a description to associate with the said endpoint.

Routing group is the routing group that the endpoint will belong to; this can be left unset and revisited once groups have been created. Alternatively, endpoints can be assigned to a group during group configuration, see [2.1.5 Groups](#).

The **Endpoint Address** should be set to the IP address of the endpoint you're expecting calls from and **UDP/TCP Port** fields are the ports to listen on for SIP calls from the aforementioned IP address.

Enabling **Monitor this endpoint** will result in the Applianx gateway periodically sending an OPTIONS message to the endpoint. If the endpoint does not reply to the SIP OPTIONS then the gateway will consider this endpoint as out of service as a result the endpoint will not be routed to.

Trust this endpoint instructs the gateway to pass CLI information to the endpoint even if the CLI is passed with the presentation restricted flag set.

Concerning call transfers, enabling **allow sending of 'INVITE with Replaces'** and/or **allow sending of 'REFER with Replaces'** will mean that a Replaces header will be present in INVITE and/or REFER SIP messages.

Concerning call transfers, enabling **allow sending of 'REFER'** will mean that a REFER SIP message can be sent during call transfers.

This endpoint is an Aculab Applianx IP Gateway can be enabled to allow additional support over a SIP trunk for DPNSS Route Optimisation or QSIG Path Replacement. **This endpoint is the central PBX** indicates that an endpoint can be used as the central PBX when Local office survivability mode (see [2.1.10 Survivability](#)) is employed. Only one endpoint can be indicated as the central PBX at any time. Enabling this option, when not using survivability mode, has no effect on the behaviour of your Applianx.

Enabling the **Register a user name with this endpoint** option causes the ApplianX to register a user name at the **Endpoint Address** by sending a SIP REGISTER message to the endpoint whose address is **Endpoint Address**. Enabling this option will make the following hidden options visible as shown in Figure 2-8b. The **User name** field is the user name part of the URL to be registered. The **Contact address** field is the contact address of the user name to be registered (If this field is left empty the address of the gateway shall be used as the contact address).



Registration	
Register a user name with this endpoint	<input checked="" type="checkbox"/>
User name ?	<input type="text" value="applianx"/>
Contact address ?	<input type="text"/>

Figure 2- 8b Endpoint Registration Options

The **T.38 Fax Gateway Configuration** options have been provided to aid interoperability with endpoints boasting support for T.38 Fax Gateway functionality.

Allow T.38 on this endpoint will enable T.38 on this endpoint. So, upon fax tone detection a T.38 re-INVITE will be sent to this endpoint.

Disabling **Allow ECM negotiation for this endpoint** will prevent ECM (error correction mode) from being negotiated with this endpoint.

The **Redundancy Level** specifies the number of T.38 fax redundancy packets that are sent.

Re-INVITE delay is the length of time in milliseconds that the ApplianX will wait before sending a re-INVITE to a T.38 endpoint when a CNG tone is detected.



T.38 Fax Gateway Configuration	
Allow T.38 on this endpoint ?	<input checked="" type="checkbox"/>
Allow ECM negotiation for this endpoint ?	<input checked="" type="checkbox"/>
Allow V.17 Modem to be negotiated for this endpoint ?	<input checked="" type="checkbox"/>
Redundancy level ?	<input type="text" value="2"/>
Re-INVITE delay ?	<input type="text" value="500"/>

Figure 2- 8c T.38 Fax Gateway Configuration

2.1.5 Groups

This page lists all the defined groups. A group is a collection of Trunks or endpoints that are grouped together for the purpose of routing.

To change an existing Group click **Edit**. Click **Add a new group** to create a new Group. To delete an existing Group click **Delete**.

Editing: My configuration

General	Trunks	Endpoints	Groups	Routes	Clocking	SIP	Codecs	Survivability	Test		
Name^		Description									
ApplianX IP Gateway group		Default group to contain the ApplianX IP Gateway itself. Certain supplementary services can result in the ApplianX IP Gateway being asked to call itself. This group should be assigned rules that route these calls successfully.								Edit	Delete
Default Incoming SIP group		Group for incoming SIP calls								Edit	Delete
Proxy group		Group containing the SIP proxy								Edit	Delete
Registrar		Group owning the registrar. Rules routing TO this group will attempt to route to registered users.								View	
TDM trunks		Default group for TDM trunks								Edit	Delete
Add a new group											

Figure 2- 9 Groups

2.1.5.1 Adding or Editing a Group

Each Group requires a name distinct from all other Groups. Changing the name of a Group causes all references to the Group to also change. A free format description for the Group can be entered. The trunks/endpoints assigned to this Group are listed. The association of a Trunk with a Group is specified on the individual Trunk/Endpoint page. Finally, as shown in figure 2-10 there is an option to select the method by which the next trunk/endpoint is chosen. The options are round robin or first in the list.

Editing: My configuration

>> General settings

Routing Group Name

Routing Group Description

Endpoint selection method

>> Endpoints assigned to this Routing Group

Endpoint Name	Endpoint Description	Endpoint Type	
Trunk 3		TDM	▼ ▼ ✕
Trunk 4		TDM	▲ ▲ ▼ ▼ ✕
Trunk 1		TDM	▲ ▲ ▼ ▼ ✕
Trunk 2		TDM	▲ ▲ ✕

Figure 2-10 Edit Group

2.1.6 Routes

This page allows modification of the routes assigned to a group. The drop down box at the top allows you to select the group that you wish to route from, see figure 2 – 11 Editing Routes.

Editing: My configuration

General Trunks Endpoints Groups Routes Clocking SIP Codecs Survivability Test

Routing Options

Use the same rules for all groups

Allow calls from unknown endpoints

Routing Rules

Select the group for which you want to configure the routing TDM trunks

Name	DDI/DID criteria	DDI/DID man.	CLI/ANI criteria	CLI/ANI man.	Destination
Registrar	%	%	%	%	Registrar

Add new rule Use these rules for all groups

Figure 2-11 Editing Routes

By default there exists a route to the Registrar group this should be considered a pseudo route for use by the Appliance gateway, please do not delete or edit this route.

By default the **Use the same rules for all groups** option is enabled and the **Allow calls from unknown endpoints** option is disabled.

When **Use the same rules for all groups** is enabled, routing rules that route based on the destination and/or originating address are required. These rules are applied to all routing groups. This is the easiest method to use when configuring the Appliance as it automatically deals with cases where the Appliance is diverted or transferred to itself by a SIP endpoint.

When **Use the same rules for all groups** is disabled, each routing group has its own list of routing rules that are applied to incoming calls. When SIP supplementary features are in use (e.g. diversion or transfer) this can sometimes result in the gateway being asked to call itself. In a configuration where this is likely to happen you will need to set up additional routing rules that allow these calls to be routed to the correct destination.

The **Allow calls from unknown endpoints** option controls the ability of the gateway to route calls from endpoints that it doesn't know about. When this option is enabled, calls from unknown endpoints match the Default SIP Endpoint endpoint and are routed according to the rules assigned to the group that the Default SIP Endpoint is in.

Routes can be added using the **Add** button. The **DDI Criteria** and **CLI Criteria** fields define the pattern used to match the dialled destination address and originating address. The following characters are used to define the pattern:

- % matches any sequence of digits
- ? matches any single digit

- individual digits match themselves

For example, 81% will match any number beginning with 81, whereas 8??2 will match any 4 digit number beginning with an 8 and ending with a 2.

The DDI and CLI Manipulation fields define how the destination and originating addresses will be changed. The following characters are used to define the translation:

- ? uses the next character from the incoming string
- ! deletes the next character from the incoming string
- % uses the remainder of the incoming string (any further characters in the translation string will be ignored)
- \$ deletes the remainder of the incoming string
- Any other digit is copied to the outgoing string.

For example, if the incoming Destination number is 8120 and the destination address manipulation field is set to 123!% then the destination address used for the outgoing call will be 123120.

By selecting the edit icon on the right of any routes advanced options are available for that route. Figure 2-12 shows this.

>> Codecs	
<input checked="" type="checkbox"/> Defer codecs to the global setting	
>> Transport	
<input checked="" type="checkbox"/> Defer transport to the global setting	
>> Fax	
<input checked="" type="checkbox"/> Defer fax to the global setting	
>> TDM Options	
Originating address screening ?	Transparent
Originating address presentation ?	Transparent
Originating address numbering plan ?	Transparent
Originating address numbering type ?	Transparent
Destination address numbering plan ?	Transparent
Destination address numbering type ?	Transparent
>> Manipulation	
Default originating address ?	<input type="text"/>
Originating URI parameters ?	<input type="text"/>
Destination URI parameters ?	<input type="text"/>
>> Echo Cancellation	
Apply echo cancellation ?	<input checked="" type="checkbox"/>
Apply automatic gain control ?	<input type="checkbox"/>
Echo cancellation span ?	<input type="text" value="0"/>
Use non-linear processing ?	<input checked="" type="checkbox"/>
Generate background noise ?	<input checked="" type="checkbox"/>
Non-linear processing limit (in dbM0) ?	<input type="text" value="0"/>

Figure 2-12 Advanced Route Options

The Codecs option allows different codecs to be selected for a particular route. The transport option allows UDP or TCP to be the default for outgoing calls. The fax option controls the behaviour for incoming fax calls. By default these options will defer to the global settings that can be set in the General, SIP and Codecs sections of the gateway configuration pages.

Following these are a number of options to force the gateway to use particular values for screening, presentation, originating address plan and type and destination address plan and type.

For voice calls, the echo cancellation set of options govern whether echo cancellation is enabled or not, as well as various other customisable aspects, as described here.

The **Apply echo cancellation** check box will enable or disable echo cancellation on input signals.

The **Apply automatic gain control** check box will enable or disable automatic gain control (AGC) on the input signal. Simply put, this feature ensures a call maintains more or less the same volume, throughout the call.

The **Echo cancellation span** field for the majority of deployments this value can be left at 0, which would trigger the use of a predetermined default value.

The most advanced form of echo cancellation technology that Aculab provides, can be activated by enabling the **Use non-linear processing** check box. By default this is always on.

The non-linear processing technology can be further customised by changing the value in **Non-linear processing limit** field.

Residual echo on a signal is replaced with “comfort noise”, when **Generate background noise** is enabled. When this is disabled then complete silence is heard when neither party in the call are speaking. By default this is enabled, which is the best solution, as comfort noise is usually a good indicator that a call is still active, when neither party are speaking.

The screenshot shows two configuration sections. The first section, titled "Tone Elimination", includes a yellow warning box stating: "WARNING: To mitigate resource use, enabling Tone Elimination will disable Echo cancellation. NOTE: You must enable 'Bridge media streams' on the SIP configurations page." Below this are three fields: "Apply tone elimination" with a checked checkbox, "Minimum duration of tone (milliseconds)" with a dropdown menu set to "No minimum", and "Call leg" with radio buttons for "Incoming call" (selected) and "Outgoing call". The second section, titled "Tone detection", includes a yellow note box stating: "NOTE: The current selected method for sending DTMF digits over IP is RFC2833 encoded RTP. (See SIP configuration tab). Valid options for this method are shown here." Below this is a field for "Minimum duration of tone (milliseconds)" with a dropdown menu showing "No minimum" selected, and a list of other options: "No minimum" and "40".

Figure 2-13 Advanced tone detection/elimination options

The **Tone Elimination** section presents some options to allow a user to customise the elimination of DTMF tones from calls. The **Minimum duration of tone** specifies the amount of DTMF tone to be present before it is considered to be DTMF tone and hence eliminated.

The **Tone detection minimum duration** is by default set to **No minimum**. The default value will trigger the Appliance into eliminating tones as soon as a sample of audio can be identified as containing tone. Other valid values are **40mS** and **64mS**. These values will require at least 40 milliseconds or 64 milliseconds of tone before identifying that a sample of incoming audio contains a DTMF tone, and only then excluding the tone from the outgoing audio.

The **Call leg** option should be set according to which end tones are expected to arrive on. If an incoming call is expected to carry DTMF then **Incoming call** should be selected as the **Call leg** to eliminate tone on. If an outgoing call is expected to carry DTMF then **Outgoing call** should be selected as the **Call leg** to eliminate tone on.

The **Tone detection** section will check which SIP DTMF transmission method is currently selected and will present the appropriate **Tone detection min duration** options in a dropdown box as shown in Figure 2-13.

When the SIP DTMF transmission method is **RFC 2833** the options available are **No minimum** and **40** milliseconds. When the method is **SIP-INFO** or **SIP-INFO-RELAY** the options available are the same as for RFC 2833 but with an additional **64** milliseconds option.

The **No minimum** option will typically expect an audio signal sample to contain less than 40 milliseconds of tone before accepting the presence of tone.

Setting the tone detection minimum duration to 40 milliseconds or 64 milliseconds will mean that for the Appliance to gateway tones, a tone must be present in the audio signal for at least 40 or 64 milliseconds respectively. The Appliance will generate an appropriate SIP message based on which SIP DTMF method was agreed upon during the initial SIP INVITE transaction.

2.1.7 Clocking

This page controls the source of the Gateway's telephony clock. A correctly configured clock is essential for proper operation of the Gateway.

The page displays two columns. The left-hand column shows the Available Clock Sources. These are all the TDM Trunks not currently selected as a possible clock source. The right-hand column shows Selected Clock Sources. These TDM Trunks are currently selected as possible clock sources.

To move an "Available" trunk to the "Selected" column, highlight it and then click the ">" button.

To move a "Selected" trunk to the "Available" column, highlight it and click the "<" button.

To move all "Available" trunks to the "Selected" column, click the ">>" button.

To move all "Selected" trunks to the "Available" column, click the "<<" button.

The Selected Clock Sources are listed in the order of application. This order can be changed by highlighting individual trunks and then selecting **Move Up** or **Move Down**.

There is an additional option to fallback to a locally generated clock source when no other clock source is available.

In operation, the first listed Selected Clock Source that is found to be functional will be used. If, at any time, this should be detected as failed, the Gateway will automatically switch to the next listed functional clock source.

By default all physical TDM trunks are pre-selected with **Fall back to local clock** also selected. This should only need to be changed if the local Gateway installation requires it.

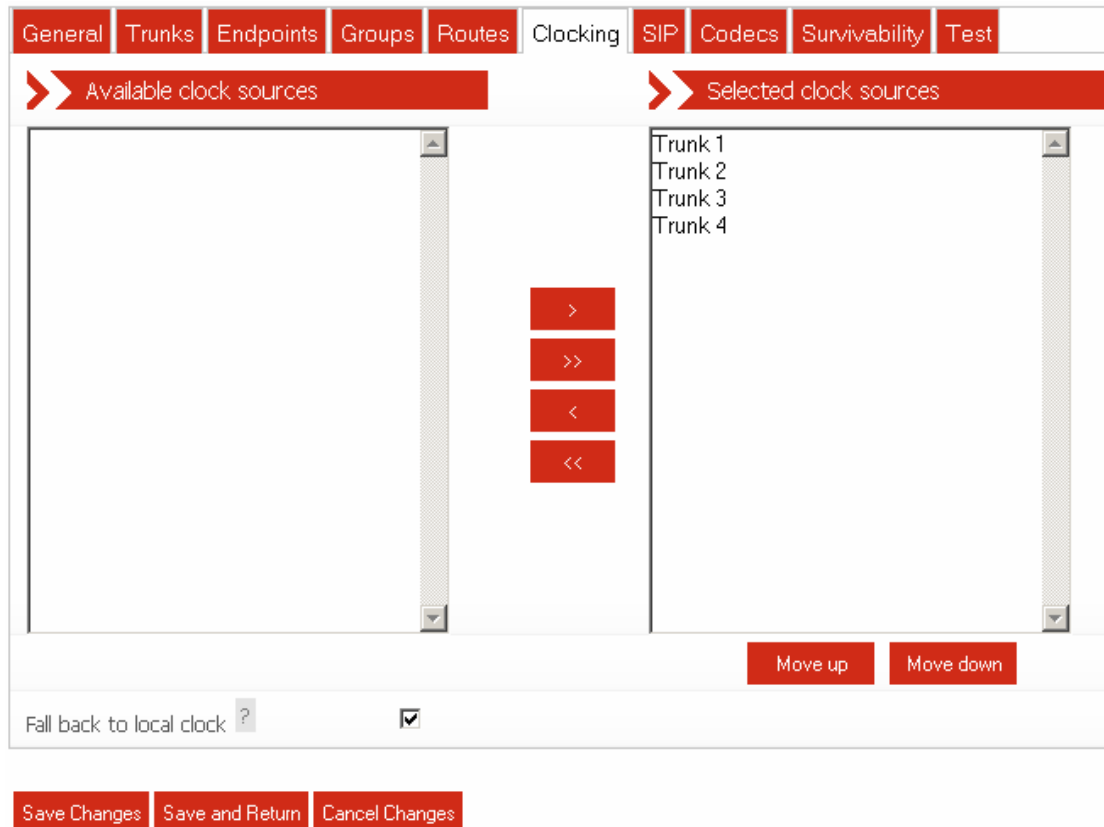


Figure 2-13 Clocking control page

2.1.8 SIP

This page configures SIP telephony settings.

SIP transport for outgoing calls can be UDP, TCP, or TLS as required for the user's network.

DTMF over IP send method: When the RFC2833 encoded RTP option is enabled DTMF tones will be encoded using the RFC2833 codec. When the option to use current codec is selected the tone will be sent in band. Other options are to send the tone in SIP Info messages as SIP Info dtmf or SIP Info dtmf-relay. For SIP Info DTMF type each DTMF tone is stripped from the audio stream and sent as a SIP INFO application/dtmf message. The body of each SIP INFO message indicates the dialled DTMF digit. For dtmf-relay type each DTMF tone is stripped from the audio stream and sent as a SIP INFO application/dtmf-relay message. The body of each SIP INFO message indicates the dialled DTMF digit and its duration.

Tone duration of regenerated DTMF: This option allows the specification of the duration of DTMF tones, regenerated in response to SIP INFO messages of application/dtmf mime type.

Interdigit duration of regenerated DTMF: This option allows the specification of the duration of silence in between DTMF tones, regenerated in response to SIP INFO messages.

Support comfort noise: This option, when selected, will advertise in SIP SDP that comfort noise handling is supported.

Enable 183 provisional Responses: This option allows the sending of 183 provisional responses.

Discontinuous Transmission (DTX): This setting has three options.

- 1) Disabled
- 2) Enabled – No Comfort Noise
- 3) Enabled – With Comfort Noise

When option (1) is selected, during silent periods in a call, the Appliance will send RTP packets containing silence audio.

When option (2) is selected, during silent periods in a call the Appliance will stop sending RTP packets.

When option (3) is selected, during silent periods in a call the Appliance will send packets indicating that comfort noise should be generated (if the remote agreed to comfort noise).

The default is option (3) Enabled – With Comfort Noise.

Enable Packet Loss Concealment (PLC): There is the possibility of RTP packets containing audio being lost on a network, with this feature enabled the Appliance will attempt to conceal gaps in the audio that are the result of lost packets.

Enable RTCP: This option is to enable the sending of RTCP packets.

Use 'sendonly' for Hold: The SDP attribute is set to “a=sendonly to indicate a hold.

Use 'inactive' for Hold: The SDP attribute is set to “a= inactive to indicate a hold.

Use 'recvonly' for Hold: The SDP attribute is set to “a= recvonly to indicate a hold.

Use SRTP on: This option controls how the Appliance applies SRTP. By default SRTP is turned off. You can opt to enable SRTP on all outgoing TLS calls or all calls regardless of transport.

Bridge Media Streams: Applies to SIP to SIP calls. When this option is disabled, the RTP streams flow directly between the SIP caller endpoint and SIP called endpoint. This allows for a greater number of SIP calls to take place. When enabled the RTP streams will flow through the Appliance and as a result the number of SIP to SIP calls will be governed by the available resources on the Appliance and hence will be less. By default this option is disabled.

General	Trunks	Endpoints	Groups	Routes	Clocking	SIP	Codecs	Survivability	Test
>> Transport for outgoing calls									
Transport protocol							UDP		
>> Media options									
DTMF over IP send method ?							RFC2833 encoded RTP		
Tone duration of regenerated DTMF ?							250		
Interdigit duration of regenerated DTMF ?							250		
Support comfort noise ?							<input checked="" type="checkbox"/>		
Send 183 for Ringing ?							<input type="checkbox"/>		
Discontinuous Transmission (DTX) ?							Enabled - With Comfort Noise		
Enable Packet Loss Concealment (PLC) ?							<input checked="" type="checkbox"/>		
Enable RTCP ?							<input type="checkbox"/>		
Use 'sendonly' for Hold							<input checked="" type="radio"/>		
Use 'inactive' for Hold							<input type="radio"/>		
Use 'recvonly' for Hold							<input type="radio"/>		
Bridge media streams ?							<input type="checkbox"/>		
>> Jitter Buffer									

Figure 2-14a SIP configuration page

Require SRTP on incoming calls: When this option is enabled, the ApplianX will reject incoming calls that do not specify SRTP in their SDP.

For incoming SIP calls, the **SIP listening ports** can be changed if required from the default of 5060 (for TCP and UDP) and 5061 (for TLS). In addition, by default, the SIP service will listen on both UDP and TCP ports for incoming calls. If either of these is not required, enter 0 (zero) to disable the port. NOTE: If both ports are set to 0 (zero), the Gateway will be unable to make or receive SIP calls.

Endpoint Monitoring is enabled on a per-endpoint basis. You can control the interval between polling attempts with the **Polling Interval** option.

Call diversion enabled: When enabled, the ApplianX will process diversion information for all SIP calls.

History-Info Message Preferred: When enabled, the ApplianX will use 'History-Info' headers to convey diversion information. When disabled, 'diversion' headers will be used.

Divert as proxy: When enabled, the ApplianX will make a new 'diverting' call if a divert request is received from an outgoing call. When disabled, the ApplianX will

route divert request information received from an outgoing call to the associated incoming call.

Divert unmatched to outgoing group: This option is only applicable when 'Divert as proxy' is enabled. After receiving a divert request, the ApplianX will search for a suitable destination group on which to make the diverting call (Its decision is based on the divert-to address received and the routing rule configuration). When this option is enabled, if no suitable group can be found the diverting call will be made on the same group as the residing outgoing call. When disabled, if no suitable group can be found the diversion will be aborted.

Send Diverted Address: When enabled, incoming calls will be informed of diversions that have occurred on outgoing call legs. When disabled, incoming calls will not be informed of diversions that have occurred on outgoing call legs.

Exchange transfer information:

Exchange Route Optimisation/Path Replacement information:

Following a call transfer involving a SIP endpoint, it is possible that two TDM endpoints may be connected over a SIP call leg where the gateway has called itself. To support subsequent transfers or for DPNSS Route Optimisation or QSIG Path Replacement, the gateway will send itself custom SIP INFO messages. In the unlikely event that this causes problems, either of these features can be disabled here.

CBWF/CBWNNU Enabled: When enabled, this option allows CallBackWhenFree / CallBackWhenNextUsed supplementary service information to be conveyed over SIP to another ApplianX.

Jitter Buffer

Manual jitter buffer configuration

Secure RTP

Use Secure RTP on... No calls (i.e. SRTP not used)
 On outgoing TLS calls
 On all outgoing calls

Require SRTP on incoming calls

Listening ports

UDP listen port (0 to disable)

TCP listen port (0 to disable)

TLS listen port (0 to disable)

Endpoint monitoring

Polling interval

Message Waiting Supplementary Service Support

Accept unsolicited message summary

Send unsolicited message summary

Call Diversion Supplementary Service Support

Call Diversion Enabled

History-Info Method Preferred

Divert as proxy

Divert unmatched to outgoing group

Send Diverted Address

Custom messages conveying non-SIP features

Exchange transfer information

Exchange Route Optimisation/Path Replacement information

CBWF/CBWNNU Enabled

Figure 2-14 SIP configuration page

2.1.9 Codecs

The Applianx Gateway can negotiate and exchange RTP audio with SIP devices using a range of codecs. This page allows the selection and prioritisation of these codecs.

The page displays two columns. The left-hand column shows the Available Codecs. These are all the codecs not currently selected. The right-hand column shows Configured Codecs. These codecs are currently selected.

To move an “Available” codec to the “Selected” column, highlight it and then click the “>” button.

To move a “Selected” codec to the “Available” column, highlight it and click the “<” button.

To move all “Available” codecs to the “Selected” column, click the “>>” button.

To move all “Selected” codecs to the “Available” column, click the “<<” button.

The Configured Codecs are listed in the order that they will be offered in a SIP INVITE SDP. This is also the order of preference when accepting a SIP INVITE. This order can be changed by highlighting individual codecs and then selecting **Move Up** or **Move Down**.

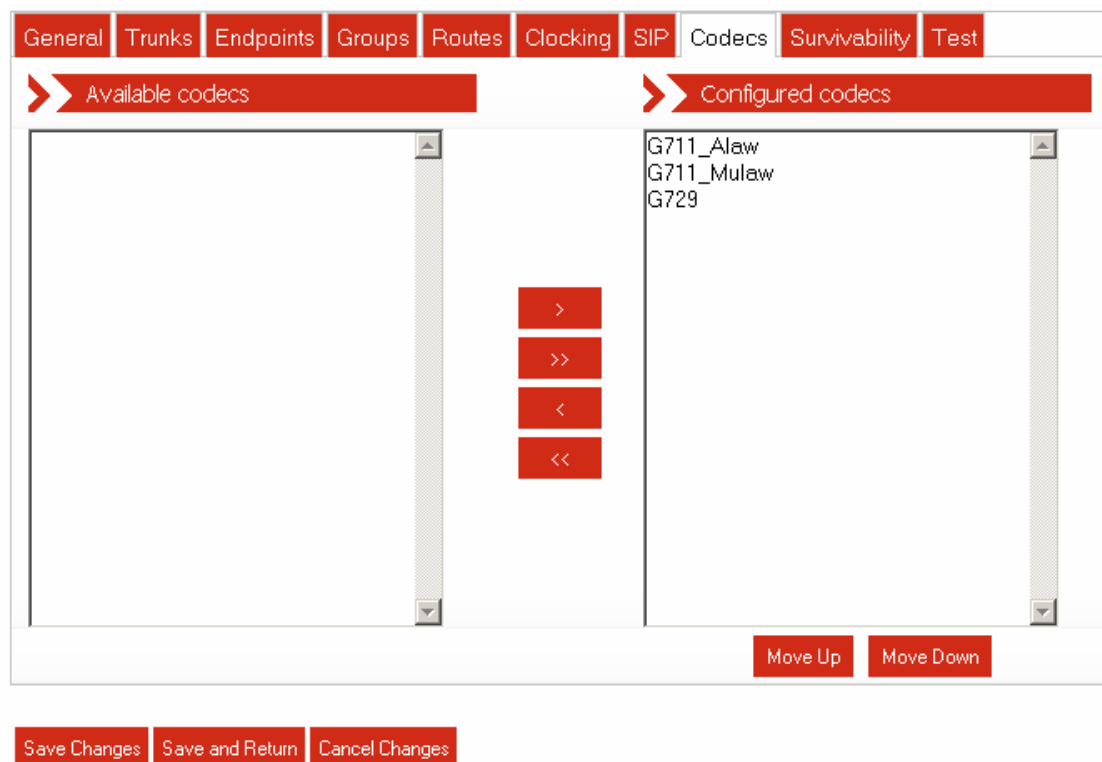


Figure 2-15 Codec configuration page

2.1.10 Survivability

To operate an Appliance in Local Office survivability mode, select the appropriate SIP endpoint to be the Central PBX in the **Central PBX Endpoint** dropdown box. Otherwise set this to [Disabled].

The time (in seconds) taken to switch from active survivability to passive survivability is controlled by setting **Delay Passive Mode Switch Over** (see [3.7 Local Survivability](#) for clarification on active versus passive mode)

Ticking the **Challenge incoming SIP calls from registered and unknown callers** will cause the Appliance to demand incoming SIP endpoints to authenticate themselves, before allowing calls to succeed.

The port used by the Appliance proxy process (see [3.7 Local Survivability](#)) is specified by **Proxy port**.

The port the Appliance Gateway process receives SIP calls on is specified by **Gateway port**. Please do not set **Proxy port** and **Gateway port** to the same value.

It's recommended that, **Promote large UDP packets to TCP**, is enabled. Certain SIP PBX can decorate SIP message headers with long strings of characters which lead to SIP messages that can be very large. The Appliance proxy will promote a large UDP packet to TCP to ensure success. Promotion from UDP to TCP will occur when the value specified in **Maximum UDP packet size (bytes)** is exceeded.

The period for which a SIP endpoint's registration is valid is set by **Registration expiry**.

SIP endpoint authentication realm specified in **Authentication Realm**, this must be the same as that of the designated Central PBX.

The domain used for registrations, **Registrar Domain** is typically the same as the value for **Authentication Realm**.

DDI digit count, the number of trailing digits, in a dialled number, that are used to find a match in the registered users.

When specified **Max CLI prefix** will be used to match against the registrar database.

If an Appliance detects that the Central PBX is responding after a period of non response, the SIP user registrations that the Appliance has records of will be replayed to the Central PBX, **Registration replay batch size** at a time. The replays will pause for **Registration replay interval** seconds between batches of registrations. It is recommended that this is a positive value to prevent the Central PBX from being overloaded.

General	Trunks	Endpoints	Groups	Routes	Clocking	SIP	Codecs	Survivability	Test
> General Configuration									
Central PBX Endpoint ?									CentralPBX ▾
Delay Passive Mode Switch Over ?									0
Challenge incoming SIP calls from registered and unknown callers ?									<input checked="" type="checkbox"/>
Proxy port ?									5060
Gateway port ?									5080
> Proxy Configuration									
Promote large UDP packets to TCP ?									<input checked="" type="checkbox"/>
Maximum UDP packet size (bytes) ?									1300
> Registrar Configuration									
Registration expiry ?									300
Authentication Realm ?									applanx
Registrar Domain ?									
DDI digit count ?									4
Match CLI prefix ?									
Registration replay interval ?									100
Registration replay batch size ?									20

Figure 2-16 Survivability configuration page.

2.1.11 Test

Gateway configuration is quite complex. The Test page is provided to help the user validate the configuration without the need to place live calls.

The **Configuration Issues** section lists any detected inconsistencies that may be a problem e.g. Trunks that are not assigned to a group, or Groups without any routing rules.

The **Test Routing** section allows the user to enter destination and originating telephone numbers along with the incoming call trunk. Selecting **Test!** causes the routing rules for this configuration to be applied as if this were a real call. The different steps of the routing decisions made will be shown similar to that shown below.

Test Routing

This page provides a way to see how the ApplianX IP Gateway will route calls without having to make a configuration live first.
Enter the destination and originating numbers as the ApplianX IP Gateway will see it and select a trunk that the call is to be received on.

Destination address	<input type="text" value="1234"/>
Originating address	<input type="text" value="5678"/>
Trunk call arrives on	<input type="text" value="Trunk 1"/>

Test!

```
INCOMING CALL IS TDM
Matched rule: Route to proxy
Using endpoint: Proxy
  Destination address: sip:1234@192.168.9.91
  Originating address: sip:5678@10.202.205.244
  Numbering presentation: ALLOWED
  Numbering screening: NOT SCREENED
```

Figure 2-16 Test page

2.2. Backing up and Restoring Configurations

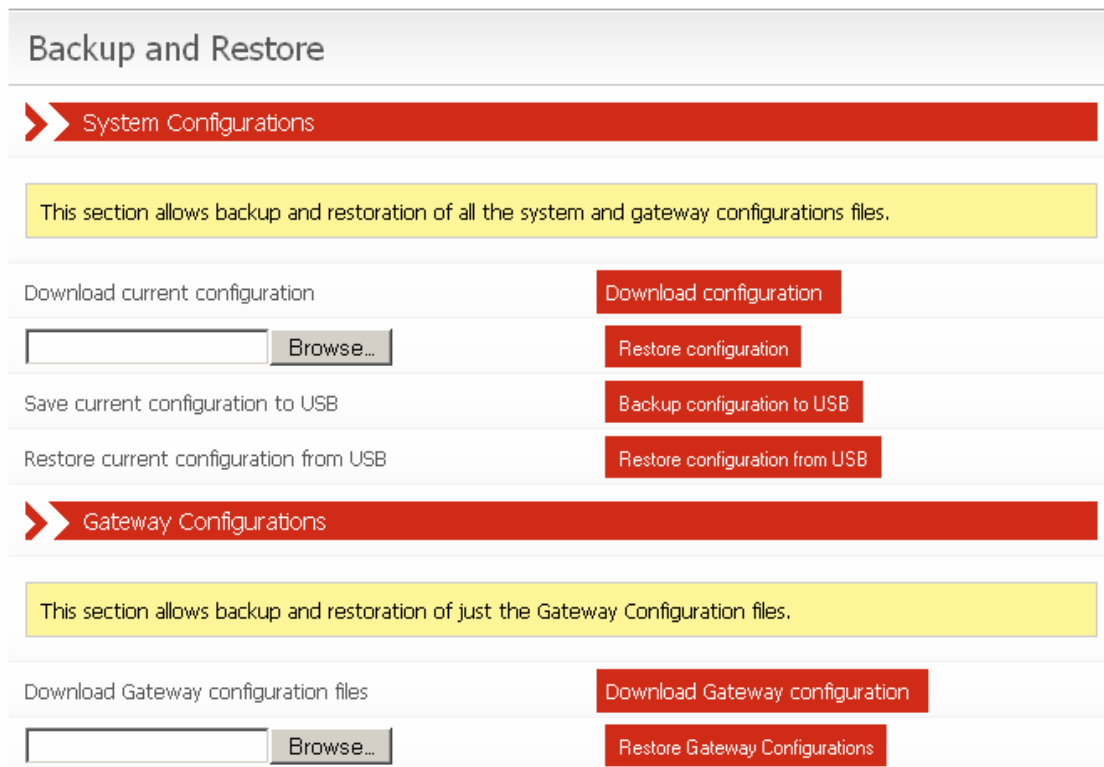


Figure 2-17 Backup and Restore

To save or restore configuration information select **Global Configuration** under the **System Configuration** section in the main menu. This will reveal further options. From here select **Backup and Restore**. This will bring up the backup and restore page as shown below in Figure 2-17.

2.2.1 System configurations

NOTE: The ApplianX backup files contain sensitive information about the ApplianX, including the administrative user passwords, and keys used for TLS and HTTPS encryption. So, backups should be stored securely.

A file download dialog, see figure 2-18, is displayed when **Download configuration** is clicked. This will permit the saving of the Applianx configurations to a zipped tar file to the local disk.

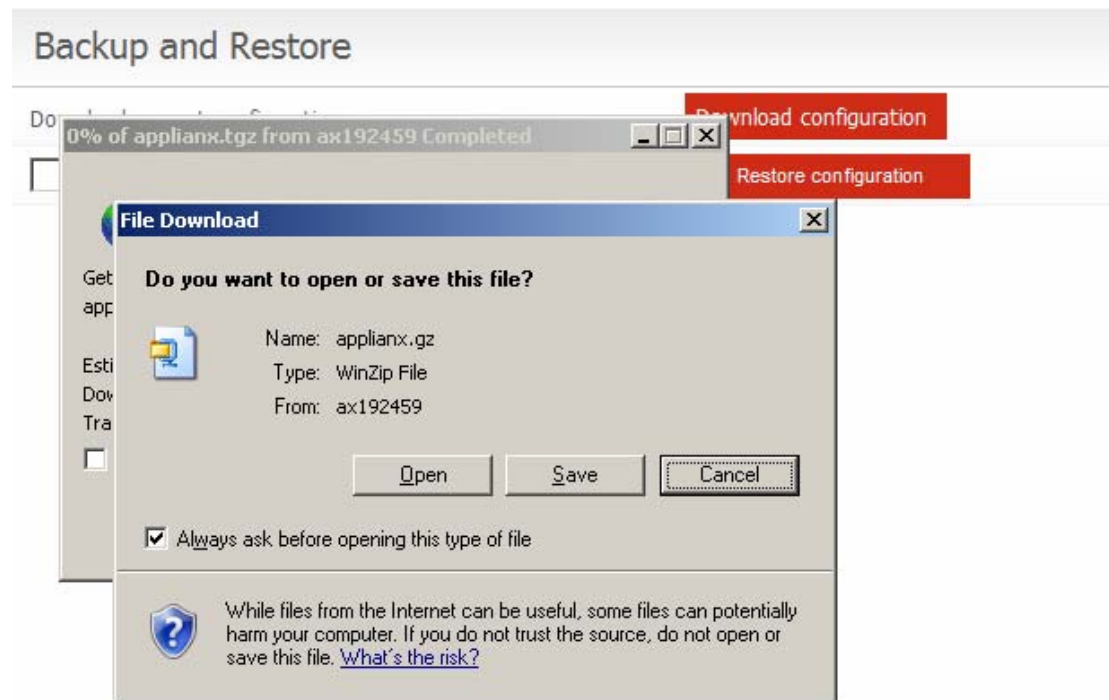


Figure 2-18 Saving the File

To restore a previously saved configuration, some web browsers may allow manual typing of the path and filename to a configuration backup file, in the box provided (see figure 2-19). Alternatively click browse to locate and select the necessary backup file. Click **Restore configuration** to use the selected backup.



Figure 2-19 Restore configuration

If this is successful then you will see the message as shown in figure 2-20

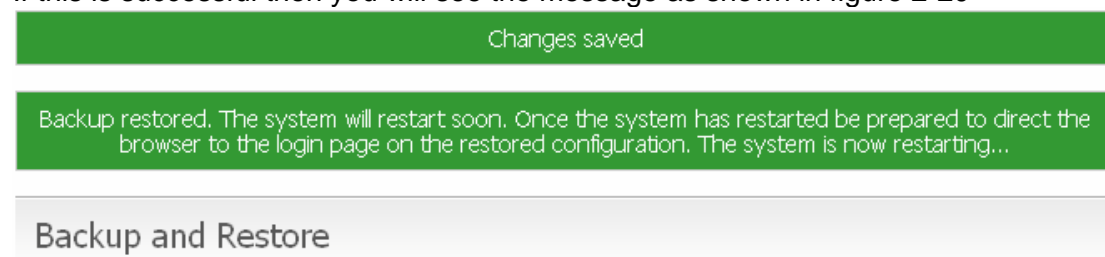


Figure 2-20 Backup restored

Configuration backup files can be saved to and retrieved from a USB flash memory device.

2.2.1.1 Restoring system configuration via USB port

A USB port is located at the front of the Applianx.

To apply an Applianx configuration backup, (see [Figure 2-19 Restore configuration](#)). Copy the applianx.tgz file to a USB storage device. Place the USB storage device in the USB port and reboot the Applianx.

The Appliance will come into service with the configuration settings in appliance.tgz. The configuration file must be called appliance.tgz. The USB storage device should be non-bootable and may contain other arbitrary files.

2.2.2 Gateway Configurations

This feature has been provided for when multiple Appliance devices are to be deployed with the same/similar set of gateway call configuration files, yet maintaining their own network address settings.

Typically a source Appliance device will be configured and tested. Once the operator is satisfied with the gateway call configuration(s), the gateway call configuration files may be downloaded. See figure 2-17. The downloaded gateway call configuration files can then be uploaded (restored) to Appliance devices waiting to be configured with the same gateway call configuration needs.

2.3 Factory Reset

Note: Factory reset will erase any configuration changes you have made and restore the Appliance to the state it was when it left the factory.

To perform a factory reset:

- Reboot the Appliance. This can be accomplished in one of three ways
 - 1) Via the Restart page on the web interface
 - 2) By inserting an appropriately sized tool into the Reset button hole on the front panel of your compact Appliance. The Reset button has to be held in for about a second until all the LEDs extinguish.
 - 3) Or turn off the power and turn it back on again.
- Watch the LEDs on the front panel when the Appliance is rebooting. You need to briefly press the reset button (don't hold it in too long) when the "Warning" LED is lit the first time (approximately four seconds into the boot sequence).
- The Error light will illuminate and then the LEDs will go out leaving only the power LED and Initialising LED on. There will be a short period where it appears as though nothing happens. Then the Appliance will reboot.

It will take approximately 4 minutes for the factory reset to complete. Please note that once up and running the Appliance Admin IP address will revert to the IP address 192.168.1.100.

3.0 Additional Information

3.1 Routing Overview

The routing of telephone calls forms the core function of the Gateway and is the most complex area to configure. A caller dials a number that causes a call to arrive at the gateway. The Gateway applies user-defined rules to the dialled number in order to identify the target user and how they can be contacted. The Gateway then makes an outbound call to this target user and connects the two calls together. This whole process is termed call routing.

Some definitions:

- **Trunk** – a physical connection capable of carrying many calls
- **Group** – a user defined logical group of trunks or endpoints
- **Telephone number** – a sequence of digits associated with a physical telephone, e.g. 01234567890
- **SIP user address** – a sequence of characters in SIP URL format associated with a SIP client user, e.g. johnsmith@hiscompany.com
- **Originating Address** – the telephone number or SIP address of the caller
- **Destination Address** – the telephone number or SIP address of the callee
- **Route** – a set of information that specifies :
 - a pattern to match against a call destination address
 - a rule that allows changes to the originating address
 - a rule that allows changes to the destination address
 - the type of routing to perform (to a Trunk Group or a User)
 - a trunk group on which to make outgoing calls

Some important things to know:

- Each Group must have at least one rule associated with it
- Each Group can contain TDM trunks or SIP endpoints. Not a mixture of both.

3.2 X.509 Certificates

This section provides some information about the use of X.509 certificates for both HTTPS and SIPs. This is not a primer on X.509 or the use of certificates. Instructions are provided (below) for creating a local Certificate Authority and issuing certificates using OpenSSL. Your organisation may have another procedure for obtaining certificates – if so you should use that.

For the purposes of HTTPS and SIP over TLS each device needs an X.509 certificate and a private key. The ApplianX uses two such chains of trust certificates – one for HTTPS and one for TLS.

Out of the box the ApplianX provides a default HTTPS chain of trust but does not provide one for SIP over TLS.

The ApplianX uses X.509 certificates in base64-encoded Privacy Enhanced Mail ("PEM") format. The chains of trust for HTTPS and SIP over TLS are formed by concatenating the private key and the certificate together into a single text file.

The ApplianX will check the validity of its certificates nightly and will warn of expired or nearly expired certificates via SNMP. The ApplianX will warn ten days prior to the expiry of a TLS or HTTPS certificate. Certificate problems are also indicated on the Overview page.

The check for validity is also re-run whenever a change is made to the HTTPS or TLS configuration and this will also lead to the generation of SNMP traps.

3.3 Creating X.509 certificates using OpenSSL

These instructions assume you have downloaded OpenSSL and PERL for your platform. Most Unix-like operating systems (including OS X) will include both PERL and OpenSSL or make it available from their software repositories. For Windows you can obtain OpenSSL by following links on this page:

<http://openssl.org/related/binaries.html>. PERL can be obtained from: www.perl.com.

NOTE: Your organisation may have a set procedure for obtaining certificates. If so, you should follow that procedure rather than these instructions.

First, you need a Certificate Authority which will issue certificates for your devices. You only need to create this once and you should keep a backup of it.

There are a number of ways of doing this using OpenSSL to issue certificates, but for this we will use the CA.pl PERL script that is provided in the OpenSSL package. On a Unix-like system this could be in /usr/lib/ssl/misc/ or /usr/share/ssl/misc/CA. On Windows it will be in the bin directory of the OpenSSL distribution.

In the following instruction, replace the `path/to/CA.pl` with the appropriate path for your system. Unless otherwise noted, all commands will work on Windows and Unix-like operating systems. Here the “>” represents a command line prompt for an operating system shell, your prompt maybe different. Commands for you to type are in *italics*. Make sure that both the perl and openssl executables are in your path.

CA.pl will create a certificate database in your current directory so first you need to create a directory to work in.

```
> mkdir certificates
> cd certificates
```

Create a new Certificate Authority:

```
> perl /path/to/CA.pl -newca
CA certificate filename (or enter to create) [ENTER]

Making CA certificate ...
Loading 'screen' into random state - done
Generating a 1024 bit RSA private key
.....+++++
+
.+++++
writing new private key to './demoCA/private/cakey.pem'
Enter PEM pass phrase:
```

Here you need to enter a secure pass phrase for your Certificate Authority. This is intended to keep your CA secure and make it harder for somebody to issue certificates. You need to remember this phrase as you will need it to issue certificates.

```
Verifying - Enter PEM pass phrase:

Enter the same phrase again.
```


You will be prompted for further information about your CA. You can optionally enter a number of things to provide information about your certificate.

```
-----
You are about to be asked to enter information that will be incorporated
into your certificate request.
What you are about to enter is what is called a Distinguished Name or a
DN.
There are quite a few fields but you can leave some blank
For some fields there will be a default value,
If you enter '.', the field will be left blank.
-----
Country Name (2 letter code) [AU]:UK
State or Province Name (full name) [Some-State]:Bucks
Locality Name (eg, city) []:Milton Keynes
Organization Name (eg, company) [Internet Widgits Pty Ltd]:Wainwright's
Fruit Emporium
Organizational Unit Name (eg, section) []:Kiwi Division
Common Name (eg, YOUR name) []:Wayne Wainwright
Email Address []:yeswehavenobananas@wainrights.com

Please enter the following 'extra' attributes
to be sent with your certificate request
A challenge password []:
An optional company name []:
Using configuration from C:\OpenSSL\bin\openssl.cnf
Loading 'screen' into random state - done
```

That's all of the information you need to give. The tool will now prompt you for the Certificate Authority pass phrase so it can print out information about the certificate:

```
Enter pass phrase for ./demoCA/private/akey.pem:
Check that the request matches the signature
Signature ok
Certificate Details:
  Serial Number:
    82:ac:ff:1e:be:8c:16:32
  Validity
    Not Before: Nov 13 12:46:21 2009 GMT
    Not After : Nov 12 12:46:21 2012 GMT
  Subject:
    countryName           = UK
    stateOrProvinceName  = Bucks
    organizationName     = Wainwright's Fruit Emporium
    organizationalUnitName = Kiwi Division
    commonName           = Wayne Wainwright
    emailAddress         = yeswehavenobananas@wainrights.com
  X509v3 extensions:
    X509v3 Subject Key Identifier:
      6D:8D:85:42:CA:91:B6:FB:F9:CB:53:CE:10:62:15:B5:45:D3:B7:7B
    X509v3 Authority Key Identifier:
      keyid:6D:8D:85:42:CA:91:B6:FB:F9:CB:53:CE:10:62:15:B5:45:D3:B7:7
      B
      DirName:/C=UK/ST=Bucks/O=Wainwright's Fruit
      Emporium/OU=Kiwi Div
      ision/CN=Wayne Wainwright/emailAddress=yeswehavenobananas@wainrights.com
      serial:82:AC:FF:1E:BE:8C:16:32
    X509v3 Basic Constraints:
      CA:TRUE
Certificate is to be certified until Nov 12 12:46:21 2012 GMT (1095 days)

Write out database with 1 new entries
Data Base Updated
```

Now you have a Certificate Authority, you can use it to create certificates for devices on your network.

Creating a certificate is a two step process. Firstly you need to create a certificate request. As part of this process a private key is created for the device. Once the request has been generated you need to sign the certificate with the Certificate Authority's key.

```
> perl /path/to/CA.pl -newreq
Loading 'screen' into random state - done
Generating a 1024 bit RSA private key
.....++++++
.....++++++
writing new private key to 'newkey.pem'
```

You will be prompted for a pass phrase for the private key. You can remove the pass phrase later.

```
Enter PEM pass phrase:
Verifying - Enter PEM pass phrase:
```

Next you will be prompted for information about the device that the certificate is for.

The important field is the Common Name field which you should set to the DNS name or IP address of the device in question.

```
-----
You are about to be asked to enter information that will be incorporated
into your certificate request.
What you are about to enter is what is called a Distinguished Name or a
DN.
There are quite a few fields but you can leave some blank
For some fields there will be a default value,
If you enter '.', the field will be left blank.
-----
Country Name (2 letter code) [AU]:UK
State or Province Name (full name) [Some-State]:Bucks
Locality Name (eg, city) []:Milton Keynes
Organization Name (eg, company) [Internet Widgits Pty Ltd]:Wainright's
Fruit Emporium
Organizational Unit Name (eg, section) []:Kiwi Division
Common Name (eg, YOUR name) []:192.168.1.1
Email Address []:

Please enter the following 'extra' attributes
to be sent with your certificate request
A challenge password []:
An optional company name []:
Request is in newreq.pem, private key is in newkey.pem
-----
```

Now you need to sign the request to create the certificate:

```
> perl /path/to/CA.pl -sign
Using configuration from C:\OpenSSL\bin\openssl.cnf
Loading 'screen' into random state - done
```

Now the tool will prompt you for the pass phrase for the Certificate Authority. It will then print the information about the certificate and prompt you to check that you want to sign the certificate.

```
Enter pass phrase for ./demoCA/private/cakey.pem:
Check that the request matches the signature
Signature ok
Certificate Details:
  Serial Number:
    82:ac:ff:1e:be:8c:16:33
  Validity
    Not Before: Nov 13 12:51:22 2009 GMT
    Not After : Nov 13 12:51:22 2010 GMT
  Subject:
    countryName           = UK
    stateOrProvinceName  = Bucks
    localityName         = Milton Keynes
    organizationName     = Wainright's Fruit Emporium
    organizationalUnitName = Kiwi Division
    commonName           = 192.168.1.1
  X509v3 extensions:
    X509v3 Basic Constraints:
      CA:FALSE
    Netscape Comment:
      OpenSSL Generated Certificate
    X509v3 Subject Key Identifier:
      39:9E:FC:6B:E2:17:B0:D7:8A:7D:B0:21:F0:9A:E8:A9:C7:D9:10:DA
    X509v3 Authority Key Identifier:
      keyid:6D:8D:85:42:CA:91:B6:FB:F9:CB:53:CE:10:62:15:B5:45:D3:B7:7
      B
Certificate is to be certified until Nov 13 12:51:22 2010 GMT (365 days)
Sign the certificate? [y/n]:y

1 out of 1 certificate requests certified, commit? [y/n]y
Write out database with 1 new entries
Data Base Updated
Signed certificate is in newcert.pem
```

You now have a key in the current directory called newkey.pem and matching certificate in newcert.pem. Rename these to something more appropriate.

You can remove the pass phrase from the key using the following command:

```
> openssl rsa -in newkey.pem -out newkey2.pem
Enter pass phrase for newkey.pem:
writing RSA key
```

Finally, you will need to concatenate the device private key and certificate together.

On Unix-like operating systems:

```
> cat newkey.pem newcert.pem > newchain.pem
```

On Windows:

```
> copy newkey.pem+newcert.pem newchain.pem
```

You can install the new chain of trust onto your ApplianX. For other devices to trust your applianx you will need to install the Certificate Authority's certificate as a trusted certificate.

3.4 HTTPS

HTTPS prevents users on the network from being able to eavesdrop on communication with the ApplianX admin interface.

With this release of the ApplianX IP Gateway HTTPS is mandatory. It is no longer possible to contact the admin interface using insecure HTTP - all attempts to do so will be redirected to HTTPS. To enable this, the ApplianX ships with a default X.509 certificate chain. This certificate is common to all ApplianX systems. As such, it will fail the stringent security checks that modern browsers apply. Some browsers throw up significant roadblocks to prevent you from accidentally connecting to a site that fails security checks. You can replace the default certificate with your own if you wish.

Without an HTTPS certificate the ApplianX is not accessible. To prevent the applianx from becoming inaccessible, the applianx will restore its default certificate if it detects a problem with the original one. For example, if you restore a backup taken before HTTPS support was added, the default certificate will be restored. Similarly, if you perform a factory reset the default certificate will be restored.

HTTPS Configuration

When you submit changes to this page the ApplianX IP Gateway HTTP service will be restarted. This will take a few seconds to happen.
NOTE: Changing the HTTPS certificate may trigger a security warning in your browser.

Server certificate

Serial number	DNSName	Start	Expires	Validity
9E05FB88BDEFD7920	ApplianX	22/10/2009	20/10/2019	Ok

Upload new certificate chain file ?

To upload a new certificate chain, browse to it and click “Save Configuration”.

3.5 Secure SIP over TLS

SIP over TLS provides two abilities:

- At its basic level TLS provides a level of privacy, preventing a packet sniffer from viewing the contents of the protocol exchange between parties.
- With all of the security options turned on TLS provides confidence that both parties in a call are who they say they are.

Out of the box the ApplianX has TLS disabled and contains no certificates and has no chain of trust. It is up to the user to generate a chain of trust (see instructions).

NOTE: X.509 certificates contain timestamps that are used to determine their validity. It is important that the clock on the ApplianX is accurate – NTP is the recommended method to achieve this.

NOTE: TLS protects the SIP session only, to prevent eavesdropping of conversation both TLS and SRTP are required.

NOTE: TLS does not prevent a packet sniffer such as Wireshark from determining the parties involved in a conversation. Sometimes this information alone is useful to an interloper.

NOTE: TLS is no substitute for paying attention to network security. In particular the peer validation checks can be subverted if an attacker can interfere with the normal operation of DNS on your network (see for example the tools "Cain and Abel" which are just an Internet search away).

NOTE: You shouldn't try to make a separate connection for each call - this will put unnecessary load on both endpoints as establishing a TLS connection is very CPU intensive. The ApplianX will always attempt to re-use existing TLS connections.

NOTE: In the current IP Gateway release it can be difficult to determine the cause for TLS call failure. In particular it is impossible to distinguish between attempting to connect to a non-existent host, a host that doesn't support TLS, or a host that presents an invalid certificate. Packet sniffing using a tool such as Wireshark can shed some light on this.

SIP TLS Configuration

For TLS to function effectively, this ApplianX IP Gateway needs its own server certificate and at least one trusted certificate. Diffie-Hellman files can optionally be provided to enhance the security of key-exchange.
 For maximum security, enable Require peer certificate and Validate remote host address and set Maximum chain length as low as possible
NOTE: Changes to the Require peer certificate, Maximum Chain Length, and Verify Remote Host Address options require a reboot.

General Settings

Use TLS

Require peer certificate

Maximum chain length

Verify remote host address

Server certificate

Serial number	DNSName	Start	Expires	Validity	Delete?
No server certificate installed					

Upload new certificate chain file

Private key password

Trusted certificates

Serial number	DNSName	Start	Expires	Validity	Delete?
A70F9A525E92A5F7	ApplianX Team	22/10/2009	17/09/2031	Ok	<input type="checkbox"/>

Upload trusted certificate

Diffie-Hellman parameters

Diffie Hellman parameter files installed	Delete?
512-bit Diffie Hellman parameter file installed	<input type="checkbox"/>
1024-bit Diffie Hellman parameter file installed	<input type="checkbox"/>

Diffie-Hellman 512-bit file

Diffie-Hellman 1024-bit file

Changing the following settings will require a reboot of the ApplianX:

Use TLS is the global setting that enables or disables TLS. When this option is disabled, the ApplianX will not listen for incoming TLS calls and will not be able to make outgoing TLS calls.

Require Peer Certificate causes the ApplianX to request the remote party's certificate during session negotiation. If the remote party doesn't have a certificate, the negotiation will fail.

Maximum chain length controls the number of certificates in the remote party's certificate chain that the ApplianX will examine when looking for the signature of a trusted party. This is in addition to the host's own certificate (i.e. if you set the chain length to 1 then a maximum of TWO certificates will be examined). Setting this to 0 will allow the ApplianX to examine all certificates in the chain. This setting has a number of implications. Firstly, examining more certificates simply takes longer. A malicious party could present a very long certificate chain, tying up the ApplianX for a long period of time. Secondly, each additional certificate in a chain of trust increases the opportunities for an attacker to get hold of a legitimate certificate (e.g. through compromising a host holding a CA and generating a new certificate, through social engineering, etc.). Typically you should set this to the smallest number you can.

The **Verify remote host address** option adds an additional check to the TLS handshake. This check will compare the host name in the certificate against the address of the remote host. If the two do not match then the handshake will fail.

For maximum security, all options should be turned on, with the **Maximum chain length** set to the smallest value that will work for your organisation. The ApplianX should be given its own certificate tied to its IP address/DNS name. You should add

the certificates for the Certificate Authorities that you have used to issue certificates to other devices on your network. Additionally you can provide Diffie-Hellman parameters to increase the security of the TLS handshake process.

The following configuration options do not require a reboot:

Sever certificate: This is the certificate chain that the ApplianX should present to other devices on the network. You can upload a single chain of trust which should be in base64-encoded Privacy Enhanced Mail (PEM) format and should consist of the private key for the ApplianX plus the certificate of the ApplianX plus any additional certificates in the chain.

For example:

```
-----BEGIN RSA PRIVATE KEY-----
MIICXgIBAAKBgQC7a0ZYUQX0RYI9UZatoZsmpmkBopi7n2s5cDRcxgzlTm7voUC4
eVKEGyyEZ3FfUhdjRZXazhkRlqrjh7PHBbEnz8uAEI8bEiZIRipB+ly/r8Sn75XE
20Z3gO82zergfWnwQ2oRM77fUKJE3jAfth/7x9vVK1A0FDdhZCxfceVfQQIDAQAB
AoGBAJL+YzDXc20Pq0N+n0hVTMO2lvsivNorAcUN/POanfinWJj3hzRocGmpCnRa
UXAqiY9hvlPae40jKerEvzrkevldKbOoBr75xYKNf3HXppcSC2z4qkzCu6dY4G3U
TbDbdvBduoeqqERuNZZFT4uV+zpJW7UAQ5Zht3vL1H9c0XP1AkEA+TaP0+cB/WMJ
+EsORI5SYVNs/QKB/D0Y7z+OmfrjDxUluk+LEkPMDdd7LX4/uDtraFwhSq2lCnNj
vx+g/oCvdwJBAMCF50IwGqHmSPjFIBLDyDCWMPaMM1QaP2S4GI30dYSjVQxdwyO
6C17ED0f29SZ7JfOUa9XL7ql04fuwC/2xQcCQHwTezZgPDBgv9T74WWmikNkms25
Euh3rtNnDGODcsr015JE6/OzB4QYtX4n7ieWeLS6KeUZYSJwASDl6Wzsuu8CQQCA
DRAih+i24sDISHNsWYA4Y8uyiL+I8ADFGBoSedohrrk91KDAQ5T+GypT3YrTv4Vz
+xCttSnT1VP6x7wgqtulAkEA5sfduII4ZyJgNUER82bvtreCuNzj1qw7Q7+sBl8
9FlAlfRXTIGjJUVgDBaVEuhQfIFHspOtYmPlTQAoMq49Vw==
-----END RSA PRIVATE KEY-----
-----BEGIN CERTIFICATE-----
MIICYTCCAcOCQCeBfuL3v15ITANBgkqhkiG9w0BAQUFADB6MRMwEQYDVQKKEWpB
Y3VsYWIGUGxjMREwDwYDVQQLLEwhBcHBSaWFWuWDETMBEGA1UEBxMKTW91bnQgRmFy
bTEWMBQGA1UECBMNTWlsdG9uIEtleW5lczeELMAkGA1UEBhMCMVUsxFjAUBGNVBA
MTDUFwcGxpYW5YIFRlYW0wHhcNMDkxMDIzMTAyMTIwHhcNMDkxMDI0MTAyMTIw
jBwMRMwEQYDVQKKEWpBY3VsYWIGUGxjMREwDwYDVQQLLEwhBcHBSaWFWuWDETM
BEGA1UEBxMKTW91bnQgRmFybTEWMBQGA1UECBMNTWlsdG9uIEtleW5lczeELMAk
GA1UEBhMCMVUsxDDAKBgNVBAMTA3JvYjCBnzANBgkqhkiG9w0BAQEFAAOBjQAwG
YkCgYEAu2tGWFEF9EWCPVGWraGbJqZpAaKYu59rOXA0XMYM5U5u76FAuH1ZBB
sshGdxX1IXY0WV2s4ZEdaq44ezxwWxJ8/LgBCPGxImSEYqQftcv6/Ep++VxNtGd
4DvNs3q4H1p8ENqETO+3lCiRN4wh7Yf+8fbyitQNBQ3YWQsX3H1X0ECAwEAATAN
BgkqhkiG9w0BAQUFAA0BgQCQo+DvcTkMucmPx7CWo/R3KiBESbArRKPG2OYqH5E
4t4tsqMEX0aqg/tsCwtGlnWlRt/NJidBceG43d/tukLNbNF4hDLFSb01C0CgJoLR
ZT2bFmtn3C7T6MCg4J0ujOIhKcdixuDxrQcdVmxzQE+8IPOSawx0pijEL4c8z2i4I
w==
-----END CERTIFICATE-----
```

If the key has a pass phrase then enter it into the **Private key password** field.

Trusted certificates: These are the certificates that the ApplianX will use to validate the chain of trust presented by each remote host. If multiple Certificate Authorities are in use on your network you can upload multiple certificates. You must click on the **“Submit changes”** button to upload each certificate individually.

Diffie-Hellman parameters: These are used during the TLS handshake process to improve security. You can upload 512-bit and 1024-bit parameters. These need to be Base64-encoded.

3.6 Software Updates

From the 2.1.0 release onwards, the ApplianX uses a whole-image upgrade method. This overwrites the ApplianX software with a different version, allowing upgrade or downgrade to a known version. The user configuration is unaffected by the version change.

Update images can be applied either from a USB disk or over a network by an HTTP server. A simple HTTP server is provided by the ax-img-tool utility.

A number of caveats apply:

- You must not interrupt the upgrade process.
- Downgrade to older versions than the 2.1.0 release is not supported.
- An older version of the ApplianX will not necessarily be able to fully use a configuration generated by a newer version. It is recommended that you take a configuration backup prior to upgrade and use this backup if you should need to downgrade for any reason.

Getting update images

Update images are available from <http://archive.applianx.com>.

Getting ax-img-tool

You can download the latest version of this tool from <http://archive.applianx.com>.

Validating update images

The ApplianX will validate images prior to applying them. This validation helps to ensure that the upgrade is going to succeed.

Additionally, you can manually validate an image using the ax-img-tool utility. You can do this for extra confidence when you have downloaded the image over an unreliable connection.

Validation can be performed from the command line using the ax-img-tool utility:

```
c:\Program Files\ApplianX\> ax-img-tool -i name-of-image
```

Will display:

```
ax-img-tool version 1.0
Copyright (C) 2010 Aculab

Image is IPGATEWAY version 2.0.5 (build 52) for a Standard ApplianX
Image was generated on: Tue Apr 13 16:52:38 BST 2010
Verifying checksum (CTRL-C to cancel)...OK
Image is good
```

If the tool reports that the image checksum is bad, don't attempt to use that image to upgrade your ApplianX.

To apply an update image using HTTP

To do this you need to configure an HTTP server to serve the image. Doing this is beyond the scope of this document, however the ax-img-tool can be used as a simple HTTP server and this is described below.

NOTE: It is not recommended to install updates directly from the official ApplianX update website as Internet connections can be unreliable.

NOTE: It doesn't matter what the image is called when this method is used.

To use the ax-img-tool as a simple HTTP server run it with the `-s` option.

```
C:\Program Files\ApplianX\> ax-img-tool -s name-of-image-file

ax-img-tool version 1.0
Copyright (C) 2010 Aculab

Image is IPGATEWAY version 2.0.5 (build 52) for a Standard ApplianX
Image was generated on: Tue Apr 13 16:52:38 BST 2010
Verifying checksum (CTRL-C to cancel)...OK
Image is good

INSTRUCTIONS
=====

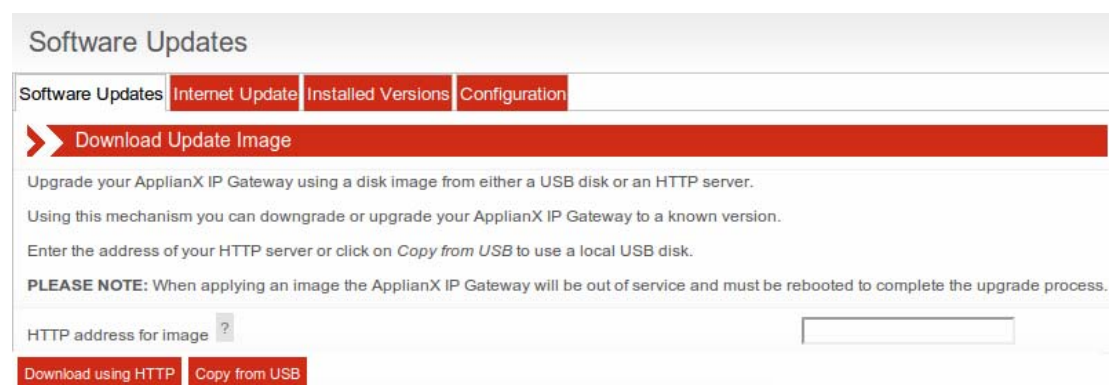
On your ApplianX go to the Global Configuration -> Software Updates page
Paste the most appropriate of the following addresses into the "Image
Address" field:
  http://192.168.1.1:8000/image
  http://192.168.9.19:8000/image
  http://10.202.205.84:8000/image

Then click on "Download image" to begin the image download.

Use CTRL-C to stop serving the image file
```

By default ax-img-tool listens on all interfaces on port 8000. You can specify the interface to use with the `-i` option and the port using the `-p` option.

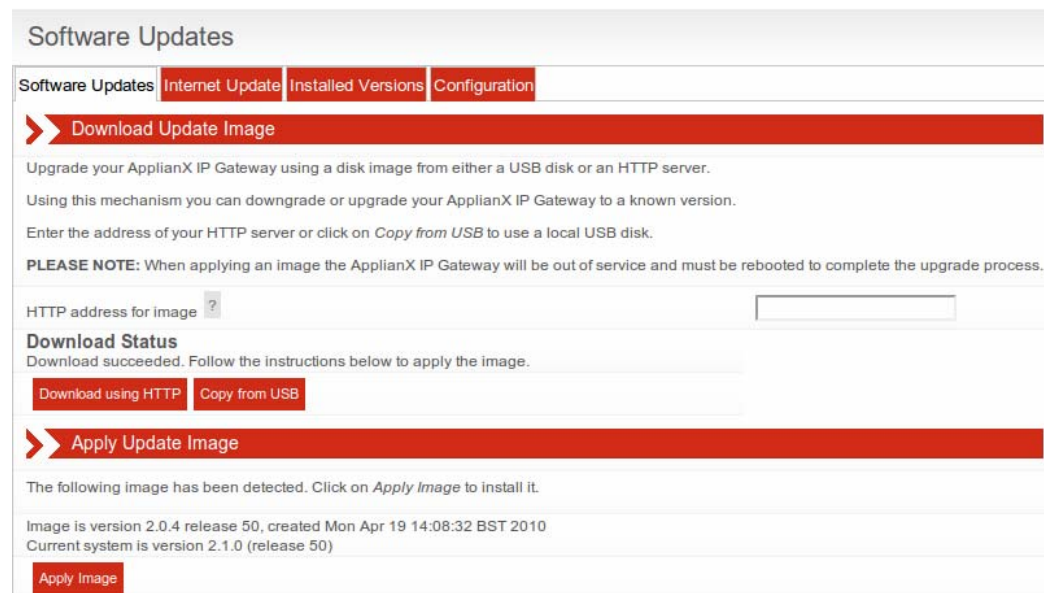
Log into the applianx web interface and navigate to System Configuration -> Software Updates.



Type the URL of the image into the "HTTP address for image" box.

Click "**Download using HTTP**" button and wait.

The image will be downloaded and validated. Assuming the image is valid, the "Apply Image" button will be available.

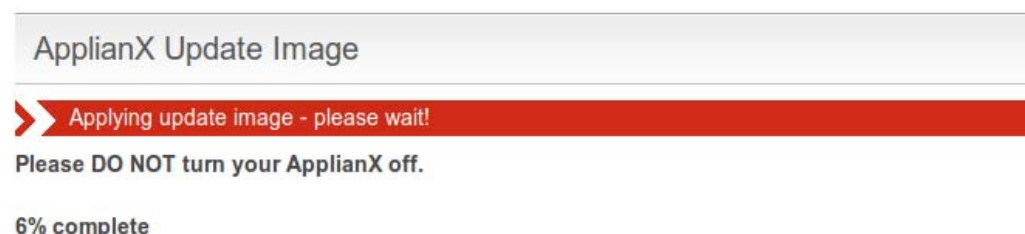


The screenshot shows the 'Software Updates' page with tabs for 'Software Updates', 'Internet Update', 'Installed Versions', and 'Configuration'. The 'Download Update Image' section is active, featuring a red arrow icon and a red bar. Below this, there is instructional text about upgrading the ApplianX IP Gateway, a 'PLEASE NOTE' warning, and a text input field for the 'HTTP address for image'. A 'Download Status' section indicates 'Download succeeded' and provides buttons for 'Download using HTTP' and 'Copy from USB'. Below this is another red arrow icon and a red bar for 'Apply Update Image', followed by text about the detected image and an 'Apply Image' button.

Click "**Apply Image**" to apply the image.


The time required to download the image depends heavily on the speed of your network (on a good network it should take less than 5 minutes).

The update will take approximately 10-20 minutes to apply. During this time the ApplianX must remain powered on.



The screenshot shows the 'ApplianX Update Image' screen with a red arrow icon and a red bar indicating 'Applying update image - please wait!'. Below this, there is a warning: 'Please DO NOT turn your ApplianX off.' and a progress indicator showing '6% complete'.

Once the update is complete the ApplianX must be rebooted. Click on "**Reboot Now**" to do this:



The screenshot shows the 'ApplianX Update Image' screen with the text 'Update complete!' and a red button labeled 'Reboot Now'.

Once the ApplianX has rebooted, follow [this](#) link to the main web interface.

The ApplianX will reboot twice before it is ready for you to login again. Log in as normal, once the second reboot is complete.

To apply an update using USB

USB disks used for software updates must be formatted using FAT32.

Copy the image to the root of the USB disk.

The image must be called "applianx-image" for the ApplianX to detect it.

Insert the USB disk into the USB socket on the ApplianX front panel marked "UPDATE".

Log into the applianx web interface and navigate to System Configuration -> Software Updates

Software Updates

Software Updates | Internet Update | Installed Versions | Configuration

➤ Download Update Image

Upgrade your ApplianX IP Gateway using a disk image from either a USB disk or an HTTP server.
Using this mechanism you can downgrade or upgrade your ApplianX IP Gateway to a known version.
Enter the address of your HTTP server or click on *Copy from USB* to use a local USB disk.

PLEASE NOTE: When applying an image the ApplianX IP Gateway will be out of service and must be rebooted to complete the upgrade process.

HTTP address for image ?

Download using HTTP | Copy from USB

Click on the "Copy from USB" button.

The ApplianX will check the USB disk for a suitable image.

The ApplianX will copy the image to internal storage prior to applying the update. This allows you to remove the USB disk as soon as the initial copy is complete.

The image will be validated. Assuming the image is valid, the "Apply Image" button will be available.

Software Updates

Software Updates | **Internet Update** | Installed Versions | Configuration

Download Update Image

Upgrade your ApplianX IP Gateway using a disk image from either a USB disk or an HTTP server.
Using this mechanism you can downgrade or upgrade your ApplianX IP Gateway to a known version.
Enter the address of your HTTP server or click on *Copy from USB* to use a local USB disk.

PLEASE NOTE: When applying an image the ApplianX IP Gateway will be out of service and must be rebooted to complete the upgrade process.

HTTP address for image

Download Status
Download succeeded. Follow the instructions below to apply the image.

[Download using HTTP](#) [Copy from USB](#)

Apply Update Image

The following image has been detected. Click on *Apply Image* to install it.

Image is version 2.0.4 release 50, created Mon Apr 19 14:08:32 BST 2010
Current system is version 2.1.0 (release 50)

[Apply Image](#)

Click “**Apply Image**” to apply the image.

The update will take approximately 10-20 minutes to apply. During this time the ApplianX must remain powered on.

ApplianX Update Image

Applying update image - please wait!

Please DO NOT turn your ApplianX off.

6% complete

Once the update is complete the ApplianX must be rebooted. Click on “**Reboot Now**” to do this:

ApplianX Update Image

Update complete!

[Reboot Now](#)

Once the ApplianX has rebooted, follow [this](#) link to the main web interface.

The ApplianX will reboot twice before it is ready for you to login again. Log in as normal, once the second reboot is complete.

3.7 Local Survivability

3.7.1 Overview

Local Survivability is a feature intended to improve the resilience of SIP phones operating in an office that uses a central (SIP) PBX that is located in a different office.

If communication with the central PBX is lost, the ApplianX allows users in the local office to continue calling each other and out of the office using a back up method (e.g. via the PSTN).

The ApplianX Local Survivability solution has three components

- The proxy. This is responsible for examining SIP messages as they go past, looking for registration messages.
- The registrar. This is responsible for keeping track of the registration status of local phones, regardless of the status of the central PBX.
- Gateway engine. This is responsible for routing SIP requests when they are received from the central PBX, and takes on Central PBX responsibilities if contact is lost.

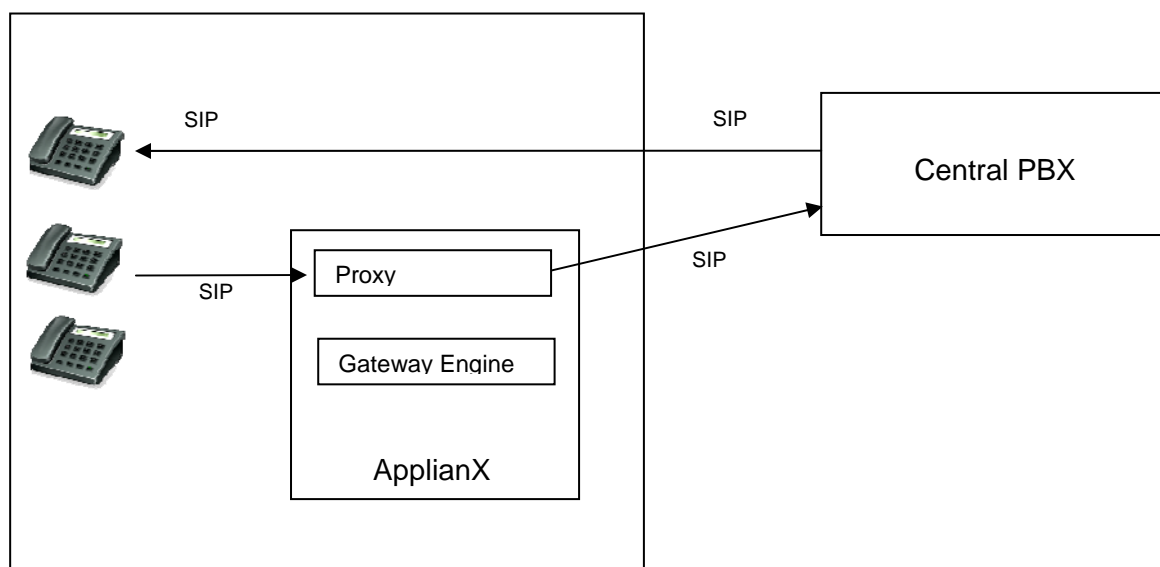
In Local Survivability, the ApplianX listens for SIP messages on two different UDP/TCP ports. The standard SIP port, 5060, is the Proxy. All requests sent to this port are sent directly to the Central PBX.

Another port, 5080 by default, is the “gateway” port. SIP messages arriving on this port are handled by the Gateway engine. Calls arriving here can be routed to the PSTN.

When the Central PBX is responding, the ApplianX is in “passive” mode.

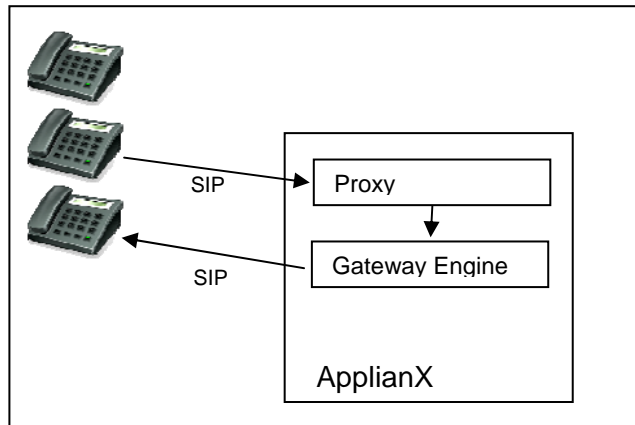
SIP calls from SIP Phones in the local office are proxied straight to the Central PBX. If these calls are intended for phones within the office, it is up to the Central PBX to route them appropriately. It will do this using the registration information supplied by those phones. If calls are destined for the PSTN, the Central PBX must route the calls back to the Gateway port of the ApplianX.

When making a SIP call, all routing is performed by the Central PBX



If the Central PBX becomes unavailable, the ApplianX will switch automatically into “Active” mode. In this mode, the Proxy will route all requests to the Gateway engine. Calls to other offices may be possible using SIP trunking or via the PSTN.

Note: After the switch from Passive to Active mode, SIP calls that were already established may be dropped. This is because the SIP endpoints involved in those calls are likely to have been told by the Central PBX to keep it involved in the exchange of SIP messages and now that the Central PBX is unavailable those messages will not be able to be sent.



3.7.2 Enabling Survivability mode

For an Applianx to be operating in Survivability mode the Applianx must be running a configuration that has a SIP endpoint defined as the central PBX and that endpoint should be configured as the Central PBX for Local Office survivability. See section, [2.1.10 Survivability](#), for more details.

3.7.3 Central PBX responsiveness

Like any SIP endpoint with monitoring enabled, a central PBX is periodically checked for responsiveness. When it is checked, is set by the configuration options found in [2.1.8 SIP](#) polling interval.

3.7.4 Switching to active survivability

When a central PBX becomes unresponsive, it has approximately 30 seconds to become responsive before an Applianx will switch to Active survivability mode.

3.7.5 Registrar

The ApplianX has a simple registrar that can take over local routing responsibilities when the Central PBX is down. The registrar will only operate when Local Office Survivability is enabled.

The registrar requires a list of valid extensions. If SIP devices are using passwords on your network (this is recommended!) then the registrar will need to know these too. For information on how to configure this list, see **Uploading a list of users**, below.

When the ApplianX is in “passive” mode (i.e. the Central PBX is operating), the registrar will examine registration requests from SIP devices that are using the ApplianX as a proxy. The ApplianX will not respond to those requests itself but takes note of the response that comes back from the Central PBX.

When the ApplianX switches into “active” mode, the ApplianX registrar will field registration requests from SIP devices. Any existing registrations will be honoured.

In “active” mode, the ApplianX will route calls to the most recent registered address for a given endpoint.

Matching of calls to registered users has two aspects:

1. There must be a Routing Rule that routes to the Routing Group that contains the Registrar endpoint (this group is called “Registrar”). Calls that match this rule will automatically be compared to users known to the registrar
2. For calls that match the registrar group, the following is done:
 - Take the last n digits from the number (the number of digits is controlled by the “DDI digit count” field on the Survivability tab)
 - If the number is not long enough, leading 0s are added to make it the correct length.
 - The number is then compared against the extensions in the registrar database.

Calls that do not match entries in the registration database will be matched against subsequent routing rules.

When the Central PBX recovers, the ApplianX will switch back into passive mode. At this point it will pass its current registrations on to the Central PBX, unregistering phones that are not registered locally, and re-registering phones that are registered locally.

Common scenarios

By default a rule that checks all calls against the Registrar is configured:

Name ?	DDI/DID criteria ?	DDI/DID man. ?	CLI/ANI criteria ?	CLI/ANI man. ?	Destination ?
Registrar lookup	%	%	%	%	Registrar

This may not be what you want. You can safely delete this rule and add different ones.

If you only want calls with a fixed number of digits to match, add a routing rule like this (each question mark matches one digit):

Name ?	DDI/DID criteria ?	DDI/DID man. ?	CLI/ANI criteria ?	CLI/ANI man. ?	Destination ?
local users	?????	%	%	%	Registrar

If calls with a specific prefix should be routed to the registrar, use a rule like this:

Name ?	DDI/DID criteria ?	DDI/DID man. ?	CLI/ANI criteria ?	CLI/ANI man. ?	Destination ?
users from TDM	016969?????	%	%	%	Registrar

More than one rule can be specified:

Name ?	DDI/DID criteria ?	DDI/DID man. ?	CLI/ANI criteria ?	CLI/ANI man. ?	Destination ?
local users	?????	%	%	%	Registrar
users from TDM	016969?????	%	%	%	Registrar

3.7.6 Configuring SIP devices

All SIP devices are different, but in general:

- configure the SIP device to use the ApplianX as a proxy.
- configure the SIP device to use the Central PBX as a registrar (because the ApplianX is configured to be the proxy, all registration requests will pass through the ApplianX).
- for DNS resilience, configure the SIP device to use the ApplianX as a DNS server.

On the Central PBX, configure the default registration expiry to be relatively short – minutes rather than hours.

If the Central PBX needs to use the ApplianX as a SIP-TDM gateway, configure the Central PBX to connect to the “Gateway” port on the ApplianX (this is port 5080 by default).

3.7.7 Uploading a list of users

Manage Aliases

WARNING: Uploading a CSV file will overwrite your current aliases

A Comma-Separated-Value (CSV) text file containing a list of users maybe uploaded using the ‘Upload Aliases’ web page. In the CSV file, each user is represented by 3 text fields, the format of which is:

<USERNAME> , <PASSWORD> , <DESCRIPTION>

For example, the contents of “my_aliases.csv” might read:

```
joe_bloggs,JoEsPaSSWoRd,Joe is better than Jill
jill_bloggs,jiLlSpAsSwOrD,Jill is better than Joe
```

Note: Take care to ensure only two commas separate the 3 text fields, and that the last character in the CSV file is not a new line character (That would misleadingly indicate the start of an incomplete user entry).

3.7.8 Downloading Aliases

Manage Aliases

WARNING: Uploading a CSV file will overwrite your current aliases

There are 200 user record(s) available for saving.

The **Download Users** button will be visible when SIP users have been previously uploaded, as described in section [3.7.7 Uploading a list of users](#). When this button is clicked, the operator is offered the users details in CSV file format, which can be saved to local disk.

See figure above. In this example 200 SIP aliases have been uploaded, an appropriate message is displayed stating the number of available aliases.

The **Clear Users** button will delete all sip user bindings/registrations and also delete all sip users from the database of sip users.

Clear Users button has been provided for use during Applianx maintenance by the operator and is not recommended for use during normal Applianx operation.

If there are no records then both the download users and the Clear Users buttons will not appear and an appropriate message is displayed in their place.

3.7.9 View Aliases

View Aliases

User name pattern

Rows to display

 200 users registered, 150 registrations, This query resulted in 15 records.

User	Description	Registrations	Last renewal	Expiry time	Seconds remaining	
regtest_usr_009		<Empty>				
regtest_usr_019		<Empty>				
regtest_usr_029		<Empty>				
regtest_usr_039		<Empty>				
regtest_usr_049		<Empty>				
regtest_usr_059		< sip:mr_user059@1.2.3.4>	2013-02-06 15:50:40	2013-02-06 16:50:40	3000	<input type="button" value="Clear"/>
regtest_usr_069		< sip:mr_user069@1.2.3.4>	2013-02-06 15:50:40	2013-02-06 16:50:40	3000	<input type="button" value="Clear"/>
regtest_usr_079		< sip:mr_user079@1.2.3.4>	2013-02-06 15:50:41	2013-02-06 16:50:41	3001	<input type="button" value="Clear"/>
regtest_usr_089		< sip:mr_user089@1.2.3.4>	2013-02-06 15:50:41	2013-02-06 16:50:41	3001	<input type="button" value="Clear"/>
regtest_usr_090		< sip:mr_user090@1.2.3.4>	2013-02-06 15:50:41	2013-02-06 16:50:41	3001	<input type="button" value="Clear"/>
regtest_usr_091		< sip:mr_user091@1.2.3.4>	2013-02-06 15:50:41	2013-02-06 16:50:41	3001	<input type="button" value="Clear"/>
regtest_usr_092		< sip:mr_user092@1.2.3.4>	2013-02-06 15:50:41	2013-02-06 16:50:41	3001	<input type="button" value="Clear"/>
regtest_usr_093		< sip:mr_user093@1.2.3.4>	2013-02-06 15:50:41	2013-02-06 16:50:41	3001	<input type="button" value="Clear"/>
regtest_usr_094		< sip:mr_user094@1.2.3.4>	2013-02-06 15:50:41	2013-02-06 16:50:41	3001	<input type="button" value="Clear"/>
regtest_usr_095		< sip:mr_user095@1.2.3.4>	2013-02-06 15:50:41	2013-02-06 16:50:41	3001	<input type="button" value="Clear"/>

The uploaded aliases, [3.7.7 Uploading a list of users](#), can be viewed on the View Aliases page. This page auto refreshes every few seconds.

Aliases can be filtered by specifying a pattern in the **User name pattern** field. The pattern can be an exact string or a substring containing the SQL wildcard character % one or more times. For unfiltered results leave this field blank.

The number of records displayed can be controlled by setting **Rows to display**. Specifying 0 will return all available records that match the query parameters; this is not recommended on a system with a very large number of registered aliases. A default has been provided, which overrides the user specified value after each query execution and display.

More Records will be enabled when **Rows to display** is less than the number of records resulting in the latest query. When clicked, More Records will display the next batch of records.

To run a query, click the button **Run Query**. Query statistics are displayed next to the **Run Query** button.

Individual registrations can be deleted by clicking the **Clear** button displayed next to each record. Alternatively all registrations can be deleted by clicking **Clear All Bindings**. These buttons have been provided for use during Applianx maintenance by the operator and are not recommended for use during normal Applianx operation.

3.7.10 Overview page

Overview	
Status	Running
Incoming calls	0
Outgoing calls	0
Unroutable calls	0
Central PBX	Alive
Local Survivability	Passive Force Active
Clock source	Local
Active configuration	My configuration

Required Actions

If the running configuration has a central PBX endpoint defined then the overview page will show the entries **Central PBX**, **Local Survivability** and the button labelled **Force Active**.

Central PBX shows whether the nominated central PBX is responding (“Alive”) or not responding (“Unresponsive”).

Local Survivability reports the survivability mode in which the ApplianX is currently operating. It will report “Passive” when the central PBX is doing the call processing and the Applianx is simply acting as a proxy. It will report “Active” when the Applianx has automatically taken over call processing duties because the central PBX has become unresponsive.

The **Force Active** button has been provided for the occasions when it may be desired for a central PBX to be taken out of service for maintenance. Clicking this button will put the ApplianX in “Active (Forced)” survivability mode. The ApplianX will now handle call processing locally (as in the case when the central PBX is unresponsive). The central PBX can then be taken out of service. Once clicked, the **Force Active** button will change into the **Release** button. When the maintenance on the central PBX is complete and is operational, **Release** maybe clicked to return call processing duties to the central PBX.

3.8 DDI Barring

DDI Barring

Browse... Upload DDIs

DDI barring file is present on the system Download Barred DDIs Delete Barred numbers file

This feature allows the administrator to place a restriction on the phone numbers that can be called via an Appliance device.

The administrator will supply a plain text file containing the list of numbers to be barred. The expected format is

- Numeric digits only.
- Single entry per line.

Whole telephone numbers need not be specified. For example, an entry in the file could be the dialling code for an area or the prefix for a premium rate number.

An example of a DDI barring file,

```
0845
01604
```

In this example, calls to any premium rate numbers that begin with 0845 will fail, as will calls to Northampton (01604).

The administrator is able to clear the barred numbers by clicking the button "Delete Barred numbers file". The administrator will be given a chance to cancel this action.

The administrator is able to download the list of barred numbers by clicking the button "Download Barred DDIs".

3.9 DNS Caching

The ApplianX contains a DNS cache, intended to improve resilience during periods of network instability.

The ApplianX can be configured with up to two upstream DNS servers on the Network Configuration page. The ApplianX will periodically poll these servers for aliveness. If DNS servers are down it will be reported on the Overview page.

DNS entries are cached according to the expiry time received from the server and if valid will be used in preference to sending a request to the DNS server. If the DNS server is unavailable, cached entries will be used even if they have expired.

When Local Survivability is active, the ApplianX proxy will look up hostnames from relevant SIP headers so that the names are in the DNS cache if the Central PBX fails.

Note: The DNS cache will not contain DNS entries for hosts that have not been mentioned in headers seen by the ApplianX Proxy. To mitigate this, you can enter static DNS entries on the Static DNS page accessed by clicking on the “Networking” menu option.

4.0 Diagnostics

4.1 Remote Logging

On the main menu on the left of the screen, as seen through the ApplianX web interface, you will see a Diagnostics section. Selecting Remote Logging takes you to the following, Figure 4-0.

Remote Logging

Enable remote logging ?

Host to receive logs ?

Logging port ?

Log sources

Log Type	Status	
Call activity trace	stopped	<input type="button" value="Start"/>
SIP protocol trace	stopped	<input type="button" value="Start"/>
Switch trace	stopped	<input type="button" value="Start"/>
TING trace	stopped	<input type="button" value="Start"/>
Trunk 1 protocol trace	stopped	<input type="button" value="Start"/>
Trunk 2 protocol trace	stopped	<input type="button" value="Start"/>
Trunk 3 protocol trace	stopped	<input type="button" value="Start"/>
Trunk 4 protocol trace	stopped	<input type="button" value="Start"/>

Figure 4-0 Remote Logging

There are no facilities for storing Logging information on the ApplianX. However the ApplianX supports the use of Syslog and can send information using the syslog protocol to a client that can receive the said information. The majority of Linux distributions will include a syslog daemon and it will most likely be running by default. For Windows there are freeware implementations available. Also the ApplianX Trace Tool can receive and decode the protocol messages. The trace available through the remote logging is currently targeted at the ApplianX development engineers and support staff. Check the ApplianX web site www.applianx.com for announcements with respect to the addition of self help tools.

4.1 Diagnostic Log

This page gives a high level record of actions carried out by the gateway. It will also show any errors that the gateway encountered while coming into service. This information should be passed to your support contact if you think that there is a problem with the gateway.

Diagnostic Log

```
2009-03-23 14:32:55 Info System booted
2009-03-23 14:32:55 Info Waiting for hardware detection
2009-03-23 14:33:07 Info Loading configuration My configuration
2009-03-23 14:33:07 Info Clock source is now: Local
2009-03-23 14:33:07 Info Starting protocol firmware download
2009-03-23 14:33:08 Info Firmware download to trunk Trunk 1 succeeded (firmware=dpnss.pmx)
2009-03-23 14:33:10 Info Firmware download to trunk Trunk 2 succeeded (firmware=dpnss.pmx)
2009-03-23 14:33:11 Info Firmware download to trunk Trunk 3 succeeded (firmware=dpnss.pmx)
2009-03-23 14:33:13 Info Firmware download to trunk Trunk 4 succeeded (firmware=dpnss.pmx)
2009-03-23 14:33:13 Info Firmware download complete
2009-03-23 14:33:13 Info Configuration My configuration loaded
2009-03-23 14:33:13 Info System Starting
2009-03-23 14:33:13 Info System Started
```

Figure 4-1 Diagnostic Log

In the above example the system boots and then waits for internal hardware detection to complete. Configuration loading commences and then the Protocol firmware is downloaded to the TDM trunks.

5.0 Troubleshooting

5.1 Logging into the remote interface

5.1.1 I can't get access to the ApplianX Gateway Web Interface

- Try using the ApplianX Search Tool on a Windows PC to detect the ApplianX and to obtain its IP address.
- Try checking the cabling and then try to log in again. The PC and the ApplianX administration port must be connected directly together for initial setup.
- On a Windows XP PC are you using Microsoft Explorer Version 6 or 7? If not try using one of these browsers. Note that version 7 is preferred.
- Try connecting the network port of your PC directly to the ApplianX administration port.
- Try accessing the web interface from an up to date Linux or MAC OS X PC using axnnnnnn.local address if you have one available.
- Did this work? If so you may have DNS/DHCP network issues. Move to using static IP addresses
- Try setting the IP of the admin port to a known static IP address using a USB flash memory stick as described in section 1.9.2.

5.1.2 I log on but the overview screen has errors at the top

- Wait a couple of minutes and then refresh the screen. The web interface can start before the gateway, which means that until the gateway has started, the interface will report that it cannot talk to the gateway engine.

5.1.3 I get a warning saying that the gateway cannot.

- This is normal when the unit has started or been rebooted or has had its IP settings changed. The elements that make up the gateway are just starting and establishing their communication paths.

5.2 Making Calls through the Gateway

5.2.1 I can't make a call from the TDM side of the Gateway to an IP client.

- Check the Call Status Page by selecting "Calls" under the Status section on the menu on the left of the Administration web interface. Now make the call in from the TDM side of the gateway. Check the Call Activity at the bottom of the screen to see whether the Call was received by the gateway. In this case you can see that the Gateway did indeed receive the call but could not route it. You will need to check your routing rules so that the Gateway has the information it needs to route the calls. See section 2 of this User Guide.

Call Activity

Time	Location	Numbers	Message
2007-09-03 17:21:51.054	Trunk 1 Ts: 1	From: 666 To: 888	Released (request_terminated, raw cause=0x10)
2007-09-03 17:21:50.950	Trunk 1 Ts: 1	From: 666 To: 888	Call released (LC_NORMAL, raw cause=16)
2007-09-03 17:21:45.799	Trunk 1 Ts: 1	From: 666 To: 888	Unroutable cal
2007-09-03 17:21:45.798	Trunk 1 Ts: 1	From: 666 To: 888	Incoming call detected

- If there are no calls present then check the Status of the Trunk. This is done by selecting **Trunk Status** from the **Status** section of the menu. If the Trunk is good then the Layer 1 should be showing zero for Slips Errors, Bipolar Violations and Frame Alignment Errors. If there are errors on these then please check the cabling. Ensure that you have configured the correct protocol for the TDM trunks. Also check the options that have been chosen for the protocol and ensure that these are in line with the TDM lines that you are connecting to the ApplianX Gateway.

Layer 1 Information

Slip errors	0
Bipolar violations	0
Frame Alignment errors	0

- If there are no Layer 1 Errors then check the Layer 2. If this isn't showing "green" for the bearer channels on the trunk then there is a layer 2 problem. Check that you have the correct protocol loaded for the TDM trunks that you are connecting to the Gateway. Check with your service provider or PBX maintenance team for set up information for the protocol.

Layer 2 Information



5.3 Configuring the Gateway

5.3.1 I have made changes to the configuration but they don't seem to have any effect.

- The Gateway does not allow you to edit a configuration that is in use. For this reason you can copy a configuration and edit this. Before these changes can take effect you must select that the gateway use this edited configuration. This is done by selecting the “Use” button by the side of the edited configuration on Edit Configurations page.

5.3.1 I used the wizard to create an initial configuration but I have an error saying that there is no active configuration.

Required Actions

Error No active configuration (Please apply an available configuration (See 'Edit Configurations' page), or use the 'Setup Wizard' to create a new configuration)

- On completion of the Wizard a skeleton configuration is created. This configuration though is not automatically activated. On completion of the Wizard you will be directed to the Edit Configurations screen. Here the Skeleton created in the Wizard will be shown under the Available configurations section. Select **USE** to activate that configuration.

Available configurations			
Name	Description	Last updated	
Copy of My configuration		2007-09-20 14:37:53	Edit Delete Copy Use

6.0 Glossary

ApplianX – is a product brand of Aculab and has been developed in order to provide robust and reliable systems for the fast execution of Internet-based communication strategies, with rapid deployment and integration into existing infrastructures.

CAS – Channel Associated Signalling. This is a type of signalling associated with telephony where some dedicated bits in the transmitted stream are directly used to signal information about a particular voice channel. “T1 Robbed-Bit” is well known example of a CAS protocol used in United States.

E1 – 2.048 Mbit full duplex Communication Interface. Used in most countries outside of the United States, Canada and Japan.

HTTP - Hypertext Transfer Protocol. Used on the ApplianX to send information to and from Web Browsers

ISDN – Integrated Services Digital Network. Used within the ApplianX and this document to describe the family of protocol that have there origins in the ITU's Q931 and Q921 specifications. ETS300 102 in Europe and National ISDN 2 (NI2) are typical examples of ISDN protocols.

LAYER 1 – Known as the physical layer in the OSI (Open Systems Interconnection) 7 layer model. Responsible only for getting raw bits from one node to another. It has some alarm and error transmitting capabilities. Basically Layer1 accepts requests from Layer 2.

LAYER 2 – Known as the data link layer in the OSI (Open Systems Interconnection) 7 layer model. This transfers data between two nodes on the same network. It usually has error detection and possibly correction. Within this document and the ApplianX user screens we refer to Layer 2 for TDM protocols. For ISDN protocols this is based upon the ITU (International Telecommunications Union) Q921 standard.

LINUX – A Unix like operating system that is supported and distributed by many organisations. Well know distributions include RedHat, Fedora, Suse, Debian and Ubuntu to mention just a few.

MAC OS X – The Unix based operating system used on Apple (Apple Incorporated formerly Apple Computers Incorporated) PC's.

PBX – Private Branch eXchange. This is a local switch that traditionally terminates POTS (Plain Old TelephoneS) and routes calls between users and into other switches on TDM networks (and more recently IP networks).

SIP – Session Initiation Protocol. A signalling protocol that has been defined by a number of IETF RFCs (Internet Engineering Task Force) that can be used for, among other things, setting up and controlling IP voice communications. The ApplianX IP gateway uses this for the setting up of IP telephony calls.

SNMP – Simple Network Management Protocol. This can be set up on the ApplianX so that SNMP software (not supplied) can be used to monitor elements of the

ApplianX status remotely. This requires use of the MIB (Management Information Base) that can be found on the ApplianX web site <http://www.applianx.com>.

T1 – 1.544 Mbit full duplex Communication Interface. Used mostly in the United States, Canada and Japan.

TDM – Time Division Multiplexed. Used in this document to reference the ISDN and CAS Trunks. Also known as the T1 or E1 interfaces on the ApplianX

Timeslot – A dedicated slot on the TDM interface used for carrying digitised voice and data information. Typically an E1 interface will have 30 of these and T1 will have 23 or 24.

TRUNKS – Either an E1 or T1 interface. A wired connection that carries a collection of voice channels and signalling channels. Sometimes the ApplianX will refer to “SIP Trunks”. This is a virtual concept that all IP Telephony traffic is a “Trunk”. This is for the benefit of writing routing tables and rules.

URI – Uniform Resource Identifier. Within the context of the ApplianX this is used for identifying the addresses of SIP User Agents (IP Phones). It is used in the wider networking world and is not SIP specific.

USB – Universal Serial Bus. Used with the ApplianX for inserting external memory devices for the configuring of IP settings and saving and restoring of configurations.

User Agent –Used within this document to indicate a SIP Telephone although it does have meaning in other contexts such as the World Wide Web.

Web Interface – This is the User Interface on the ApplianX that has been designed to work with a web browser (not supplied) to allow administrators to configure, monitor and maintain the ApplianX. Examples of web browsers are Microsoft Internet Explorer (ie6/ie7), Safari, Firefox and Opera to name just a few.

Windows – Within this document Windows is used as a collective term for a number of operating systems developed by Microsoft Corporation. Namely Windows XP, 2003 Server and Vista. (Previous versions such as 3.1, 95, 2000 and ME have not been tested against the ApplianX)

ZEROCONF – This is a set of techniques that automatically creates a usable network without DHCP and DNS servers or manual configuration. This is used in the ApplianX when the unit is set to DHCP and no DHCP server can be found on the network.