



Com.X

Hybrid IP PBX / Gateway

Administrator Guide

(Com.X 1.1 product release)

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Document History

Version	Date	Description of Changes
1.0	01/22/10	Preliminary
1.1	03/02/10	Added Flex Path, Changing administrator password. Call Detail Records, Asterisk PBX Software Management, Call recording, Upgrading the Software and Troubleshooting sections
1.2	03/17/10	Added "Identify by username" to SIP trunks, section on Asterisk Flash Operator Panel, and minor document fixes
1.3	03/17/10	Added Third party modules chapter and hangup detection.
1.4	04/05/10	Updated troubleshooting, added IAX section. Updated screen shots re: GUI updates.
1.5	04/23/10	Updated troubleshooting, updated default IP assignments and access details
1.6	05/04/10	Added Match Digit Timeout field and updated MAN 3000 section, updated Restore section with file upload.
1.7	05/12/10	Added a section on restoring to a replacement device
1.8	05/20/10	Removed the Administration advanced menu option, improved fax-detect, added troubleshooting entry for call token support and SIP call problems. Added status indicators section. Made document Com.X applicable.
1.9	05/26/10	Removed broken reference, added note re: delayed start-up re: disk integrity check. Added instructions for setting the date and time. Added follow-me and note on outbound trunk CID override and CDR implications.
1.10	05/31/10	Updated Class of service re: service break during apply. Updated call recording to include recording conditions and on-demand recording. Added a troubleshooting section for on-demand recording. Added notes re: voicemail disabling resetting call recording settings.
1.11	06/03/10	Updated Man3000 with note on outbound pattern matching.
1.12	06/03/10	Call recording updates and disclaimer
1.13	06/17/10	Added information re: extension blocking and slow dial-out troubleshooting, fixed reference problems
1.14	06/22/10	Added a section on hot-desking, added a section on legacy PBX gateway configuration. Updated the Monitoring section with the latest options available
1.15	07/12/10	Added a Flex Path section on legacy PBX feature set extension. Added a sections on configuring music on hold from and configuring a streaming server, as well as enabling CSV CDR logging and installing queue stats. Also added a section on using hylafax for incoming fax to email services. Added a trouble-shooting section on FXO health issues due to earthing problems. Added a trouble-shooting section on DNS name resolution. Added a section on accessing CDR records in the MySQL database Updated section on extensions re: free ports and free extensions. Updated hot-desking. Added section on TNG integration.
1.16	07/22/10	Added sendmail relay configuration section. Updated Hylafax re: DID-based

Version	Date	Description of Changes
		fax routing, corrected document description. Added section on rp-pppoe. Updated default IP addresses to 192.168.10n.x. Added section on P\$X. Updated NFS instructions. Noted items that are not backed up by the Com.X backup algorithm. Updated and expanded the section on updating software, boot firmware and factory firmware. Updated front panel LED descriptions to include flashing green. Added system recording troubleshooting. Added a GUI troubleshooting entry. Added intercom.
1.17	08/18/10	Corrected Hylafax DID section. Added instructions for adapting commaman3000 for generic use. Update section on dial patterns. Corrected sendmail configuration instruction error. Added a section on echo cancellation and fax support. Included sections on configuring hylafax for PDF and sending faxes using Hylafax, and accessing fax logs. Added a troubleshooting entry for transfer using Snom phones. Improved Man3000 section, improved fax troubleshooting section. Added a section on beep codes. Simplified commaing installation. Added a section on remote SIP extensions and firewall settings. Added a section in re-installing a Com.X. Updated gnump3d URL. Added a section on packet capture re: DHCP detection. Added MSN information. Formatted notes differently. Added warnings re: streaming music configuration and apply locking up. Added notes on modem usage in the Fax section, expanded trouble-shooting. Improved inbound dialplan examples to use X. instead of '.' Made minor adjustments to Flex Path examples, updated third party modules to include streaming music server support. Removed commaqueuestats.
1.18	09/06/10	Marked as supporting comx-1.1.x
1.18.1	09/06/10	Added detail on dynamic agents logging in and out of queues, included notes on callback, expanded section on parking lot, additional FXO trouble-shooting, nmap port scan example, section on configuring the NTP server. Updated fax troubleshooting.
1.18.2	09/29/10	Added a section on echo cancellation.
1.18.3	09/30/10	Added a section on configuring voicemail to email, updated echo cancellation trouble-shooting.
1.18.4	10/11/10	Added a section on securing a Com.X unit. Added backup location.
1.18.5	10/14/10	DHCP server range no longer configured by default on lan3 re: possibly conflicting on client networks. Added more information re: IVR direct dial and music on hold upload.
1.18.6	10/15/10	Updated sendmail relay section re: sendmail configuration bug. Comment on directed pick-up's dependence on follow-me.
1.18.7	11/12/10	Added a section on transfer, parking and ring-back. Removed the trouble-shooting section content in favour of using the Com.X Troubleshooting Guide.
1.18.8	11/15/10	Added Man3000 images and port back
1.18.9	01/21/11	Updated Man3000 instructions re: new version of Man3000 released. Updated copyright
1.18.10	02/22/11	Updated Man3000 instructions re: release of commaman3000 1.0.4 and improved back-of algorithm
1.18.11	02/23/11	Added instructions for configuring the system's time zone
1.18.12	02/28/11	Added instructions for booting a Com.X from USB and recovering the comma user password
1.18.13	03/07/11	Added additional sendmail relay configuration information
1.18.14	03/10/11	Improved instructions for configuring time zone and time
1.18.15	03/28/11	Specified pin-out for null-modem serial cable
1.18.16	04/06/11	Added instructions for enabling TFTP services
1.18.17	04/11/11	Added more information around BRI bus termination.
1.18.18	04/11/11	Added a section on video codec support and alternative download URL for the GNU streaming music server.
1.18.19	04/21/11	Added a section on configuring static routes (networking)
1.18.20	05/09/11	Added instructions for using Outcall and asttapi re: Microsoft Outlook

Version	Date	Description of Changes
		integration and fail2ban.
1.18.21	05/11/11	Added note on user pin requirement for hot-desking
1.18.22	05/12/11	Updated call recording behaviour, fixed NFS reference to recording directory
1.18.23	05/12/11	Updated Flex Path include section.
1.18.24	05/19/11	Added note re: external primary DNS lookup
1.18.25	05/24/11	Added VPN to security section
1.18.26	08/01/11	Added instructions for adjusting log levels
1.18.27	08/08/11	Added a section on mounting USB drives
1.18.28	08/18/11	Expanded section on fax timing
1.18.29	08/25/11	Updated call recording sections re: admin configuring settings for all users. Also updated section 3.6.4.2 re: overlap dial support only on PRI
1.18.30	08/31/11	Added note in timing section on rxfifo in ita → dahdi direction
1.18.31	09/06/11	Added note on phonebook CSV export delimiter ','
1.18.32	09/08/11	Updated the section on FXO trunk pickup testing
1.18.33	09/09/11	Updated repository location to http://update.commanet.co.za/repo/comx-1.1.x
1.18.34	09/23/11	Updated fail2ban section with note re: starting the service to ensure the configuration is correct
1.18.35	09/26/11	Added the L trunk dial flag for the maximum call length to General Settings
1.18.36	10/24/11	Added details on using + and for number insertion in dial patterns section
1.18.37	11/10/11	Removed “,” from the dial pattern list. Correction to tftp service configuration (-s missing)
1.18.38	11/11/11	Added description and updated CDR note re: extension account code field. Also added multiple pickup group instructions
1.18.39	11/11/11	Corrected ring-back archive URL
1.18.40	11/17/11	Added a section on using the Com.X with a UPS.
1.18.41	11/22/11	Synchronized date format for date/time configuration with example
1.18.42	12/02/11	Added '*' in extension pickup group field description
1.18.43	03/15/12	Added note re: trunk qualification support, also updated link to FSN support page for trouble-shooting
1.18.44	03/16/12	Improved section on static routes

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1 Introduction

1.1 Overview

This document is intended for the Com.X PBX and Gateway administrator and details access to, commissioning, management and monitoring of the Com.X range of products, release 1.1, including the Com.X1 and Com.X2 products. It forms part of a suite of manuals and guides for the Com.X product range, which also includes the Com.X End User manual and the Com.X installation guide.



2 Managing the unit

2.1 Accessing the unit

2.1.1 Factory network configuration

The Com.X units ship with three external ethernet ports labeled lan1 - lan3 (enumerated eth0 – eth2 in the Operating System) and one internal ethernet bus (eth3), factory-configured as follows.

Note: When connecting directly with the device (no switch) a cross-over ethernet cable will be required.

Interface	DHCP?	IP	Netmask
lan1 / eth0	DHCP Client	-	-
lan2 / eth1	No	192.168.101.1	255.255.255.0
lan3 / eth2	No	192.168.102.1	255.255.255.0
lan4 / eth3 (internal)	No	192.168.103.1	255.255.255.0

Note: that for Com.X products with serial numbers prior to FSN1X0072 the default IP addresses are: 192.168.1.1, 192.168.2.1 and 192.168.3.1

2.1.2 Authentication

Web login has been pre-configured with the Administrator username and password for convenience.

2.1.3 Remote web login

Once the unit has been powered on and started up, it can be accessed from a Firefox or Internet Explorer web browser with the URL set to the IP address of the unit. Select the Administrator button to log in. Enter your username and password and select 'Login' to authenticate. Please see section 2.2.4 for more detail on changing the administrator's password.

2.2 Configuration management

2.2.1 GUI layout

The Com.X GUI facilitates status monitoring and control of the unit and peripheral devices via a set of menus accessible from the menu bar. After any changes have been made to the configuration, these need to be reviewed and applied before they can take effect.



2.2.2 Configuration status

The current configuration status, indicating which tasks, if any, are pending, can be viewed from the configuration menu:

When the status on the Status tab is SYNCHRONIZED, the configuration is up to date and synchronized with the underlying system configuration resources. If the status is PENDING or MODIFIED, the Review/Apply button will be emphasized and the administrator needs to review and apply the configuration using the Review/Apply button in order to synchronize.

2.2.3 Applying and reverting a configuration

Select the Review/Apply button. A list of tasks to be applied will appear. Review the task list and select Apply to proceed. To revert changes and return to the configuration last applied, select the Revert button. To continue configuration without applying, select Cancel.

During apply, individual task progress and success or failure will be indicated. Once the process has completed, a confirmation is displayed.

2.2.4 Changing the default administrator's password

The password for the default administrative user (admin) can be changed by selecting the "Change Password" option from the Start menu. Enter the existing password and the new password and Accept to apply the change.

2.2.5 Rebooting and Shutting down

Select Start and then Restart or Shutdown (wait 2 minutes before power down).

Note: Any changes that have not been applied will be lost. When the system has been shut down, it remains powered on but halted. Only a physical power cycle will power it on again.

Note: Startup typically takes 2 – 3 minutes, but that the system will check its hard disk integrity every 20th start-up. This could delay start-up by a few minutes.

2.3 Backup and Restore

The entire Com.X configuration can be backed up to the unit's internal hard drive, or downloaded to another machine. Backups can be scheduled daily, weekly and monthly, and individual sections of the configuration can be configured for backup, each with its own schedule. Immediate backups can also be initiated. Backups are stored on the system in `/var/lib/asterisk/backups/`

From the Start menu, select Backup/Restore:

2.3.1 Backup contents

The following selection of backup sets are provided for backup:



2.3.1.1 Configuration, Recordings

All system configuration, including network, device, trunk, extension, voicemail, inbound, outbound and flex path configurations, as well as all recordings present on the system, including system recordings, auto attendant and music.

2.3.1.2 Voicemail

All user voice mailboxes with messages and the listened-to state of the messages. This could result in a substantial archive.

2.3.1.3 CDR

Backs up the system's Call Data Record database.

2.3.1.4 Excluded from backup

The following items are **excluded** from backup and should be provided for by the system installer, using scheduled scripts or other mechanisms:

- NFS configuration
- Mail relay configuration
- Third party module / service configuration (e.g. P\$X, FOP, Queue statistics, Hylafax, PPPoE)

2.3.2 Immediate backup

To initiate an immediate backup, select the Options menu for the desired backup set, and select Backup Now. After successful completion, the backup archive will be listed under the appropriate section.

2.3.3 Scheduled backup

To schedule a backup, select the Options menu for the desired backup set, select schedule and select the desired backup frequency. To remove a schedule, select Manual Only.

The schedule setting is reflected in the backup set's title bar:

2.3.4 Managing existing archives

As scheduled backups complete and immediate backups are initiated, the existing archives are listed under the appropriate sections on the backup panel. Most recent backups appear at the bottom of each section. All columns in these tables can be sorted by clicking on the column header.

Individual backup archives can be managed by right-clicking on the entries. To download the archive to a local disk, select Save As and follow the browser prompts:

2.3.5 Restoring an archive onto the same system

To restore from an existing archive, right-click the archive in the list and select Restore. The list of steps to be executed during the restore are displayed. Select Continue to restore, or cancel.

Reboot the system once restore has completed (please see 2.2.5).



To restore from a data file saved at an external location, first upload the archive to the Com.X and then follow the procedure above:

2.3.6 Restoring an archive onto a replacement system

In case of hardware failure, it might be important to restore the configuration of a system onto a replacement unit.

Follow the instructions in section 2.3.5. Once restored, select Start – Configuration and the Hardware panel. The restored device should be listed, but the MAC address will not match that of the device in the replacement unit. As a result some telephony spans may not be available.

To update the restored configuration with the replacement device's MAC address, edit the hardware entry (double-click or right-click-edit) and select the selection button to the right of the MAC address field.

It is important to note that the telephony port configuration of the replacement unit must match that of the system to be replaced. The restore process will only list detected devices for which the port configurations match.

Select the appropriate detected device, accept and apply the changes. A system restart may be required to ensure all telephony spans are working.



3 Commissioning the unit

3.1 Global settings

A range of global settings are available for configuration,

Double-click a field to configure the following fields:

Field	Content
Host Name	The system hostname
Domain	The system domain name. Use 'local' to indicate a local domain.
External Address	If specified, outgoing SIP packets will include this address as the source address of the SIP packets.
External is Dynamic	When true, indicates that the SIP source host specified should be treated as a dynamic host.
Primary DNS Server	The system's primary Domain Name resolution server <div style="border: 1px solid black; padding: 5px; margin-top: 5px;"><p>Note: the Com.X uses an open-source, synchronous DNS lookup library. As a result, the Com.X makes use of a DNS proxy to facilitate DNS lookups. If the primary DNS is configured to an IP address outside the LAN (i.e. in front of the network gateway router) and this DNS server should become unavailable, the Com.X and the DNS cache might both block on DNS lookup, resulting in loss of SIP registration and general call failures. Always configure the primary DNS as the LAN gateway router, and the secondary DNS as an external DNS server if required.</p></div>
Secondary DNS Server	The system's secondary Domain Name resolution server
Jitter Buffer Enabled	When enabled, a SIP jitter buffer will be instantiated for each SIP call.
Jitter Buffer Max Milliseconds	Indicates the maximum number of milliseconds that the jitter buffer is allowed to buffer media before streaming out.
Jitter Buffer Adaptive	When enabled, uses an adaptive buffering algorithm that adjusts to the media stream conditions. When disabled, the buffer length is fixed to the maximum value specified.
Match Digit Timeout	The timeout in milliseconds after which, if no additional digit is pressed, the dialed digit series is deemed complete and the call is attempted.



Note: Global settings take immediate effect and do not need to be applied as described in 2.2.3.

3.2 Network configuration

The Com.X product provides functionality to configure all four ethernet interfaces, as well as provision of DHCP IP allocation for IP phones and booting of Comma iTA devices. The Com.X unit can serve both DHCP and Comma addresses simultaneously, provided the IP ranges do not overlap.

3.2.1 Listing ethernet interfaces

To access the list of network interface configurations, select Configuration from the Start menu and then select the Setup panel and the "Ethernet Interfaces" section.

3.2.2 Configuring an interface

Double-click an ethernet interface to configure the following fields:

Field	Content
Name	Operating System allocated interface name
Description	A description for the interface
DHCP Client	Determines whether the unit will try and obtain its IP configuration from a DHCP server. If not selected, the fields below need to be configured
IP Address	An IP address unique to the network. Auto-assigned if DHCP Client is enabled.
Network Mask	The network mask to use on the network. Auto-assigned if DHCP Client is enabled.
Gateway	The routing gateway for this interface. Auto-assigned if DHCP Client is enabled.
DNS Server	The primary Domain Name Server to use for IP address lookup and name resolution. Auto-assigned if DHCP Client is enabled.

Apply the configuration.

Note that after a network apply, the unit needs to be rebooted (please see 2.2.5).

3.2.3 Providing DHCP Server IP allocation to clients

Double-click an ethernet interface and select the Servers tab.



Field	Content
DHCP Start Address	The start address for the range of IP addresses from which new DHCP requests should be served
DHCP End Address	The end address for the range of IP addresses from which new DHCP requests should be served
DHCP Router Address	The address of the IP router / gateway to be used by the device requesting IP configuration, normally the same as the Com.X address on this interface.
DHCP DNS Server	The primary DNS address to be configured on the device requesting IP configuration, may be left blank.

Apply the configuration.

3.2.4 Providing Comma iTA boot services

Double-click an ethernet interface and select the Servers tab.

Field	Content
Comma Start Address	The start address for the range of IP addresses to allocate to new Comma units. Once a unit has been assigned an IP, the system will remember this assignment, across restarts, and will always reserve the IP for that unit. The IP will no longer be served to future commas.
Comma End Address	The end address for the range of IP addresses to allocate to new Comma units.

Apply the configuration.

Note: After a network apply, the unit needs to be rebooted (please see 2.2.5)

Note: The eth3 interface is the internal ethernet bus serving the Com.X Media processor card and should not be changed from its factory configuration.

3.3 Device configuration

To list, detect and configure devices on the network, select the Start button, then Configure and open the Hardware panel.



3.3.1 Configured devices

The list of configured devices is displayed on the hardware panel. Devices are classified as either Comma devices or Generic devices. For each device, the device name, type, model, MAC address and Auto IP values are listed. Where available, the device IP address and current status are also listed.

3.3.2 Detecting and scanning for devices

The Com.X unit can detect network devices present on all network attached to it. By default, device requesting either DHCP IP information, or Comma boot information will automatically be added to the list of detected devices. A scan of all network interfaces can also be initiated by selecting the Scan button. This process may take a couple of minutes.

The number of devices detected is displayed in the top right corner of the Hardware panel. To add detected devices to the configuration, select Add followed by Detected.

This results in a list of detected devices from which one can be selected for configuration. Where available, the existing MAC address, device type, model, IP address and vendor information is listed.

3.3.3 Manually adding devices

If a device is not present on the network or does not appear on the detected list, it can be added manually by providing its MAC address and IP information. Both Comma and generic (e.g. a SIP phone) devices can be added by selecting the Add button, followed by the New menu and the desired device type:

3.3.4 Configuring devices

After a device has been selected for configuration, manually added or edited, a device configuration form will appear:

3.3.4.1 Configuring Comma devices

Double-click an ethernet interface to configure the following fields:

Field	Content
Device Name	A custom name for the device
Model Type	The card configuration of the device. If the device was detected, this field will be populated automatically.
Auto IP	On first boot, request an IP address from the Com.X Manager. Once an IP has been allocated, the Auto IP field becomes disabled and the IP is statically reserved for the device by the Comma Manager, always issuing the same IP to the device.
IP Address	An IP address unique to the device on the network. Auto-assigned if Auto IP Client is enabled.
Network Mask	The network mask the device should be configured with. Auto-assigned if Auto IP is enabled.



Field	Content
Gateway	The routing gateway the device should use. Auto-assigned if Auto IP is enabled.
Attached network	<p>The network interface the device is attached to.</p> <div style="border: 1px solid black; padding: 5px;"> <p>Note: the interface selected here must be configured as providing Comma services, or the addition of the device will fail with the message 'Device not supported on this interface'. The Comma device manager only listens on an provides configuration to devices on boot-up on interfaces configured for Comma services.</p> </div>
Timing Mode	<p>The timing source for the device. Options are:</p> <ul style="list-style-type: none"> •Local: Recover timing from the local Com.X clock (i.e. this device is a PBX master) •System: Recover timing from another master PBX (i.e. this device is in slave mode) This is equivalent to the iTA "Network" clocking mode. •Line 1: Configure this device as the priority 1 timing source with timing recovered from the ISDN interface •Line 2: Configure this device as the priority 2 timing source with timing recovered from the ISDN interface

3.3.4.2 Configuring Generic devices

Field	Content
Device Name	A custom name for the device
Model Type	The type of the device. In most cases this will remain blank for manually added generic devices.
Auto IP	Indication whether the Com.X DHCP server should allocate an IP to the device on DHCP request. Once the DHCP lease on the IP expires, the Com.X DHCP server might issue a different IP to the device. To ensure a device is always allocated the same IP (static), disable Auto IP and enter the IP manually. If the static IP falls in the DHCP pool, the DHCP server will always allocate it to the device (i.e. the DHCP client on the device should still be enabled). If the static IP falls outside the DHCP server IP pool, the device should be manually configured with the static IP and its DHCP client disabled.
MAC	The device's unique hardware address.



Field	Content
IP Address	An IP address unique to the device on the network. Auto-assigned if Auto IP Client is enabled.
Network Mask	The network mask the device should be configured with. Auto-assigned if Auto IP is enabled.

3.3.5 Configuring device ports

When a device is selected in the device list, the available ports on the device are grouped and listed. Comma ports are enumerated using the device's name, the card index in the device and the port number. I.e. device:a2-5 would indicate the fifth port on the second analogue card (a – analogue, b – bri, p – pri) in device. Doh

3.3.5.1 Configuring Analogue Trunk (FXO) ports

Field	Content
Input gain	Indicates how much, if any, gain adjustment should be performed in the input direction. Values are in deciBels.
Output gain	Indicates how much, if any, gain adjustment should be performed in the output direction. Values are in deciBels.
Echo canceller	The echo cancellation delay and algorithm to be employed on the port. The delay up to which the echo canceller can compensate is specified in milli-seconds. An optional non-linear processing echo suppression algorithm (NLP) can also be employed during echo cancellation. If you expect the devices connected to the port to be in a noisy environment how-ever, you might prefer not to enable NLP, as it might result in the near-end noise being cut off when speech ceases. Some users are uncomfortable with this. NLP is recommended for quiet environments.
DID number	A direct inward dialing number to associate with this port. A call incoming on this port may be routed using the DID.
DID description	A custom description for the DID.
Detect caller ID	If this is enabled, the incoming called ID, if available, will be detected and may be used in inbound routing.
Hangup detection	Devices such as Least Cost Routers connected to FXO ports, or the Telco equipment in various parts of various countries may signal hangup differently. Change this setting to match the duty cycle of the hangup signal you are receiving. Options include 250, 500 or 2500 ms of tone followed by the same period of silence and any regular pattern.



3.3.5.2 Configuring Analogue Extension (FXS) ports

Field	Content
Input gain	Indicates how much, if any, gain adjustment should be performed in the input direction. Values are in deciBels.
Output gain	Indicates how much, if any, gain adjustment should be performed in the output direction. Values are in deciBels.
Echo canceller	The echo cancellation delay and algorithm to be employed on the port. The delay up to which the echo canceller can compensate is specified in milli-seconds. An optional non-linear processing echo suppression algorithm (NLP) can also be employed during echo cancellation. If you expect the devices connected to the port to be in a noisy environment how-ever, you might prefer not to enable NLP, as it might result in the near-end noise being cut off when speech ceases. Some users are uncomfortable with this. NLP is recommended for quiet environments.
Allow transfer	If enabled, the phone using this port is allowed to transfer calls.
Immediate mode	When enabled, ring tone is not provided when the phone goes off hook and the PBX is allowed to take immediate action, such as dialing an emergency number. Please see section 3.8.10.7 for more detail on using Flex Path with his feature.
Fax channel	Indicates that this port is to be used for fax. Enabling this setting disables echo cancellation on the port. Please see section 3.3.8

3.3.5.3 Configuring BRI ports

Field	Content
Link Type	Indicates whether this device is part of a point-to-point (i.e. DIDs issued by the telco) or point-to-multipoint (i.e. one or more multi-subscriber numbers issued, or multiple BRI devices sharing the BRI bus.) The link type needs to match the mode of the service provided by the telco.
Act as NT	When disabled, the device is an ISDN TE, connected to the telco NT. When enabled, this port acts as though it is an ISDN NT device. This is useful when the Comma device is placed between the PBX and the telco NT, e.g. for least-cost routing.
Bus termination	Comma BRI cards have terminating resistors on-board, removing the need for external terminators. If this device should terminate the BRI bus (usual case), enable this setting. If multiple devices are attached to the BRI cable (point-to-multipoint) and another device is providing termination, the



Field	Content
	<p>Com.X can be configured not to terminate.</p> <p>A BRI link should always be terminated by a device, or electrical reflection on the link could interfere with BRI signalling and result in loss of layer 1 link establishment, or CRC errors.</p>
Echo canceller	<p>The echo cancellation delay and algorithm to be employed on the port. The delay up to which the echo canceller can compensate is specified in milli-seconds. An optional non-linear processing echo suppression algorithm (NLP) can also be employed during echo cancellation. If you expect the devices connected to the port to be in a noisy environment how-ever, you might prefer not to enable NLP, as it might result in the near-end noise being cut off when speech ceases. Some users are uncomfortable with this. NLP is recommended for quiet environments.</p> <div style="border: 1px solid black; padding: 5px; margin-top: 10px;"> <p>Note: This setting applies to all BRI ports in the same BRI trunk group as this BRI port.</p> </div>
Detect Fax	<p>When enabled, all BRI ports in all trunk groups to which this BRI port belongs will detect fax tones and disable echo cancellation for the duration of a fax call. We recommend enabling this feature on all BRI ports that are likely to support fax calls. This option is not available if the port is not part of at least one BRI trunk group.</p>

3.3.5.4 Configuring PRI

Field	Content
Physical interface	<p>The service offering, either 23B+D T1 or 30B+D E1. For E1, double frame or CRC4 multi-frame can be enabled, and for T1 framing can be either Super Frame (D4) or Extended Super Frame (ESF, using CRC-6)</p>
Switch type	<p>National switch type options are:</p> <ul style="list-style-type: none"> •US National ISDN phase 1 (NI-1) •US National ISDN phase 2 (NI-2) •ETSI "Euro" ISDN: Europe ISDN standard •QSIG: protocol for switching between PBXs
Signalling role	<p>When configured as Terminal, the device is an ISDN TE, connected to the telco NT. When configured ad Network, this port acts as though it is an ISDN NT device. This is useful when the Comma device is placed between the PBX and the telco NT, e.g. for least-cost routing.</p>



Field	Content
Overlap dialling	Enabling this setting will result in digits being transmitted across the PRI link as they are dialed, as opposed to the full number being collected before dialing. Overlap dialing allows a destination number to be specified partially resulting in the first stage of a routing path to be established before the remaining digits arrive.

3.3.6 Removing devices

To remove a device configuration, select Remove from the device's context menu.

Note: Any of the device's ports that are included in trunk groups will be removed from the groups, outbound routes that make use of the ports will no longer be associated with the ports and any extensions associated with device ports or channels will become unassigned.

3.3.7 Echo cancellation

When two phones, B and C are in a call, and the person's speech at phone C is reproduced at phone B, some of the audio is returned as echo. Person C then hears their own voice after a variable delay, i.e. echo.

The person hearing the echo should look to the remote party for the cause of the echo and not to their own phone and system. If both parties experience echo, both systems should be investigated.

In analogue phones, both the 2-wire to 4-wire converter (hybrid) and the handset can introduce echo. The Com.X impedance parameters are aligned with South African telecommunications standards. Non-compliant handsets with a large impedance mismatch is likely to introduce noticeable echo. The delay introduced by VoIP based systems can result in the returned echo becoming noticeable to the person who is speaking.

The degree of echo and the delay between the original audio and the echo is a function of the handset, gain settings, potential wiring problems and the audio transmission path. On short loops the delay before echo is very short, and one usually does not notice the echo. In some cases the call might sound slightly 'tinny'. If echo is noticeable on short loops, the environment in which the call is placed might be conducive to echo.

Echo cancellation algorithms buffer a number of milliseconds of audio, and then analyzes the incoming audio stream and if the buffered audio signature is detected in the incoming stream, attempts are made to remove it (cancelling the echo). The echo cancellations typically buffer for 16, 32 or 64 milliseconds of audio. The larger the buffer configuration, the longer the algorithm might take to converge. In extreme cases convergence might take 3 – 10 seconds, during which call quality would increase to an optimum.

After primary echo has been cancelled in this way, some residual echo may remain (echo present beyond the time-frame the canceller buffered). Such echo is then removed by a non-linear echo cancellation algorithm, if enabled. This may result in



one of the parties' audio being clipped when both parties are talking. Some users may find this uncomfortable. If this is the case it might be preferable to disable the NLP.

Echo cancellation settings may need to be adjusted on a per-port basis, as the audio environment might differ from one port to the next (e.g. an office environment for one range of extensions vs a factory floor for another and an open plan arrangement for another range of extensions.)

Wiring problems, such as power cables crossing telephone cables, or insufficient earthing of the Com.X or iTAs may add noise to the audio path that may cause echo cancellation algorithms to diverge.

In diagnosing echo cancellation problems, establish good earth (~0.5 ohm or less, if resistance approaches 1 ohm earthing might be a concern), and proceed to eliminate each of the potential contributing elements from the equation by testing with different phones, connecting phones directly to the Com.X FXS ports, turning echo cancellation off completely to establish a quality baseline and then trying different combinations of echo cancellation, with and without NLP, on the different extensions. Also try adjusting the gain on problematic extensions.

Note: In exceptional circumstances, if the echo cancellers take a long time to converge (e.g. > 10 seconds), the acoustics of the environment might be introducing too much echo, and / or the handset might be too sensitive.

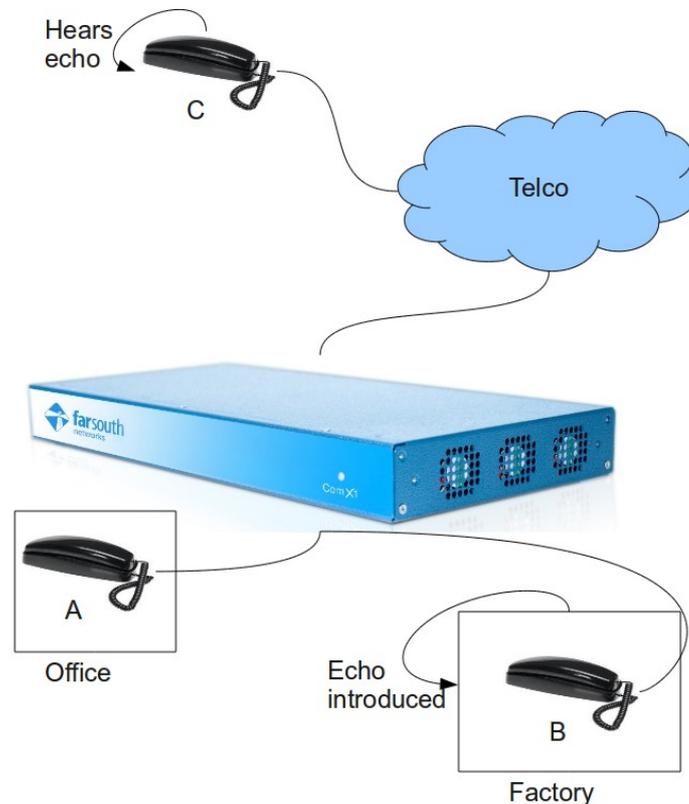
Adjusting the gain on the port down might help, or fixing the Fourier coefficients in the echo canceller for that port might help. After restart, the first call on ports configured thus would take a long time to converge, and there-after the co-efficients are re-used for subsequent calls. Due to the dynamic nature of echo, fixing the co-efficients is not recommended for most cases.

To fix the co-efficients, in `/etc/comma.conf`, add the 'keep' keyword at the end of the port configuration:

```
chan al-1 config ec128 off keep
```

Following this, restart the iTA for the new settings to take effect with:

```
comma-ls -r
```



3.3.8 Configuring devices for Fax / modem support

3.3.8.1 Echo cancellation

Echo cancellation plays an important part in ensuring voice quality through the telecommunications network.

The Com.X product range incorporates state of the art, innovative, embedded on-DSP-chip echo cancellation that off-loads resources from the system server.

Echo cancellation can however affect fax and modem signalling and data. It is therefore important to understand the impact of echo cancellation on fax and modem negotiation and configure devices correctly for fax use.

Echo cancellation is performed at the source device, where 'source' here indicates a network endpoint that introduces a signal into the PBX system. In the Com.X systems echo cancellation takes place on FXO, BRI and PRI ports/channels terminating traffic from the PSTN for routing in the system, as well as FXS ports originating calls via the PBX.

If not configured properly for fax support, faxes and modems may fail or higher baud rate operation might not be possible.

At these sources, if a fax CNG tone is detected and the port / channels are correctly configured for fax, the echo cancellers are automatically disabled, and a clear channel is offered for fax service.



In the case of PRI and BRI channels, channels may serve both voice and fax traffic, and detection of the fax tone is required before echo cancellation can be switched off. This detection is configurable from the Hardware panel for BRI trunk groups and PRI ports.

Since fax devices are usually affixed to individual ports, and specific incoming DIDs reserved for fax use, in the case of FXS and FXO, ports need to be deliberately reserved for fax use by disabling echo cancellation. This is done for FXO ports by disabling echo cancellation, and for FXS ports by enabling the “Fax channel” option in the hardware panel, which disables echo cancellation and configures additional jitter buffering for the port.

3.3.8.2 Telco issues

Telco FXO connectivity, cable distance and environment, etc. introduces additional signal to noise ratio and impedance elements into the equation and may not support high baud rates.

3.3.8.3 Timing

Timing in the system is very important to ensure successful fax / modem communication. The iTA device (e.g. the internal BRI / PRI enabled iTA MPC card in the Com.X1 case or the BRI / PRI iTA in the Com.X2 case) connected to the telco interface should have the highest timing priority (line N), and other iTAs connected should derive timing from this device (system). Please see section 3.3.4.1 for detail on configuring the timing source on an iTA device.

In scenarios where multiple PRI links are configured to the same telco, setting the first PRI as PRI Line 1 is recommended (i.e. deriving clocking from the telco line) and setting all other PRIs to System (i.e. sharing the clocking of the first PRI in the transmit direction and honouring the telco's clocking on each PRI in the receive direction).

In gateway scenarios, where one or more PRIs face the telco and one or more PRIs face a 3rd party device, the telco-facing PRIs should be configured as above, and the PRIs facing the 3rd party device should be configured to use System timing. In this scenario how-ever a further configuration for each of the PRIs facing the 3rd party device is required in `/etc/comma.conf`

```
config.N = t1e1 config t1e1_tclk=sys
```

This indicates to the Com.X that the receive direction clocking (from the 3rd party device) should not be honored (as in the telco case), but that the Com.X should propagate its system-derived timing to the 3rd party device. In such configurations the 3rd party device should be configured as a clock slave.

This additional configuration is not required for BRI ports, as the NT/TE mode setting on BRI pre-selects the correct timing driver on the gateway leg facing the 3rd party device. BRI timing configuration follows the same Line Priority and System configuration pattern as described above for PRI.

If an iTA is connected to a switch located between the iTA and the Com.X host server, network packet delays could occur which could also affect timing. If this is the case, `cat /proc/dahdi/dahdi_dynamic_stats` would show slips and skips increasing on the span to the iTA. The Com.X can be instructed to buffer in the receive direction



from the iTA in order to alleviate such network delays. This is done by increasing the rxfifo value in `/etc/modprobe.d/options` by adding:

```
options dahdi_dynamic rxfifo=4 tasklet=0
```

Please also refer to the Com.X troubleshooting guide for more information on fax and modem troubleshooting.

3.3.8.4 Additional configuration for faxing over PRI

In order to enable clear-channel communication for fax over PRI, the PRI needs to be configured to allow fax detection and the disabling of echo cancellation on fax calls. This is performed by editing / adding the following to `/etc/asterisk/chan_dahdi.conf` to the `[channels]` section

```
echocancel=128
echocancelwhenbridged=yes
faxdetect=incoming (set to 'both' if outbound faxing is also required)
```

Also enable the iTA to recognize the echo cancellation request by adding stream message capability (`stream_msg=on`) in `/etc/comma.conf`

```
config.N = stream p config Y eth stream_msg=on
```

After this configuration, restart asterisk:

```
sudo /etc/init.d/asterisk restart
```

3.4 Extension configuration

3.4.1 Listing existing extensions and ports

To manage the list of configured extensions, select Start, followed by Configuration and the Extensions panel.

Configured extensions are categorized as those having been assigned to extension ports, and those that are unassigned (free extensions). Free extension ports are also listed. These ports are available for either extension assignment or for dynamic extension login (hot-desking).

For each extension, the number, friendly name, port assignment, extension type (analogue or SIP), status and voicemail indicator are displayed.

At the right of the extensions panel, available extension (FXS) and SIP ports are shown, with current extension number assignments to these ports appended in square brackets.

3.4.2 Adding Extensions

Extensions can be added either by selecting extension ports in the ports list and assigning extensions to the ports, or by using the Add button on the Extension panel to add free or generic SIP extensions. Individual extensions, or a range of extensions can be added:



3.4.2.1 Adding extensions from ports

Single extensions or a group of extensions can be assigned to available extension ports by selecting the ports in the ports list and selecting Assign from the context menu.

A starting extension number can be specified, and upon acceptance extensions are numbered incrementally from the starting number for the number of extensions selected.

Note: The end of the range is not specified, instead the number of extensions desired is specified.

3.4.2.2 Adding Unassigned extensions

When a range of extensions is added, the friendly name is set equal to the extension number.

3.4.2.3 Adding SIP extensions

The following password generation schemes are supported for SIP extensions:

- Fixed: The same password is set for all the newly created extensions
- Same as Extension: The password for each newly created extension is the same as the extension number
- Random: A random password generator is used to generate a unique password for each newly created extension.

3.4.3 Configuring extensions

3.4.3.1 Basic configuration

Field	Content
Number	The extension number selected during creation. This cannot be changed after extension creation.
Friendly name	A custom name for the extension
Port	The device and port the extension has been assigned to, if any. This can only be changed by unassigning and re-assigning the extension (please see 3.4.8 and 3.4.7)

3.4.3.2 Advanced configuration

Field	Content
Voicemail	Enable this to enable voicemail for the extension. <div style="border: 1px solid black; padding: 2px; margin-top: 5px;">Note: Disabling voicemail will reset any call recording settings</div>



Field	Content
	that may have been configured on the extension!
Pin	The user pin used to access voicemail and features
Email	An email address to which voicemail will be emailed
Call waiting	Enables the call waiting feature, allowing an indication to be provided to the user when an incoming call arrives while the user is on the phone
Pickup group	Assign this extension to a pick-up group, allowing other members of the pick-up group to pick up this extension when it rings using a specified feature code (*8). Specify multiple pickup groups using “,” E.g. “1,2,3” In order to allow an extension to be picked up using directed pickup (**) from any group, enter a * in this field.
Account code	The account code to be recorded in the CDR record for calls placed by the extension

3.4.3.3 Configuring Follow me / Divert

Extensions can be configured to forward calls to other extensions in a follow-me list. From the Extensions panel, right-click the extension and select 'Follow me'.

Configure the follow-me settings:

Field	Content
Disable	<p>When checked, the follow-me configuration is ignored and the extension rings as usual.</p> <p>Note: when follow-me is disabled here, directed pick-up of the extension is no longer available.</p>
Initial Ring Time	The number of seconds to ring the extension before processing the follow-me list. If set to 0, the follow-me list is processed immediately (e.g. divert). The extension may be in its own follow-me list, in which case it will also ring.
Ring strategy	<p>Determines the strategy in which extensions in the follow-me list rings. One of:</p> <p>Ring all (v2): Ring the primary extension for the initial ring time, then rings all the</p>



Field	Content
	<p>extensions in the follow-me list simultaneously until one answers</p> <p>Ring all: Rings all the extensions in the follow-me list simultaneously until one answers</p> <p>Hunt: Rings available extensions, one at a time, in turn.</p> <p>Memory Hunt: Rings the first extension in the follow-me list, then the first and the second, then the first, second and third, etc.</p> <p>First available: Rings only the first available extension (if that extension is engaged, but has call waiting, it will ring call waiting) I.e. 'available' includes the concept of call waiting</p> <p>First not on phone: Rings the first available extension for which the phone is not off-hook (i.e. ignores call waiting settings)</p> <p>(strategy)-prim: This setting behaves as the selected strategy described above, how-ever, if the primary extension is engaged or set to do not disturb, the extensions in the follow-me list are not processed.</p>
Ring time	<p>The maximum time to ring phones. If the selected strategy is a hunting strategy, this is the duration of each ring iteration.</p> <p>The maximum value for this field is 60 seconds.</p>
Follow-me list	<p>The list of extensions to process according to the ring strategy. This list may include the primary extension itself and may also include external numbers, suffixed with a #, in which case the number would be called using the applicable outbound route. E.g. including 0115551234# in the follow-me list would call the external number 0115551234</p>
Announcement	<p>An announcement to be played to the</p>



Field	Content
	calling party before processing the follow-me list. Please see section 4.16 for more detail on system recordings.
Play Music on Hold?	If set to Ring, will play ring-tone to the calling party while processing the follow-me list. If set to Default, will play music to the calling party while processing the follow-me list.
CID Name Prefix	An optional CID prefix to display to the ringing extensions.
Alert info	Optional SIP alert info to enable SIP extensions in the follow-me list to ring differently on receiving forwarded calls from the primary extension
Confirm calls	When enabled, extensions in the follow-me list are required to press 1 when picking up the call to confirm that the call has been answered. This is useful for to avoid voicemail on external numbers from picking up the call, when another extension down the follow-me list could still be picked up by a person. This feature is only available for ring-all strategies.
Remote announce	A message to be played to the person answering the call in the case where call confirmation is required. Please see section 4.16 for more detail on system recordings.
Too-late announce	A message to be played to a person answering the call in the case where they have picked up the call, have not pressed 1 to confirm yet, and in the interim someone else has picked up and confirmed. Please see section 4.16 for more detail on system recordings.
Destination if no answer	The destination to route the call to if the ring strategy is unsuccessful in delivering the call (i.e. no-one answers). Please see section 3.7.4 for more detail on call



Field	Content
	destinations.

3.4.4 Logging into phones with an extension (hot-desking)

Users can log into ports that are categorized as free ports on the Extensions panel by using the associated feature codes (the default is *11 to log in and *12 to log out) followed by their extension number and pin. Calls placed from a device connected to such a port will be flagged in CDRs as having originated from the extension number that is logged in to the device.

To make a port that currently has an extension assigned to it available for dynamic login, first unassign the extension from the port, resulting in a free extension and a free port.

Note that in order to log in dynamically to a free device using a free extension, the extension must have a user pin configured.

3.4.5 Configuring SIP extensions

In addition to the basic and advanced configurations (please see 3.4.5), SIP extensions also allows the configuration of various voice codecs in order to provide compatibility with SIP providers and phones.

Codecs supported by a SIP extension can be selected by using the Codecs button. Use Ctrl-Left-Click (holding Ctrl down) to select multiple codecs. Individual codecs can be moved up and down the codec priority list by selecting the codec entry, and selecting the Up or Down buttons. Codecs may specify the integration time in milliseconds after the codec name in the form codec:integrationtime, e.g. 'g729:40'

When negotiating a SIP call, the codec with the highest priority (at the top of the list) will be attempted first, then the second and so on.

3.4.6 Remote SIP extensions

When configuring SIP extensions as remote extensions at a remote site (e.g. home ADSL), if the Com.X is located behind a NAT/firewall on the network, configure the NAT/firewall to forward UDP ports 5060 and 10000-20000 to the Com.X internal IP address.

You may also need to set the external address on the Com.X1 by selecting Start -> Configuration -> Setup and then setting the External address to the dynamic DNS name or the FQDN that resolves to the external (public) address (i.e. the external address should be set to the same as will be programmed into the remote phone).

If you are using dynamic DNS, make sure that the External Is Dynamic field is set to true.

Service might also depend on correctly configuring the SIP phone.



3.4.7 Assigning extensions

Free extensions can be assigned to a port by editing the extension, and selecting the Port button. From there a menu of available extension ports can be accessed. Only available ports are listed in the menus.

3.4.8 Unassigning extensions

Extensions' associations with ports can be removed by selecting the extension in the extensions list, and selecting Unassign from its context menu.

3.4.9 Configuring multiple extensions

Multiple extensions can be configured simultaneously. When the list of extensions for configuration include both analogue and SIP extensions, only those fields that are not unique to any individual extension can be configured:

When only SIP extensions are included in the list, only those fields not unique to any of the SIP extensions can be configured. This includes the SIP codec configuration.

Note that, since multiple extensions may have different values for shared fields, these fields default to disabled during a multiple-edit, i.e. the form is not a reflection of the underlying configuration when first opened.

Simply accepting this form will set all the extensions' values to disabled. Cancelling the form will leave all extensions' configurations unchanged. Enabling one of the fields will enable the field for all of the extension being edited.

3.4.10 Deleting extensions

Extensions may be deleted individually, or as a group, by selecting Delete from the extension context menu.

Note: Deleting an extension that is in use in an inbound route will result in an update to the route to terminate with hangup.

3.5 Trunk groups and VOIP providers

3.5.1 Listing existing trunk groups and VOIP providers

To view the list of configured trunks, select select Start, followed by Configuration and then the Trunks Panel. The list is populated with analogue trunks in the top section and VOIP providers in the lower section of the panel.

3.5.2 Analogue trunk groups

Analogue and BRI trunks can be grouped to facilitate simplified routing. PRI channels are always grouped into a trunk per PRI. Each analogue and BRI trunk may only belong to one group, of the same type as the trunk (analogue or BRI).



3.5.2.1 Adding trunk groups

A new trunk group may be created by selecting the Add, selecting the New Group option from the Trunk Groups node.

A new group may also be created by selecting trunks of the same type, and selecting New Group from the context menu.



Field	Content
Name	Unique identifier for this group. Analogue groups may only be numbers from 1 – 62, and BRI groups may be any unique combination of alpha-numeric characters.
Type	The group type. Only trunks of this type may be included in the group. Allowed types are: <ul style="list-style-type: none">•Analogue trunk (FXO)•ISDN BRA (BRI) When the group was created from a selection of trunks, the type will automatically be selected

After creation, the new group and any ports belonging to it will be listed under the Trunk Groups node.

3.5.2.2 Adding trunks to a group

Trunks may be added to a group either by selecting the trunks and creating a group using them, as detailed in 3.5.2.1, or by selecting the trunks and then selecting Add to group from the context menu and selecting an existing group of the same type. Only groups of the same type will appear as options for selection.

After being added to a group, the trunks under the Trunk Ports node will show the group name they belong to in paranthesis.

3.5.2.3 Removing trunks from a group

To remove a selection of trunks from a trunk group, find the group under the Trunk Groups node and select the trunks (using Ctrl-left-click to select multiple). Select the Remove from Group option from the context menu. The turnks will be removed from the group and the reference to the group removed from the trunks under the Trunk Ports node.

Note: You will not be prompted to confirm this action.

3.5.2.4 Removing a trunk group

To completely remove a trunk group and remove the group association from all the trunks included in the group, select the group in the Trunk Groups list and select the Remove Group option from the context menu. The turnks will be removed from the group and the reference to the group removed from the trunks under the Trunk Ports node, and the group will be removed from under the Trunk Groups node.

Note that deletion of a group may affect routing. The group will be removed from routes that made use of it, potentially resulting in routes with no trunks associated.



3.5.2.5 Viewing group members

Membership of trunks to their respective groups may be observed either by expanding individual devices listed under the Trunk Ports node, where group membership is denoted in parenthesis after trunks, or by selecting the desired group in the group listed under the Trunk Groups node, where the devices and trunks included in the group are listed:

3.5.3 SIP trunks

Multiple SIP trunks can be configured for call routing. SIP trunks can also act as SIP providers, providing the capability to configure back-to-back SIP trunks for inter-office connections.

3.5.3.1 SIP call setup

SIP uses a peer to peer mechanism for call setup. In order for a call to be established, two peers need to be defined and be able to communicate with each other. This necessitates a SIP trunk entry for each peer of a given system.

Communication between two peers implies that the peer initiating the call knows the IP address of the recipient, and that the recipient can match the caller's credentials in order to allow and configure the call.

The peer initiating the call can either have the recipient's IP address configured, or may know the recipient's IP address as the result of the recipient registering with the it (i.e. the calling peer is acting as a provider.) When acting as a provider, a peer allows registration of a remote peer by matching the incoming registration request's username with the provider's SIP trunk name. At this point the provider's SIP trunk moves from having an UNKNOWN peer to being associated with the registered IP address.

An incoming call is matched by the recipient by looking at the incoming call's IP address and FROM field, which defaults to the caller ID of the caller, or the calling peer's from-user field if configured. The recipient first looks at the caller's FROM field, and if this matches the username field of a peer that has "Identify by user" enabled or a configured extension, a successful match is made.

If no such match can be made, the recipient matches the first SIP trunk entry that is associated with the IP address of the caller. This implies that, if multiple SIP trunks are associated with peers with the same IP address, they have to have "Identify by user" enabled and a username specified, and the remote peers have to have their from-user fields set to the appropriate username.

If a successful match is made and the password and codecs that are configured for that matched peer are compatible with those in the call setup request, the call is established.

This mechanism is used for call setup in both directions.

3.5.3.2 Adding a SIP trunk

To add a SIP provider, activate the Add menu on the Trunks panel. Select SIP Provider.



3.5.3.3 Sip trunk configuration

SIP trunk configuration is divided into basic, advanced and codec configuration, accessible from their accompanying tabs on the SIP Provider form.

Field	Content
Name	The unique name of this SIP trunk. If this trunk is a SIP provider, the name of the trunk must match the username.
Description	A custom description of this SIP trunk
Registrar	The address of the SIP provider's registration server
Username	<p>The registration user name provided in the registration string to the SIP provider when registering. When registering, the username must be the same as the username configured in the SIP provider.</p> <p>When not registering, if "Identify by username" is enabled, this username is used to match incoming SIP calls. I.e. the from-user field for the remote peer must be set to this value.</p>
Password	The registration password required by the SIP provider
Act as Provider	<p>When enabled, this trunk allows other SIP users to register with it. An UNKNOWN peer will be created and the IP address of the peer associated with this trunk on registration.</p> <div style="border: 1px solid black; padding: 5px;"><p>Note that when configuring a Com.X to Com.X SIP trunk, if the registering trunk is configured and applied on the registering Com.X before the SIP provider configuration is applied on the Provider Com.X, the registering box will need a 'sip reload' to initiate registration. We therefore recommend configuring and applying the provider trunk before the registering trunk.</p></div>
Register with Provider	When enabled, the SIP trunk will attempt to register its IP address with the SIP host at the provider address configured. The registration string will use the configured username.
Identify by username	When enabled, will use the username field to identify a SIP client. This is necessary when multiple SIP trunks are registered with the same SIP provider from the same IP address and allows the provider to determine the incoming trunk from the SIP FROM field provided. The remote SIP peer's from-user field needs to match this trunk's name and username field if enabled. If bi-directional call establishment is required, ensure the SIP trunks have the same name on both Com.X1s.



Field	Content
Proxy	The SIP proxy to use when establishing outgoing connections in peer to peer setups. This will in most cases be left blank when registering with a service provider, although some service providers may require registration via a SIP proxy.
From user	Defaults to the extension number of the initiator of the SIP call. If the provider rejects the registration, the from user might need to be set to the provider's registration / login number / username. The from-user field only affects calls, and not trunk SIP registration. This fields overrides the caller ID number sent in the SIP headers.
NAT	When enabled, the system ignores the SIP and SDP headers' address and port and replies to the sender's address and port. This should be enabled when the Com.X is located behind a NAT router.
Secure	If this is setting is enabled, after registration with the SIP provider, all invites will require successful authentication as well as port number matching. Disabling this settings allows invites to proceed without further security requirements after successful registration.
Qualify connection	When enabled, checks the reachability of the peer every 60 seconds. <div style="border: 1px solid black; padding: 5px;"><p>Note that SIP trunk qualification is an additional (non-standards-based) feature which allows for monitoring of SIP trunks. Not all SIP providers support this additional feature. For providers that do not support this add-on feature, the standard SIP trunking configuration is recommended (Qualify turned off)</p></div>
Contact # for incoming	A DID number to assign to incoming calls on this SIP trunk. This facilitates inbound routing where the DID is not provided by the SIP provider.
Maximum # channels	The maximum number of simultaneous SIP channels supported on this trunk. Leave blank for no limit.

Codecs supported by a SIP trunk can be selected by using the Codecs button. Use Ctrl-Left-Click (holding Ctrl down) to select multiple codecs. Individual codecs can be moved up and down the codec priority list by selecting the codec entry, and selecting the Up or Down buttons.

When negotiating a SIP call, the codec with the highest priority (at the top of the list) will be attempted first, then the second and so on. Codecs may specify the integration time in milliseconds after the codec name in the form codec:integrationtime, e.g. 'g729:40'



3.5.3.4 Editing a SIP trunk

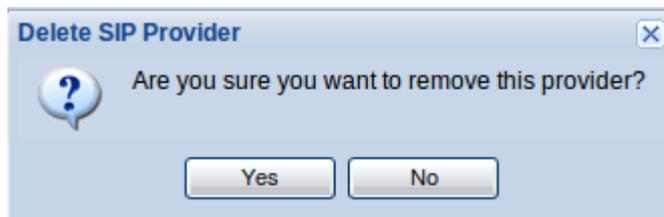
To update the configuration of a SIP trunk, select the Edit Provider option from the trunk's context menu.

3.5.3.5 Removing a SIP trunk

To remove a SIP trunk, select Remove from the Sip Provider node's context menu.

Note that removing a provider will remove the SIP trunk from all routes that are using it, possibly resulting in routes with no trunks associated.

Note that if the trunk was a provider, with active registrations from clients, the clients need to re-register, or their registrations will remain invalid until they time out.



3.5.4 Inter-Asterisk Exchange (IAX) trunks

IAX is the native Asterisk PBX server to server communications protocol designed to work well across firewalls and NAT.

Multiple IAX trunks can be configured for call routing. IAX trunks can be configured as providers, or as peers registering with providers.

IAX supports multiplexing of channels over a single link. Data from multiple calls are merged into a single stream between two endpoints, reducing the IP overhead without creating additional latency.

If the Com.X is located behind a NAT/firewall on the network, configure the NAT/firewall to forward UDP port 4569 to the Com.X internal IP address.

3.5.4.1 Adding an IAX trunk

To add an IAX provider, activate the Add menu on the Trunks panel. Select IAX Provider.

3.5.4.2 IAX trunk configuration

IAX trunk configuration is divided into basic, advanced and codec configuration, accessible from their accompanying tabs on the IAX Provider form.

Field	Content
Name	The unique name of this IAX trunk. If this trunk is an IAX provider, the name of the trunk must match the username.



Field	Content
Description	A custom description of this IAX trunk
Registrar	The address of the IAX provider's registration server
Username	The registration user name provided in the registration string to the IAX provider when registering. When registering, the username must be the same as the username configured in the IAX provider.
Password	The registration password required by the IAX provider
Act as Provider	<p>When enabled, this trunk allows other IAX users to register with it. An UNKNOWN peer will be created and the IP address of the peer associated with this trunk on registration.</p> <p>Note that when configuring a Com.X to Com.X IAX trunk, if the registering trunk is configured and applied on the registering Com.X before the IAX provider configuration is applied on the Provider Com.X, the registering box will need a 'iax2 reload' to initiate registration. We therefore recommend configuring and applying the provider trunk before the registering trunk.</p>
Register with Provider	When enabled, the IAX trunk will attempt to register its IP address with the IAX host at the provider address configured. The registration string will use the configured username.

Field	Content
Qualify connection	When enabled, checks the reachability of the peer every 60 seconds.
Contact # for incoming	A DID number to assign to incoming calls on this IAX trunk. This facilitates inbound routing where the DID is not provided by the IAX provider.
Maximum # channels	The maximum number of simultaneous IAX channels supported on this trunk. Leave blank for no limit.

Codecs supported by an IAX trunk can be selected by using the Codecs button. Use Ctrl-Left-Click (holding Ctrl down) to select multiple codecs. Individual codecs can be moved up and down the codec priority list by selecting the codec entry, and selecting the Up or Down buttons.

When negotiating an IAX call, the codec with the highest priority (at the top of the list) will be attempted first, then the second and so on. Codecs may specify the



integration time in milliseconds after the codec name in the form `codec:integrationtime`, e.g. 'g729:40'

3.5.4.3 Editing an IAX trunk

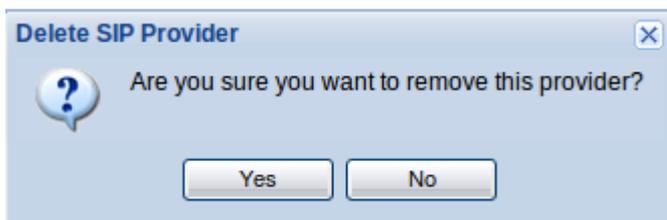
To update the configuration of an IAX trunk, select the Edit Provider option from the trunk's context menu.

3.5.4.4 Removing a SIP trunk

To remove an IAX trunk, select Remove from the IAX Provider node's context menu.

Note that removing a provider will remove the IAX trunk from all routes that are using it, possibly resulting in routes with no trunks associated.

Note that if the trunk was a provider, with active registrations from clients, the clients need to re-register, or their registrations will remain invalid until they time out.



3.6 Outbound route configuration

A variety of trunk types can be combined and prioritised to form outbound routes, routing calls based on dial-patterns. The Com.X supports routing to trunk groups, as well as individual FXO trunks and BRI channels.

3.6.1 Viewing configured outbound routes

The list of existing configured outbound routes can be accessed by selecting Start, followed by Configuration and the Outbound panel. For each route listed, the route's name, the pin code and an indication of whether the route is an emergency and / or intra-company route is displayed. Routes can be managed from their context menu.

To prioritise one route above another, select the Up and Down options in the context menu. Routes appearing first in the list have the highest priority. Dial pattern matching tries each route in turn for a match, starting with the highest priority route and progressing to the last route in the list.

3.6.2 Adding outbound routes

New routes can be added by selecting the Add button on the Outbound panel. For the route configuration, please see 3.6.3.

3.6.3 Configuring outbound routes

Field	Content
Name	A unique name for the route.



Field	Content
Pin	If set, after a user has dialed a number that matches the dial pattern of this route, the user will be prompted to enter the pin code in order for the call to proceed.
Emergency	When enabled, emergency dialing indicates that the emergency CID set on a device or port should override all other CID settings and be sent with the call information.
Intra-company	When enabled, the CID sent with the call information is the internal, local CID, and not the outbound CID of the extension or trunk.

3.6.4 Dial Patterns

3.6.4.1 Adding dial patterns to an outbound route

The Dial Patterns for Outbound routes are analyzed, and the highest priority route for which a pattern match is made is tasked with routing the call. New dial patterns can be added to a route by selecting New from the context menu of the Dial Patterns section of the route. The route has to be edited first before this option becomes available. If no existing patterns are included in the route, the context menu can be accessed by right-clicking the Dial Pattern header.

3.6.4.2 Dial pattern syntax

Dial patterns consist of the following characters only:

0 1 2 3 4 5 6 7 8 9 X Z N | . ! [] , -

X matches any digit from 0 – 9

Z matches any digit from 1 – 9

N matches any digit from 2 – 9

| separates a dialing prefix from numbers dialed, e.g. 123|4567 would match the dialed number 1234567, but would only send 4567 on to the trunks.

. matches one or more digits, e.g. 0. would match a 0 followed by one or more digits

! provides legacy overlap dialing support and matches as soon as it can unambiguously, without waiting for the dialing to complete, allowing subsequent digits to be transferred directly to the trunk (only supported on PRI channels).

[] allows a collection of digits to be matched, e.g. [7-9] would match 7, 8 and 9

- indicates a range of sequential numbers to match

+ in conjunction with | allows insertion of digits into the dial string, e.g. to change 0114567890 to 27114567890 use dial rule '27+0|X.' in the Dial Rules section on the trunk from the Advanced menu



3.6.4.3 Removing Dial Patterns from a route

A Dial Pattern can be removed from a route by selecting Remove from the pattern's context menu.

3.6.5 Adding trunks to an outbound route

Trunks can be added to a route by selecting Add from the context menu of the Trunk Priorities section of the route. The route has to be edited first before this option becomes available. If no existing trunks are included in the route, the context menu can be accessed by right-clicking the Trunk Priorities header.

A list of available trunks, including analogue trunks (FXO), BRI trunks, PRI, SIP providers and Trunk Groups will be displayed. Multiple trunks and groups can be selected for addition by using Ctrl-Left-Click.

Once added, trunks can be prioritized by selecting a trunk and selecting Up or Down from the trunk's context menu. Calls will be attempted on the trunks in the order that they appear in this list.

3.6.6 Removing Trunks from a route

A Trunk can be removed from a route by selecting Remove from the trunk's context menu.

3.6.7 Deleting outbound routes

To delete an outbound route, select Delete from the route's context menu.

3.6.8 Outbound route priority

Outbound routes can be moved up and down the overall list by right-clicking and selecting Up or Down. An outbound number that matches a pattern in more than one route will be routed via the highest priority route.

3.7 Inbound route configuration

Inbound routes allows incoming calls to be routed to a particular destination based on the incoming call's DID and CID number pair. The Com.X supports a rich set of destination types, greatly simplifying incoming call routing.

Note: The Com.X will automatically try and find extension numbers that match incoming DIDs. E.g. if the system receives a call from a BRI link with DID 9031, and an extension 9031 exists, no inbound route is necessary. The call will automatically be routed to extension 9031.

3.7.1 Viewing existing inbound routes

This list of configured inbound routes can be viewed by selecting Start, followed by Configuration and the Inbound panel. For each configured inbound route, the DID and CID pair used to match the route, the route description, destination type and destination are displayed. To manage inbound routes, select an option from the desired route's context menu.



3.7.2 Adding a new inbound route

To add a new inbound route, select the Add button on the Inbound panel.

3.7.3 Configuring inbound routes

Field	Content
DID	<p>Direct Inward Dialing allows a number of incoming numbers to be serviced across a limited number of physical connections from the telco to the PBX. This field may contain a pattern to be matched against an incoming number. Please see 3.6.4.2 for more on the dialed pattern syntax. If left blank, this route will match any dialed number or if no DID information is present.</p> <p>Be careful of using a single '.' as the pattern for an inbound route, as this might also match dial plan constructs such as fax and hangup and may result in unexpected behaviour. Rather use 'X.'</p> <p>The DID and CID together form a pattern matching pair which must be unique amongst the list of inbound routes. If this rule matches the DID and CID best, the route is used to route the call.</p> <div style="border: 1px solid black; padding: 5px;"><p>MSN note: DID numbers typically are sequential, and so the telco might only signal a subset of the DID numbers, e.g. the last 4 digits. MSN numbers are not guaranteed to be consecutive. As a result the MSNs signalled by the telco might be longer than the DIDs. The full MSN number arriving at the PBX needs to be matched in order to process the call.</p></div>
CID	<p>Caller ID number is the caller's number sent to the PBX by the telco. This field may contain a pattern to be matched against an incoming caller ID. Please see 3.6.4.2 for more on the dialed pattern syntax. If left blank, this route will match any caller ID or if no CID information is present.</p> <div style="border: 1px solid black; padding: 5px;"><p>Note: The DID and CID together form a pattern matching pair which must be unique amongst the list of inbound routes. If this rule matches the DID and CID best, the route is used to route the call</p></div>
Description	A custom description describing the inbound route
Destination Type	Extension, Day/Night, Conference, Terminate, Ring Group, Voicemail (busy), Voicemail (no message), Voicemail (unavailable)
Destination	The destination extension, conference extension, termination destination (busy, congestion, hangup, on hold forever, ring, sit ring), ring group number, voicemail extension



3.7.4 Configuring call destinations

3.7.4.1 Conferences

To configure conferences, select Advanced from the main menu bar and Conferences. Once configured, the destinations will become available selections in the Inbound Routes destination menus.

3.7.4.2 Day/Night

To configure day/night inbound routing modes, select Advanced from the main menu bar and Day/Night Control. Once configured, the destinations will become available selections in the Inbound Routes destination menus.

3.7.4.3 Extensions

To configure conferences, select Start from the main menu bar, followed by Configuration and the Extensions panel. Please see 3.4 for more detail. Once configured, the destinations will become available selections in the Inbound Routes destination menus.

3.7.4.4 Ring Groups

To configure ring groups, select Advanced from the main menu bar and Ring Groups. Once configured, the destinations will become available selections in the Inbound Routes destination menus.

3.7.4.5 Termination

The following termination options are available for inbound routing:

- busy: a busy tone is provided to the caller
- congestion: an 'All circuits are busy now' message is provided to the caller
- hangup: hang up the call
- on hold forever: places the call on hold without termination
- ring: provides the caller with a ring tone
- sit ring: provides the caller with a specific ring tone.

3.7.4.6 Voicemail

To choose Voicemail for an extension as a destination, select Start from the main menu bar, followed by Configuration and the Extensions panel. Select the desired extension and edit the extension configuration. Please see 3.4 for more details. Only extensions that have voicemail enabled will be available for voicemail termination.

3.7.4.7 IVR

Interactive Voice Response systems allow the playback of a set of audio messages and the transfer of calls based on user interaction via the phone keypad. To select an IVR, first configure an IVR entry and detailed in 4.7 and then select it here.



3.7.4.8 Phonebook

To choose an entry in the phonebook as a destination, first configure the phonebook entries and then select the appropriate entry from the options menu for the phonebook destination.

3.7.5 Deleting an inbound route

To delete an inbound route, select Delete from the route's context menu.

3.8 Flex Path configuration

The innovative Com.X Flex Path feature is extremely powerful and allows flexible routing, trunk to trunk routing, gateway and class of service capabilities. Flex paths allow grouping of call sources into contexts (paths) which can be associated with outbound routes, inbound routing tables, transformations, destinations, etc.

To access Flex Path, select Configuration from the Start menu, and then the Flex Path panel.

3.8.1 Flex Path Route Types

3.8.1.1 Include

Include routes allow the inclusion of all, or part of the Inbound Routes configured in the system. A calls originating from ports in a flex path that has includes will, if no other route entries in the flex path match the call source, be forwarded to the route destinations specified in the order of includes.

Inclusion type	Description
Inbound DID	Includes inbound rules that match incoming DIDs
Inbound Direct	Includes configured extensions only (including services that have extensions, such as queues and groups). Excludes feature codes. This type forms a bridge between DIDs and internal extensions. If a call is inbound with a DID that matches an extension, having Inbound Direct enabled allows the call to be routed directly to the extension without the need for an inbound route.
Internal	Includes configured extensions and feature codes, as well as the phonebook.

Outbound destinations can also be included, as well as another Flex Path. A Flex Path can include only one additional Flex Path. If multiple Flex Paths need to be in the include chain, chain them together by including the second in the first, the third in the second and so forth. Though multiple Flex Paths includes in a Flex Path are allowed, only the first will be processed.



3.8.1.2 Match

Matching routes allows calls to be routed based on DID and CID matches. This functionality differs from normal inbound routing in that, upon match, the DID can be transformed, routed to a subset of inbound routes, or routed to an outbound route (i.e. trunk to trunk routing.)

3.8.1.3 Start

Start routes are special cases of Match routes and match when no DID are available. These routes are used to assign DIDs to originating calls when no DID information is present. Start routes will only match if no DID are available. If it is possible that a DID will be present, include a Match rule as well.

3.8.2 Flex Path Destinations

Calls originating from ports that have been included in a flex path may be routed to the following destinations:

Destination	Description
Flex Path	Existing flex paths. Beware of circular references. Once a call has been routed to another flex path, the matching process starts with the first route in the destination flex path.
Other	Internal extensions, feature codes and existing inbound routes. Please see 3.8.1.1 for more detail.
Outbound	Existing outbound destinations.

3.8.3 Viewing existing Flex Paths

To view existing Flex Paths, select Start – Configuration – Flex Path.

For each flex path, the path name and the current routes included in the path are displayed. For each route, the route name, DID pattern match, CID match, description, transform, destination type and destination are shown if available.

Routes are processed in the order in which they are listed in a flex path. Paths are sorted in alphabetical order.

3.8.4 Adding Flex Paths

To add a new flex path, select the Add Path button on the flex path panel.

Field	Content
Name	Unique name for the flex path. Spaces are not allowed in the name.



Field	Content
Description	Custom description for the flex path.

3.8.5 Deleting Flex Paths

To delete a flex path, select “Delete” from the path's context menu. Confirm deletion to delete.

Note that port assignments that have been made to this flex path will become “invalid”, resulting in no service for calls originating on such ports. Service to these ports can be restored by assigning the ports either to the “default” flex path or to another flex path.

3.8.6 Adding routes to a Flex Path

To add routes to a Flex Path, select “Add Route” from the path's context menu.

Field	Content
Routing mode	One of Include, Match or Start. Please see 3.8.1 for more detail on flex path route types
Pattern	The DID pattern that this route will match
CID	The CID pattern that this route will match
Description	Custom description for this route
Transform	If specified, the originating call's DID will be transformed as configured. Please see 3.8.10.8 for more detail on transformations.
Destination	The routing destination. Please see 3.8.2 for more detail on flex path destinations.

3.8.7 Changing route priority in a Flex Path

Routes that have been added to a flex path are processed in the order they appear in the flex path. Once a route has matched in a flex path, other routes are ignored. If no Match or Start routes match, Include routes match in the order they are listed.

The priority of a route in a flex path's match list can be changed by selecting “Move up” or “Move down” from the route's context menu.

3.8.8 Editing routes in a flex path

To edit a route in a flex path, select “Edit” from the route's context menu. Please see 3.8.6 for more detail on route configuration.



3.8.9 Deleting routes from a Flex Path

To delete a route from a flex path, select “Remove” from the route's context menu.

Note that empty flex paths are valid. A port assigned to an empty flex path will not be able to route calls.

3.8.10 Flex Path use cases

3.8.10.1 Class of service

Any source of a call in the system can be pre-processed using a Flex Path. This allows class of service control on both outgoing and incoming calls.

Note: To block an extension from making calls, place it in an empty Flex Path (i.e. a Flex path with no routes.)

Note: To allow phones to only make internal calls and limit outbound calls to the phonebook, create a Flex Path with only one route. The route should contain one include mode with destination 'Internal'.

Note: Assigning analogue phones to class of service Flex Paths results in service loss while apply is in progress. Assignment of SIP phones to Flex Paths can be applied with no break in service.

Scenario

- Extension 301 is required to have access to internal and national dialing only.
- Extension 302 is required to have access to internal, national and international dialing.

Configuration:

Extensions:

- Two SIP extensions are configured:

Two outbound routes:

- national access
- international access

Two Flex Paths:

- pathNational includes the national outbound route and the Internal include.
- pathInternational includes both the national and international outbound routes as well as the Internal include.

Port assignments:

- 301 is assigned to pathNational
- 302 is assigned to pathInternational



3.8.10.2 Partitioning / multi-tenant

Scenario:

Company A and Company B share a PBX. Each company has reserved a set of BRI channels for their exclusive use.

Company A: 4 bri channels on bri ports d1 and d2

Company B: 4 bri channels on bri ports d3 and d4

Extensions 8xxx belong to Company A with reception at 8000.

Extensions 9xxx belong to Company B with reception at 9000

Outgoing calls must be routed to the companies' BRI channels.

Incoming calls on the companies' BRI lines must be routed to their reception.

Configuration:*Extensions:*

- Extensions ports (both FXS and SIP) are assigned the appropriate extension numbers for each company.

Two outbound routes:

- companyAout routes over Company A's BRI channels
- companyBout routes over Company B's BRI channels

Two Inbound routes:

- Fixed incoming DID 8000 is routed to Company A's reception. This DID will be assigned to BRI calls originating from Company A's BRI channels in a Flex Path
- Fixed incoming DID 9000 is routed to Company B's reception. This DID will be assigned to BRI calls originating from Company B's BRI channels in a Flex Path

Flex Paths:

- companyA-in: Add a Start route that transforms incoming BRI calls without DIDs to the reception extension and forward these to the Inbound routes.
- companyA-in: Add a Match route that transforms incoming BRI calls with any DID to the reception extension and forward these to the Inbound routes.
- Assign Company A's BRI trunk groups to the companyA-in flex path
- companyA-out: Include an internal route that facilitates dialing internal extensions
- companyA-out: Add a rule that matches outgoing numbers other than extensions and routes them to the companyAout outbound route.
- Assign Company A's extension ports to the companyA-out flex path
- companyB-in: Add a Start route that transforms incoming BRI calls without DIDs to the reception extension and forward these to the Inbound routes.



- companyB-in: Add a Match route that transforms incoming BRI calls with any DID to the reception extension and forward these to the Inbound routes.
- Assign Company B's BRI trunk groups to the companyB-in flex path
- companyB-out: Include an internal route that facilitates dialing internal extensions
- companyB-out: Add a rule that matches outgoing numbers other than extensions and routes them to the companyBout outbound route.
- Assign Company B's extension ports to the companyB-out flex path

Note: Inter-company calls are facilitated by default and can be accessed by directly dialing a company's extensions (e.g. the receptionist for Company B could dial 8000 to reach the receptionist for Company A.) To disable this, create empty Flex Paths that terminate service for cross-company numbers:

- Company A: Create an empty Flex Path that Matches dial pattern 9XXX
- Company B: Create an empty Flex Path that Matches dial pattern 8XXX

Note: This is a simplified example. Flex Path could also be used to match incoming DID ranges and route these to separate inbound routes for different partitions.

Note: The incoming CID can also be prefixed with a custom name for inbound routes. To configure this, access the FreePBX Inbound Route configuration (please see section 6.5) and configure the 'CID name prefix' field.

3.8.10.3 Simple partitioned DID routing:

If DIDs are provided by the telco, the use of inbound flex paths are unnecessary and, provided the extension numbers created for each company matches the DIDs provided over by the Telco, calls will automatically route to the correct extensions.

Scenario:

DID ranges routed by the Telco as follows:

Company A: 5559000 – 5559049 -> ports d1 and d2

Company B: 5559050 - 5559099 -> ports d3 and d4

Telco communicates the last 4 digits of DID across channels during call setup.

Configuration:

Creating extensions 9000 – 9049 for Company A and 9050 – 9099 for Company B assigned to the appropriate FXS and SIP ports will result in automatic call routing to the correct extensions for incoming calls.

3.8.10.4 Trunk to Trunk gateway

Scenario:

A company has a head office in a major city, and a remote office in another city. The company has an agreement with an internet connectivity provider for a high quality IP link between the two offices. The company requires a PBX system at the head office,



with internal extensions at both office locations, the two offices SIP trunked and PSTN trunks provided for call routing at the head office location.

Configuration:

Both the head office and the branch office are provided with Com.X units.

Extensions:

- Head office extensions configured as 101 – 105
- Branch office extensions configured as 201 – 205

Sip trunk configuration:

- Head office SIP trunk configured as provider
- Branch office SIP trunk configured to register with head office
- Ensure that Proxy, NAT, connection qualification and codec selection options are configured appropriately.

Outbound route configuration:

- Head office outbound route routes branch extensions across the SIP trunk
- Head office outbound route routes all calls to PSTN
- Branch office outbound route routes all calls (including head office extensions) to head office across the SIP trunk

Flex Path:

- Head office configured with a Flex path that places calls originating from the SIP trunk either internally or routes them trunk-to-trunk to the PSTN outbound route.

3.8.10.5 Legacy PBX Remote extension gateway**Scenario:**

A company has a head-office with 4 FXO trunks to the telco and 10 analogue extensions on a legacy PBX. The legacy PBX has a number of unused extension and trunk ports available. The company is opening a second branch, geographically separated from the first, and desires a stand-alone PBX at the branch, with its own BRI telco interfaces, but the ability for head-office and branch to phone one another using extension numbers across a SIP trunk. There will be four extensions required at the branch, with potential for some growth.

Configuration:

Install a Com.X1 at both sites. For the branch, the Com.X1 serves as a PBX, with the BRI outbound routes routing across the BRI interface.

In the head-office, place the Com.X1 behind the legacy PBX, connecting its FXO ports to free FXS ports on the legacy PBX and configuring the SIP trunk to the branch.

Extensions:



In this configuration, the branch extensions look like additional extensions on the legacy PBX, but the Com.X1 gateways these across the SIP trunk to the branch.

Calls from the branch to head-office are routed across the SIP trunk, prepended by caller ID, and the Com.X1 at the head-office then routes based on the branch CID to the appropriate FXO port, in effect using the correct legacy PBX extension for activity from the associated branch extension.

Flex Path:

- At the branch, create a Flex Path with a Match rule that matches head-office extensions with an outbound route as destination and the following transform: CD.
- At the head-office, add Flex Paths with Match rules that match the DIDs transformed by the branch and routes them based on the received CID number in the DIDs to the correct FXO port.
- At the head office, assign DIDs to the Com.X1 FXO ports and add a Flex path that pre-pends this DID to the dialed number before routing to the branch across the SIP trunk.
- At the branch, add Flex Paths with Match rules that routes to the appropriate extensions based on the DIDs received from the head-office.

3.8.10.6 Legacy PBX feature set extension

Scenario:

A company has a significant investment in a legacy analogue PBX. This investment includes some features, e.g. voicemail, routing, groups, feature codes and IVRs that the company would like to preserve.

They would how-ever like to extend the legacy PBX with new SIP extensions and additional features, including call recording, least cost routing and additional SIP-based features for the new SIP phones, and possibly enable the PBX to support a remote office as described in section 3.8.10.5. The existing PBX cannot be extended, or the cost associated with such extension is prohibitive.

As a result, the company want existing analogue phones to be served dial-tone directly from the existing PBX (preserving the existing feature set) and the new features and LCR to be available to new SIP extensions. Call recording should be available on all extensions, and it should be possible to block extensions from dialing.

Configuration:

Install a Com.X solution with sufficient analogue FXS and FXO ports for the company. The Com.X solution provides SIP extensions, LCR and the other features mentioned to the business.

Connect the analogue phones to the Com.X solution's FXS ports and the Com.X solution's FXO ports to the existing PBX's FXS ports.

For each FXO port, assign a DID matching the legacy PBX's extension. Create an extension with the same extension number for each of the legacy PBX's extensions.

For each of these FXS ports, create an outbound route with dial pattern '!' (please see section 3.6.4.2) and a Flex path with the extension FXS port as the source, and



destination the associated outbound route. Enable immediate dialing on the extension port (please see section 3.3.5.2)

Using this configuration, when an analogue phone goes off-hook, the legacy PBX would be providing dial-tone directly. If the legacy PBX's extension rings, the associated Com.X extension also rings. This enables calls to be recorded on the legacy analogue extensions. Though dialled numbers cannot be recorded in CDR records for the legacy extensions (since the number is not collected by the Com.X solution but rather passed straight through to the legacy PBX), the duration of calls originating from legacy extensions can be recorded. Extensions can also be blocked. Call recording would be fully featured for the new SIP extensions.

3.8.10.7 Immediate dialing

Emergency and single-purpose devices can be configured to dial immediately on pick-up.

Scenario:

An emergency phone is required to dial 911 when picked up.

Configuration:

Extension:

- From the Hardware panel, edit the port to which the device is connected and enable "Immediate mode".

Flex Path:

- Create a Flex Path that includes a Start route, configured to transform the DID to 911
- Assign the emergency port to the Flex Path.

3.8.10.8 Transformations

The Com.X allows both outgoing and incoming dialing numbers to be transformed.

To transform, create a Flex Path and add a Start or Match route to the path, with the required transform specified.

Transforms may contain digits 0 – 9 and special placeholders D, which is replaced with the current DID and C, which is replaced with the current CID.

Scenario:

All outgoing calls must be prepended with 08 and followed by 55 and the CID, if any.

Configuration:

- Add a Flex Path with a Match rule with an outbound route as destination and the following transform: 08D55C
- Assign extension ports that need this transform to the Flex Path.

Scenario:

All calls incoming from a SIP trunk needs to be prepended with 77.

Configuration:



-
- Add a Flex Path with a match rule that includes inbound DID routing transforms as follows: 77D
 - Assign the SIP trunk to the Flex Path.

3.8.10.9 Assigning DIDs to BRI ports

When DIDs are not provided over BRI channels, DIDs can be associated directly with BRI ports in the Com.X to facilitate call routing.

Scenario:

DIDs 9001 – 9004 need to be associated with 4 BRI ports.

Configuration:

- Create a Trunk Group for each BRI port. BRI ports cannot be assigned to Flex Paths directly.
- Create Flex Path for each BRI port
- Add Start routes to the Flex Paths that transform to the required DIDs
- Set the destinations of the Start routes to Inbound DIDs for inbound call routing.
- Assign the BRI ports to their specific Flex Paths.

3.8.10.10 Load-balancing calls

Outbound calls can be load-balanced across telco interfaces or network routes using FlexPath.

Scenario:

Company A has 8 extension, and wishes to load-balance outgoing calls across two DSL links, each supporting up to a maximum of 4 simultaneous calls. The DL routers have Ips 192.168.0.1 and 192.168.0.2. The company's VOIP service provider has provided the company with the IP addresses of two SIP proxies: 10.0.0.1 and 10.0.0.2

Configuration:

- Add two SIP trunks, with distinct IP address end-points (in this scenario 10.0.0.1 and 10.0.0.2) These may be two proxies with the same provider, but for IP routing capability, the IP addresses must be different. When routing across telco interfaces, one would replace the SIP trunks here with trunk groups that contain the trunks to be used.
- Create two outbound routes, one using the first SIP trunk and the other the second SIP trunk (or the appropriate telco trunk groups).
- Create two Flex Paths, and add an Include rule that includes the appropriate outbound route for each. Point the first flex path to the first outbound route and the second to the second outbound route.



- For network load-balanced scenarios, log into the Com.X using ssh and configure two routes for routing the traffic across the two network paths in /etc/network/interfaces:

```
auto eth0
iface eth0 inet dhcp
post-up ip route flush dev eth0
post-up ip route add 10.0.0.1 via 192.168.0.1
post-up ip route add 10.0.0.2 via 192.168.0.2
```

3.8.10.11 FXO Trunk pick-up to test for trunk service

When installing a Com.X, at times it might be useful to determine whether a specific trunk port has been service provisioned by a device connected to the port. For example, the installer might want to check whether a legacy PBX is providing dial-tone in the trunk, or a premi-cell is available. This might also be of use if the wiring to the Com.X1 is not clearly marked, to determine which trunks have service.

Configuration:

- create an outbound route called 'trunk-tester' with dial pattern '99!' and the trunk of interest included in the trunks list
- Dial 99

Configure for automatic test on pickup:

- Replace the dial pattern above with '!'.
- create a flex path called 'trunk-tester' and use the Include mode to include the 'trunk-tester' outbound route
- on the Flex Path panel, assign an extension port to the trunk-tester flex path and plug a phone into the extension port
- select the extension port in the Hardware panel and enable Immediate mode
- Pick up the handset. If the service is active you'll hear dial-tone or the service prompt.

Note: if immediate mode is not set for the automatic test configuration, the Com.X will still provide dial-tone on the extension as usual, which might be incorrectly perceived as an external trunk service, so be sure to enable Immediate mode on the extension port.

3.9 Call recording

Please note that the call recording features provided by Far South Networks products are best-effort services and do not meet the requirements for recording to be used in legal cases, nor are they certified in any way by any regulatory bodies or authorities. Use of call recording may be subject to regulations specific to your country as well as international law.

Calls can be configured to be recorded on an individual extension, on all extensions and in the incoming, outgoing or both directions, for internal and external calls.



Depending on the size of the hard disk in the Com.X configuration, capacity for call recording may vary (E.g. a Com.X1 with a 250Gb hard drive can record ~275,000 minutes of voice).

3.9.1 Configuring individual extensions

To configure call recording setting on an individual user basis, log into the User Portal from the Com.X's main web page, and select Settings.

To configure call recording setting for all users, log into the User Portal as 'admin' from the Com.X's main web page, and select Settings. **All users will be affected!**

Our recommendation is to use WAV file format for recording on systems with hard disk drives and GSM for systems with solid state drives. When set to "On-Demand", calls will only be recorded when the user starts recording using the in-call call recording feature code. Please see section 4.5 for more detail on feature codes.

Note that, in order for the caller or the recipient of the call to initiate on-demand call recording, the general dial command options need to include "w" and / or "W". Please see 4.6.1 for more detail.

Note that disabling voicemail on an extension will also disable call recording.

3.9.2 Recording all calls

Log into the user portal as 'admin' with password 'ari_password'. and set their Call Monitor settings for both incoming and outgoing calls to "Always". Please see 3.9.1 for more detail.

Enable the Extension Recording Override setting in the system's General Settings. Select WAV or gsm as the recording format. Please see 4.6 for more detail. If this setting is Disabled, calls will not automatically be recorded, even if "Always" is selected in the user call monitor settings.

If a user's call monitor is configured to "Never" record calls, calls received by that user will not be recorded, even if Extension Recording Override is enabled.

3.9.3 On-demand recording

If so configured, users can record calls on-demand by pressing *1

On-demand recording will only work if the Advanced – General settings dial options include the correct recording permissions.

SIP phones need to translate DTMF properly (RFC2833)

Some analogue phones may not properly relay the DTMF to the PBX, in which case on-demand recording may not work. For such phones configuring recording to always be on is a work-around.

Pressing *1 activates recording if not active, and de-activates recording if active. The last *1 to activate on-demand recording in a call will over-write previous recordings for that call. I.e. if recording was active, then de-activated and re-activated, only the re-activation recording will be present for that call.



3.9.4 Call recording conditions

Due to privacy issues in countries that enforce these, the Com.X recording feature only records calls as follows:

Outbound calls are recorded if the caller has recording enabled in the outbound direction or uses on-demand-recording on the call.

Inbound calls are recorded if the caller has recording enabled in the inbound direction or uses on-demand-recording on the call.

Internal calls are recorded in both users' call monitors if the called user has inbound recording configured, regardless of the calling user's configuration.

Internal calls are not recorded in either of the users' call monitors if the called user does not have inbound recording configured, or has on-demand inbound call recording configured and has not used the on-demand feature code, regardless of the calling user's configuration.

If on-demand recording is activated on an internal call by either party (provided that party has on-demand recording configured), the call will be recorded, regardless of the other party's call recording configuration.

In summary then, on-demand recording over-rides all call recording configurations. Otherwise, the called user's call recording setting dictates the call recording behaviour for internal calls. Calls placed over or received via trunks are subject to the user's outbound and inbound call settings.

Note: It is the legal obligation of the called party to inform callers when recording is activate on a system. Similarly, when on-demand recording is used to record calls, the DTMF feature code activation might be audible to the remote party, but is not guaranteed. In such scenarios it is the user activating on-demand recording's responsibility to inform the other party that call recording has been activated.

3.9.5 Viewing call recordings

Individual users can view and play call recordings from their user's Call Monitor by selecting the Call Monitor option.

The system administrator can view and play all call recordings for all users by logging into the User portal with username 'admin' and password 'ari_password'.

The default password can be changed by editing the `/usr/share/freepbx/recordings/includes/main.conf.php` file and specifying the password in the `$ARI_ADMIN_PASSWORD` field.

3.9.6 Using NFS to record calls to a SAN / NAS

Note: NFS configuration is not backed up by the Com.X backup algorithm.

Should you require storage of recorded calls on a separate storage device, configure an NFS mount point and mount `/var/spool/asterisk/monitor` to the NFS storage system. Ensure that permissions to the mounted directory on the NFS storage system are:

```
drwxrwx--- asterisk asterisk
```



You may need to create an asterisk user for this purpose on the NFS storage system.

To configure the Com.X:

```
sudo apt-get install portmap nfs-common
```

```
sudo mount <nfs-server-ip>:<nfs-dir> /var/spool/asterisk/monitoring/
```

To enable the mount at system start-up, add to `/etc/fstab`:

```
<nfs-server-ip>:<nfs-dir> /var/spool/asterisk/monitoring/ nfs rw,auto 0 0
```

3.10 Powering the Com.X via an Uninterrupted Power Supply

Com.X products ship with a UPS service that supports standard USB-connected UPS products. When enabled the service will do the following:

- Monitor the UPS Report in `/var/log/syslog` when mains power goes down or is restored.
- Cleanly shut down the system when battery level is low.
- Power off the UPS at the end of shut down

The UPS will automatically switch on again (after some short delay) when power is restored. If present, it is recommended to connect applicable ADSL routers and related network switches to the same UPS as the Com.X.

If your Com.X system does not have the `comma-ups` service installed, it can be installed from the Far South Networks public repository as detailed in section 3.10.2

3.10.1 Supported UPS Devices

The following UPS devices are confirmed to work with the `comma-ups` service. Other devices may also function correctly:

- Proline B1000
- Proline B600

3.10.2 Installation

Step 1: From the command shell, verify that the `comma-ups` service is available on your Com.X by entering:

```
dpkg -l comma-ups
```

If the response is "No packages found matching `comma-ups`", install the service from the Far South Networks public repository:

```
sudo apt-get update
sudo apt-get install comma-ups
```

Step 2: Connect the UPS USB port to any USB port on the Com.X. If detected properly, you should see an entry similar to `/dev/bus/usb/001/002` when executing the command below:

```
sudo find /dev -group nut
```



A restart of the system may be required.

3.10.3 Service management

To enable the service permanently (now and on reboot):

```
sudo comma-ups enable
```

To disable the service permanently (this is the default after install):

```
sudo comma-ups disable
```

To check the service status:

```
comma-ups status
```

3.11 Securing the system

It is of utmost importance to ensure that the Com.X system is secure. If a Com.X unit is public facing, large costs can be incurred if the VoIP configuration security is weak, or downtime and degraded performance may result from the Operating System being compromised. The following steps are recommended to ensure a secure Com.X system:

3.11.1 General

- Ensure that access to the unit is restricted. Install the Com.X in a secure server room or locked cabinet. This prevents unauthorized access to the unit using the serial port or USB keyboard, as well as physical damage to the unit or its power supply.
- Change the default password for the comma user.
- Regularly check the system log files for unauthorized activity (`/var/log/auth*` and `/var/log/syslog*`).
- Use strong passwords.
- Regularly backup the system configuration and important data.

3.11.2 Network

- Preferably allow remote (external) access (VoIP and data) to the Com.X only by means of a Virtual Private Network (VPN)
- Configure the Com.X and telephony devices (e.g. iTA or VoIP phones) on a separate network partition on a managed switched, or a physically separate network.
- Assign IP addresses and services on the VoIP network based on a white list of approved MAC addresses.
- Place the Com.X behind a firewall.



- Perform port forwarding on a non-standard port to port 22 on the Com.X for SSH access. Tunnel HTTP access through the SSH connection.
- Limit administrator access (SSH and web GUI) to a white list of MAC addresses, IP addresses or networks.
- Activate fail2ban (please see section 7.7)

3.11.3 VoIP

- If the Com.X is accessing external VoIP trunks via a NAT router/firewall, no special port forwarding is required. If external VoIP agents (e.g. SIP phones) need to access the Com.X, forward only ports 5060 and 10000-20000 to the Com.X for VoIP control and voice communication.
- Configure the firewall to block all incoming VoIP traffic except that from a white-list of remote extensions.
- Ensure that all VoIP extensions and trunks have strong (generated) passwords, different from the extension number.
- Activate fail2ban (please see section 7.7)

3.11.4 PBX

- Change the Com.X GUI password from the default.
- Log the PBX console messages to `/var/log/asterisk/messages` and monitor this periodically. A large number of rejected VoIP registrations may indicate an attack on the system.
- Use pins and pin sets on international and high-cost outbound routes, requiring the caller to know the pin in order to place a call.
- Disable feature codes for administrative features, or change the feature codes to access those features to something other than the default. (E.g. ChanSpy, ZapBarge).
- Require all users to set their voicemail pin and user portal login passwords to something other than the default.
- Regularly backup the PBX configuration and valuable recordings.
- If direct database access is configured for CDR access, limit the access to a white list of MAC addresses, IP addresses or networks.

3.12 Call behaviour on No answer / Busy

This section describes the Com.X behaviour and call routing for incoming calls that go unanswered or encounter a busy extensions.

3.12.1 Unattended transfer

On unattended transfer, the person transferring the call is taken out of the loop, i.e. the call is treated as though the caller called the final party directly. Voicemail or follow-me for the recipient would determine where the call goes on busy and no answer.



If both voicemail and follow-me are not configured, the extension would ring indefinitely.

3.12.2 Attended transfer

The person wanting to connect the caller with a third party may place the caller on hold by picking up the second line, pressing a hold button or initiating an attended transfer.

The caller hears music, and the intermediary has one extension channel available to call the intended destination with and confirm availability. The intermediary can interact with the destination and connect the caller with the destination by using a transfer button or by putting the phone / channel down.

3.12.3 Call parking

The person wanting to connect the caller with a third party may place the caller in a parking bay.

The caller hears music, and the intermediary has both extension channels available for use. Once the necessary calls have been made, conferences set up, availability determined, etc., the intermediary can pick up the call in the parking bay and transfer or respond as required.

Should the time the caller spends in the parking lot exceed the specified timeout period, the action configured for orphaned callers is initiated. This could include playing an announcement to the caller and keeping the caller in the bay, or transferring the caller to reception.

3.12.4 Ring-back

Optional ring-back can be enabled, which requires an updated dial plan configuration and execution script to be uploaded to the Com.X. Ring-back behaviour is as below:

Follow me enabled?	Voicemail enabled?	Disposition	Action	Next Action
√	√	No answer	Follow me	Follow me configured destination (typically voicemail)
√		No answer	Follow me	Follow me configured destination (typically voicemail)
	√	No answer	Voicemail	
		No answer (internal call)	Rings Operator Extension	Operator voicemail if configured
		No answer (external call)	Rings Operator Extension	Operator voicemail if configured
√	√	Busy	Follow me	Follow me configured destination
√		Busy	Follow me	Follow me configured destination
	√	Busy	Voicemail	



Follow me enabled?	Voicemail enabled?	Disposition	Action	Next Action
		Busy (internal call) – with or without transfer	Busy (call terminates)	Ring-back initiated from called extension to calling extension when the called extension become available
		Busy (external call) – with or without transfer	Rings Operator Extension	Operator voicemail if configured

Note: Users need to be informed as to what to expect. With ring-back enabled their phones should 'ring by itself' once available. If the hand-set is picked up, they should hear ringing while the initial calling party is being contacted.

If the other party is now on the phone and also has neither follow-me nor voicemail enabled, when the handset of the phone on which ring-back was initiated is picked up, the user would hear a busy tone. A second round of ring-back should then be initiated. I.e. once the original caller (now on the phone) ends the call, his phone would ring, and picking it up, the initial called user would be attempted again.

3.12.5 Enabling ring-back

1. Download the following files from the Far South Networks archive onto the Com.X system:
<http://archive.farsouthnet.com/support/customizations/ring-back/extensions.conf>
<http://farsouthnet.com/archive/support/customizations/ring-back/ringback.sh>
2. Make a copy of /etc/asterisk/extensions.conf so that your system can easily be reverted should you require to do so.
3. In ringback.sh, configure the number of retries and the delay between retry. Eventually trying to ring back will give up based on these values.
4. Replace /etc/asterisk/extensions.conf with the downloaded extensions.conf and place ringback.sh in /usr/share/commagui/bin/scripts/ with execute permissions.
5. In the General settings section under the Advanced menu, configure the Operator extension to the required extension.
6. Restart the Com.X in order for the changes to take effect.



4 Advanced configuration

Access to a variety of advanced features is provided via the Advanced menu. Selecting a function from this menu opens a new window, which can be maximized with detailed configuration options for the feature. Once configured, select Submit at the bottom of the configuration page. Also remember to review and apply the updated configuration to the underlying system when ready.



4.1 Announcements

Announcements added as destinations allow messages to be played on inbound calls before transferring the calls to appropriate destinations. Announcements can play recordings created or uploaded as detailed in 4.16.

Field	Content
Description	The description of the announcement that will appear in selection lists (e.g. when selecting announcements for IVR)
Recording	The audio recording to play for this announcement
Repeat	The key a user may press to repeat the announcement. If this is set, a delay will be inserted after the announcement has been played to allow the user the option of repeating. Allowed keys are: 0-9, *, #
Allow skip	If this is enabled, should the user press a key during an announcement, the announcement will be skipped.
Return to IVR	When enabled, after the announcement has been played, the call is transferred to the last IVR. <div style="border: 1px solid black; padding: 5px; background-color: #f0f0f0;">Note that even if the call was transferred to other destinations since the last IVR, this will still transfer the call to the last IVR.</div>
Don't answer channel	If enabled, the channel will not be answered and the call transferred to the destination. If the channel supports early media, the announcement will be played as early media when the next destination answers the channel.
Destination after playback	Destination to transfer this call to. Please see 3.7.4 and 4 for destination options and configuration.



4.2 Callback

This feature allows users to dial into the system and hang up while ringing, after which the system will call the user back.

Field	Content
Callback description	A custom description for this callback.
Callback number	The number to call back. If this is left blank, the caller ID will be used to callback. <div style="border: 1px solid black; padding: 5px;"><p>Note: the device to be called back must support incoming calls directly(e.g. some hotel and pay phones do not).</p><p>Note: If a callback number is not specified here, the calling device must presents a proper CID.</p></div>
Delay before callback	The amount of seconds to delay before calling back.
Destination	Once connected via callback, the destination to transfer the called-back party to.



4.3 Conferences

This feature allows multiple users to join a conference with two-way audio on all devices. Conference features include join and leave announcements, passwords, conference administration, in-conference menus, etc.

Field	Content
Conference number	The number to dial to access the conference
Conference name	A custom name for this conference
User PIN	A pin required to be entered by users in order to join the conference. This field is optional. The user must press # after entering the pin.
Admin PIN	A pin that identifies the conference leader (admin). This field is optional, unless "Leader wait" has been enabled. The user must press # after entering the pin.
Join message	If set, this audio recording will be played before a user joins the conference.
Leader wait	If enabled, the conference will not start until the leader has joined by entering the Admin pin.
Quiet mode	When enabled, audio recordings associated with joining and leaving the conference will not be played.
User count	If enabled, the number of users already in the conference will be announced to a new user joining the conference.
User join/leave	If enabled, an announcement will be made when-ever a user joins or leaves the conference.
Music on hold	If enabled, music will be played when there is only one user in the conference.
Allow menu	When enabled, users and the admin user can access their menus by pressing *
Record conference	If enabled, conferences using this conference number will be recorded.



4.4 Day / Night control

A number of day / night modes can be programmed on the Com.X. Each has a feature index and can route calls according to its own set of rules. These allow configuration of day / night to support multiple receptionists, tea or lunch breaks, after hours, etc.

Field	Content
Feature code index	The index of this day/night mode. The index is the day-night feature code (please see section 4.5 for more detail on feature codes), with the last digit of the feature code (e.g. 280) replaced by this day/night's index (e.g.283, 284, 288, etc.) Up to 10 day/night entries are available, 0-indexed.
Description	Custom description for this day/night control. This is the value that will be listed in selection lists that allow day/night selection.
Current mode	This sets the initial and current mode for this day/night control. I.e. if Day is currently active, and Night is selected here, night will become active, and on system restart night will be the initial value for this day/night control.
Optional password	If not blank, a user will have to enter this password in order to toggle this day/night control.
Day destination	<p>The destination to use when this control is in Day mode. Another day/night destination can be selected here, chaining together these conditions to form a logical AND tree. I.e. a number of decision junctions can be joined and manipulated by setting the modes for the day/night controls that have been chained together. An example:</p> <p>280 has Day -> 281 280 has Night -> Support ring group extension 281 has Day -> 282 281 has Night -> Alternative receptionist 282 has Day -> Reception 282 has Night -> Tea / lunch room</p> <p>A receptionist can enable night mode in the evenings and day mode in the mornings by using *280. When tea-time arrives, the receptionist enables night mode for 281 using *281. After tea-time she re-enables day mode for 281 using *281.</p> <p>Please see 3.7.4 for detail on destinations.</p>
Night destination	The destination to use when this control is in Night mode.



Field	Content
	Please see 3.7.4 for detail on destinations.

4.5 Feature Codes

An extensive list of feature codes are available to users and administrators to facilitate administration of the system and user call preferences using a phone. For each function, the administrator may change the feature code associated with the function, enable or disable the function, or revert to the default feature code associated with the function.

Field	Content
Name	The name of the function category or the individual function.
Code	The feature code currently associated with the function
Use default	If enabled, the existing feature code will be replaced with the default feature code for this function, if different from the existing one.
Enabled	Allows the disabling and enabling of individual functions. If disabled, the feature will not be available on the system. The code associated with the feature will remain configured even when disabled.



4.6 General Settings

General settings apply system-wide on the Com.X platform. Users can typically override general settings with their own preferences, but some general settings override all user preferences.

4.6.1 Dial command options

t: Allow the called user to transfer the call with the “in-call blind transfer” feature code (default ##).

T: Allow the calling user to transfer the call with the “in-call blind transfer” feature code (default ##).

r: Generate a ringing tone for the calling party

w: Allow the called user to start recording after pressing the “in-call recording” feature code (default *1).

W: Allow the calling user to start recording after pressing “in-call recording” feature code (default *1).

L: Limit trunk calls to a specified duration. Parameters are separated using ':' and only the first parameter is mandatory. The first parameter indicates the maximum duration in milliseconds of the call. The second parameter indicates when a warning message should first be played when the specified number of milliseconds remain on the call. The third parameter indicates the frequency in milliseconds with which the warning message should be repeated. To limit calls to 1 hour and warn the user every minute after only 5 minutes are left on the call, use 'L(3600000:300000:60000)'

Field	Content
Internal Dial command options	Options specified here applies to internal calls. Please see 4.6.1 for more details.
Outbound dial command options	Options specified here applies to outbound calls. Please see 4.6.1 for more details.
Extension recording override	<p>If disabled, all user recording settings other than the on-demand settings will be ignored, and no recording other than on-demand will take place.</p> <p>If enabled, the recording format setting specified here will override user-specific format settings.</p> <p>Recording settings for other modules, such as queues and conferences remain unaffected. Disabling this override has a positive impact on system performance, as the checks that are performed for each call re: recording becomes unnecessary.</p>
Call recording format	Recordings will be saved on the Com.X hard drive in the format selected here. Options are: wav, ulaw,



Field	Content
	<p>alaw, GSM, SLN, G.729</p> <p>We recommend using the wav format, since it results in higher quality recordings and does not use additional codec translation resources, which may affect performance on systems with high-usage.</p>
Recording location	<p>If this field is blank, the default location (<code>/var/spool/asterisk/monitor/</code>) will be used to store recordings.</p> <p>If this field is populated, ensure that the trailing <code>/</code> is included and that permissions on the directory are correct. This is required in order for the system to store recordings and the user's call monitor function to download the recordings:</p> <pre>chown asterisk:asterisk -R <dir>/recordings/ chmod g+rx <dir>/recordings</pre> <p>Also configure the <code>ASTERISK_CALLMONITOR_PATH</code> entry in <code>/usr/share/freepbx/recordings/includes/main.conf.php</code> to point to the new location.</p> <p>Users can access their recordings via the User login and Call Monitor feature.</p> <p>If there is a requirement for an administrator to get access to all recordings, set this location to <code>/var/www/recordings</code> and secure the directory (please see the Apache web server administration manuals). Then access the list using <code>http://<server-ip>/recordings/</code></p>
Run after record	<p>If not empty, this field specifies a custom script to be run after each recording has completed. The script should be executable and the location to the script the absolute path.</p>
Ringtime default	<p>This specifies the default time an extension should ring before activating voicemail.</p>
Direct dial voicemail prefix	<p>Dialling the prefix specified here followed by an extension number will take the user directly to the mailbox for that extension.</p> <div style="border: 1px solid black; padding: 5px; margin-top: 10px;"> <p>Note: please ensure the prefix specified here does not clash with any feature codes. Please see 4.5 for more detail on feature codes.</p> </div>



Field	Content
Direct dial to voicemail message type	<p>The message (or no message) the user will hear before the voicemail prompt when dialling directly into an extension's voicemail. Can be one of:</p> <p>Default - "Please leave your message after the tone. When done, hang up or press the pound key."</p> <p>No message</p> <p>Unavailable - "The person at extension <ext> is unavailable." followed by the default message.</p> <p>Busy - "The person at extension <ext> is on the phone." followed by the default message.</p>
Optional voicemail recording gain	<p>If present, applies additional gain on the recording. The number entered is interpreted as whole-numbered dBs.</p>
Do Not Play "please leave message after tone" to caller	<p>When checked, this option removes the default message to be played after the selected (if any) voicemail announcement has been made, but instead plays the tone directly after the voicemail announcement.</p>
Default Context & Pri	<p>The internal dial-plan context to use if the user dials an extension or a short number that does not match outbound routes. This should remain the default unless requested otherwise for test purposes by Far South Networks.</p>
Timeout/#-press default	<p>The internal destination to route a user after timeout. This should remain the default unless requested otherwise for test purposes by Far South Networks.</p>
Loop Exceed default	<p>The destination that a user is transferred to when exceeding the maximum number of invalid selection options during a voicemail call. This should remain the default unless requested otherwise for test purposes by Far South Networks.</p>
Timeout VM Msg	<p>Indicates whether the standard voicemail instructions should be played on timeout, or whether the tone should be played directly.</p>
Max Loop VM Msg	<p>Indicates whether the standard voicemail instructions should be played, or whether the tone should be played directly.</p>
Direct VM	



Field	Content
Msg Timeout	Time to wait after message has played to timeout and/or repeat the message if no entry pressed.
Msg Play	Number of times to play the recorded message if the caller does not press any options and it times out.
Error Re-tries	Number of times to play invalid options and repeat the message upon receiving an undefined option.
Find users in the Company Directory by	Specifies whether the first three letters entered by a user using the directory will look up by first name, last name or both.
Announce Extension	When transferring a call as a result of a directory lookup, announces the transfer extension if this option is enabled. "Please hold while I transfer you to extension xxx"
Operator Extension	When a user selects a 0 while engaging the directory, the user is transferred to this number. The number may transfer to an extension, ring group, external number, etc.
Country Indications	Configures the Com.X PBX country
24-hour format	Configures whether 12-hour AM/PM or 24-hour format should be employed by the Com.X PBX
Allow Anonymous Inbound SIP Calls?	Enabling this setting disables authentication checks on incoming SIP requests. I.e. anyone can place a SIP call to this PBX.
Check for updates	Allow the system to automatically check the internet for FreePBX and Asterisk version updates. Updates of the FreePBX and Asterisk modules are managed by Far South Networks for the Comma and Com.X product ranges. This field should be left blank.
Update email	An email address to send notifications of available updates to. Updates of the FreePBX and Asterisk modules are managed by Far South Networks for the Comma and Com.X product ranges. This field should be left blank.



4.7 IVR

The Interactive Voice Response feature allows the creation of and chaining together of interactive voice announcements and corresponding menu selections to facilitate user call transfer in the Com.X system.

Field	Content
Name	Name used to identify this IVR. This entry will be displayed in selection lists that support IVR.
Announcement	The audio message to be played to the user when this IVR is accessed. The message typically would present the user with a list of options and corresponding digit(s) to activate the option.
Timeout	The number of seconds without user response before this IVR times out. When timed out, if a destination with option 't' has been defined, this destination would be transferred to on timeout.
Enable directory	If enabled, this allows callers currently being serviced by this IVR to dial the Directory feature code (default #) to access the directory. Please see 4.5 for more information on feature codes.
VM Return to IVR	When enabled, should one of the IVR options result in the user's call being transferred to a voicemail box, after leaving a message or exiting the voicemail menu, the user is returned to this IVR.
Directory Context	If directory access is enabled, access the directory specified here if the user uses the Directory feature code.



Field	Content
Enable Direct Dial	<p>If enabled, users can dial extensions directly from within the IVR.</p> <p>Note that having direct dialing enabled introduces a delay before the IVR responds to menu options while the system gives the user a chance to dial more digits re: dialing an extension.</p> <p>Note that if enabled, if some of the configured extension start with the same number as one of the configured IVR options, the option might be inaccessible as the system will be collecting digits in order to dial an extension, as opposed to servicing IVR options. Rather disable direct dial in your main IVR and include an option for direct dialing (e.g. 'If you know the number of the extension you wish to dial, please press 8) which leads to a second IVR with a return option, and direct dial enabled, from which the caller can then dial the desired extension.</p>
Loop before t-dest	<p>If enabled, the IVR loops as described for the 'Repeat loops field' when user selection times out.</p>
Timeout message	<p>If specified, this audio message is played when user selection times out.</p>
Loop before i-dest	<p>If enabled, the IVR loops as described for the 'Repeat loops field' when user a user selects an invalid option.</p>
Invalid message	<p>If specified, this audio message is played when the user selection is invalid.</p>
Repeat loops	<p>The number of times the IVR should be played if the user does not provide an option before timeout, or if the option is invalid.</p> <p>The IVR will only loop on invalid input if the 'Loop before i-dest' option is enabled. The IVR will only loop on timeout if the 'Loop before t-dest' option is enabled.</p> <p>If the number of repeats has been reached and another timeout or invalid selection occurs, the relevant timeout 't' or invalid 'i' destination is used. If these destinations are not configured, the call terminates.</p>



Field	Content
Destinations	<p>For each valid selection in this IVR, enter the digit(s) to activate the selection and the call destination to transfer the caller to.</p> <p>A special 't' destination is used to configure a destination on user selection timeout.</p> <p>A special 'i' destination is used to configure a destination on invalid user selection.</p> <p>For each destination, if the Return to IVR option is enabled, the destination will be ignored and the user returned to the parent IVR of this IVR.</p> <div data-bbox="515 759 1348 916" style="border: 1px solid black; background-color: #f0f0f0; padding: 5px;"><p>Note: it is possible that the last IVR was not the transfer directly before this IVR, but some steps removed. The Return to IVR option will cause the call to return to the last previous IVR, regardless of how many non-IVR transfers may have taken place in the interim.</p></div>



4.8 Music on hold

The Music on hold feature allows for the upload of audio files (e.g. music, announcements, advertisements, informationals, etc.) in either WAV or MP3 format for playback to users when calls are on hold.

Note: when mp3 files are uploaded, the files are internally converted to wav format. This process can take as long as the total length of the mp3, and utilized a high amount of CPU resources. We recommend that mp3 upload is performed during scheduled maintenance.

Music may be categorized and music from a specific category played for specific contexts, e.b. when a user is waiting in a support queue.

4.8.1 Configuring streaming music

To configure music on hold to stream from a media server, select Music on hold from the Advanced menu, and then add a streaming category.

Please see section 7.4 for details on configuring a streaming music server.

Enter a name for the stream, and configure the music player command as follows, using the URL to the streaming server:

```
/usr/bin/mpg123 -q -s --mono -r 8000 -f 16384 -b 100 <url>
```

The `-s` switch redirects the music stream for PBX use

The `--mono` switch transforms the stream to 1 channel audio as required

The `-f` switch adjusts the audio volume. You can also try `-f 8192` (softer) or `-f 32768` (louder)

The `-b` switch configures an audio buffer of the specified amount of Kbytes. If your connection to the streaming server suffers from delays, consider adjusting this value. Setting this to a high value might result in a delay before streaming commences.

`<url>` represents the URL to the streaming server, including any protocol and port indications.

Use the optional `-@` switch if the streaming server is publishing a playlist (e.g. `.m3u`) instead of directly streaming the audio (e.g. as in the case with the GNU MP3 daemon).

Submit and apply the configuration. This streaming category can now be used where music on hold selection is possible (e.g. queues).

For inbound calls you can set the music on hold class from the inbound route's advanced context menu.

Warning: If you incorrectly configure the application entry above, applying might result in the apply freezing up. To fix this, restart the Com.X, correct the application entry, or remove the streaming music on hold category and apply.

4.8.2 Playback configuration



Field	Content
Upload	The audio file to upload. Supported formats are: WAV, MP3
Volume	This value is linear, but human sound perception is logarithmic. As a result we recommend that you test the volume configured here for comfort and quality levels. This feature will convert MP3 files to WAV files. If you do not have mpg123 installed, you can set the parameter: AMPMPG123=false in your /etc/amportal.conf file.
Enable random play	Enabling random play will select audio files to play in a random order. If this is disabled, files are played in the order listed here.

4.8.3 Adding a music category

Music categories categorize the audio files available on the Com.X PBX server storage.

Field	Content
Category name	A unique name for this audio category

4.8.4 Adding a streaming category

Streaming categories allow the inclusion of internet streaming audio streams for features that support Music on hold.

Field	Content
Category name	A unique name for this streaming category.
Application	This is the "application=" line used to provide the streaming details to Asterisk. Please see information on the musiconhold.conf configuration for different audio and internet streaming source options.
Optional format	An optional "format=" line that is used to provide the audio format. This should be a format understood by Asterisk such as ulaw, and is specific to the streaming application you are using. Please see information on the musiconhold.conf configuration for different audio and internet streaming



Field	Content
	source options.

4.9 Outbound routes

This feature allows advanced access to outbound routes and should only be used when the Outbound Routes panel does not provide sufficient levels of detail or flexibility given the administration need.

Field	Content
Route name	A unique name for this route.
Route password	An optional password. If present, users will be prompted to enter this password before call routing using this route is allowed. <div style="border: 1px solid black; padding: 5px; margin-top: 10px;">Note: this is a single password for the route. Pin sets should be assigned using the Pin set field.</div>
Pin set	If more than one pin should allow access to this route, leave the Route password field empty and specify a pin set here.
Emergency Dialing	When enabled, the CID field of the caller will be overridden by the calling device's emergency CID value (if configured).
Intra company route	If enabled, the internal caller ID information will be retained, and not replaced by the extension's external CID nor the route's external CID.
Music on hold	If specified, the audio file will be played before routing the call. This allows country-specific or destination-specific messages or instructions to be played before placing the outbound call.
Dial patterns	This route will only route calls if it is the highest priority outbound route that matches the destination number to one of the dial patterns specified here. For more information on dial patterns, please see 3.6.4.2.
Trunk sequence	This prioritized list specifies the order in



	which trunks should be utilized to route calls. the first trunk in this list is attempted, failing which the second trunk is attempted, etc.
--	--

4.10 Paging

Paging allows groups of extensions to be created that will receive calls (provided the phones for the specified extensions allow paging).

Field	Content
Paging extension	The extension to dial to activate the page
Group description	A custom description for this group
Device list	The list of devices and extensions to include in this paging group
Force if busy	If enabled, the system will not check whether the extension is busy before paging it. Active calls may be interrupted. This is useful for emergency pages.
Duplex	When enabled, all supporting devices will allow two-way communication to all other extensions and devices in this list. I.e. any audio generated by any device will be audible to all other devices and extensions.
Default page group	If enabled, this function removes the default flag from any previous default paging group. If set, an extension, when created, may be specified to join the default paging group, resulting in it automatically being added to this group and automatically being deleted from this group if such an extension is deleted.

4.11 Intercom

The intercom feature allows for a call to be placed to phones supporting the feature, and for such phones to automatically answer the call and play the audio on speaker.

To enable intercom (disabled by default), navigate to the GUI, then Start – Configuration – Advanced and Feature Codes.

Enable the Intercom Prefix feature code and change the feature code to a code that will not conflict with existing codes (e.g. *93). Apply.



The list of phones known to support Intercom and Paging include: Aastra, Grandstream, Linksys/Sipura, Mitel, Polycom, SNOM, and possibly other SIP phones. ATAs are not supported. If your phone is set to auto-answer, it should also support intercom.

The phone must be configured to support the feature. On a Snom for example, configuration would involve turning on Auto Connect Indication, setting the Auto Connect Type to Handsfree and enabling the Intercom feature.

Please contact your phone manufacturer or consult the phone manuals for specifics on configuring your phone and model for intercom support.

4.12 Parking Lot

Callers can be placed in a parking with a time out associated, after which the caller is transferred to the specified destination. This allows receptionist, for example, to place a caller on hold while waiting for their desired extension to become available. The caller in the parking lot would then, after the parked timeout, automatically ring reception again, by which time the desired extension might have freed up.

To transfer a call to the parking lot, use a transfer feature and dial extension 70. The allocated slot will be announced. To retrieve the call, dial the slot number.

Field	Content
Enable	Call parking will only be available if this feature is enabled.
Parking lot extension	The extension to transfer calls to in order to park then (the default extension number id 70)
Number of slots	The number of parking bays available. Parking lots are numbered as follows: Parking lot extension + bay number, e.g. 71, 72, etc.
Parking timeout	The number of seconds after which the call is removed from the parking bay. The call is then transferred back to the extension from which the call was transferred into the parking bay.
Parking lot context	The dialplan context for parked calls. Changing this setting might result in parked calls no longer functioning.
Parking Alert-Info	Additional information to be added to the call when the call is transferred to the destination for orphaned calls. Phones supporting SIP Alert information could then inform the user receiving and orphaned call of the call's orphaned status by displaying a message or playing a different ring-tone.
Caller ID pre-pend	Additional information to be pre-pended to the CID when the call is transferred to the destination for orphaned calls. This allows a user receiveing an orphaned call to be informed of the call's orphaned status.
Announcement	An announcement to be played to the caller in the parking lot



Field	Content
	when the call is flagged as orphaned, before transferring the call to the orphaned call destination.
Destination	The destination for calls that become orphaned. Please see section 3.7.4 for more information on call destinations.

4.13 Pin Sets

Pin Sets are a convenient way to group sets of pins to facilitate multiple users with individual pins gaining access to features, without using a common pin for the features.

Field	Content
Description	Custom description for this Pin set. This will be listed in selection lists that allow Pin sets.
Record in CDR	If enabled, when this Pin set is used in call routing, the Call Detail Record includes the Pin set details.
Pin list	The list of pins included in this Pin set.



4.14 Queues

Queues are a powerful feature of the Com.X system, allowing callers to be placed on hold and transferred according to a number of strategies and based on priority.

Callers waiting in the queue are provided with their position in the queue, projected waiting time and the option to use self-service menus to exit the queue and transfer to other destinations.

Agents servicing queues are allowed to log in by dialing the queue number followed by *, and log out of queues by dialing the queue number followed by **, are provided with information re: callers waiting in the queue, and queues can be configured in flexible manners to transfer calls to agents.

4.15 Ring Groups

Ring groups allow groups of extensions to be notified of incoming calls using a number of powerful ring strategies. A rich set of ring group parameters allow flexible configuration of timeout, alerts, music on hold, pick-up confirmation and other features.

4.16 System Recordings

Audio can be recorded and used in functions such as announcements, IVR, etc. Recordings can be uploaded and made from an extension. Follow the instructions on the system recordings page.

Note: Remember to press the # key to indicate the end of the recording before giving it a name.

Field	Content
Extension number	If an extension number is entered here, this number will be used to record a recording should the Save Recording feature code (please see section 4.5 for more detail on feature codes) be dialled from that extension.
Upload	A recording file in PCM Encoded format, 16 Bits, at 8000Hz, single channel. Recordings are uploaded to <code>/var/lib/asterisk/sounds/custom/</code>
Name of this recording	A custom name for the recording. This name will appear in selection lists for functions that make use of recordings.



4.17 Time Conditions

Time conditions check the current Com.X system time and compares it with the specified Time Group to determine whether there is a match. The call is then routed either to a day-night entry if this over-ride has been enabled, or to the match destination if the current time falls within the time block specified in the Time Group, or to the no match destination otherwise.

Field	Content
Name	Name used to identify this time condition in selection lists that allow time conditions.
Time group	The Time Group that specifies the time range that results in a match for this time condition. If the current system time does not match this time group, the call will be routed to the no match destination. Please see 4.18 for detail on Time Groups.
Associate with	If a day-night feature code (e.g. 283) is specified here, and the day/night entry is enabled, this time condition will route directly to the day/night entry, ignoring the time condition settings specified here. Disabling the day/night entry will cause calls to be routed according to the time condition settings.
Destination if matches	Destination to route the call to if the Time Group conditions are matched.
Destination if no match	Destination to route the call to if the Time Group conditions are not matched.



4.18 Time Groups

Time groups allows the specification of blocks of time for use in call routing rules, such as time conditions. Please see 4.17 for more detail on Time Conditions.

Field	Content
Name	A name to identify this time group with. This will appear in selection lists that allow time groups.
Time to start	The start time for the block of time this time group specifies.
Time to finish	The end time for the block of time this time group specifies.
Week day start	The start day for the block of time this time group specifies.
Week day finish	The end day for the block of time this time group specifies.
Month day start	The start day in the month for the block of time this time group specifies.
Month day finish	The end day in the month for the block of time this time group specifies.
Month start	The start month for the block of time this time group specifies.
Month finish	The end month for the block of time this time group specifies.



4.19 Voicemail Blasting

Voicemail blasting allows messages to be added to a number of voicemail boxes at once.

Field	Content
Voicemail blast number	The extension to dial to leave a message to this voicemail blasting group
Group description	Custom description for this voicemail blasting group
Audio label	Select the audio recording or announcement to play before the voicemail prompt for this voicemail blasting group.
Password	Optional
Voice mailbox list	Select (using Ctrl-click) all the extensions that should receive the voicemail message in their mailboxes.
Default voicemail blast group	If enabled, this function removes the default flag from any previous default voicemail blasting group. If set, an extension, when created, may be specified to join the default voicemail blast group, resulting in it automatically being added to this group and automatically being deleted from this group if such an extension is deleted.



4.20 Phonebook

The phonebook feature allows names and speed dialing codes to be matched with extension and outbound numbers. It allows a user to dial numbers by phoning the phonebook feature code (please see section 4.5 for more detail on feature codes) and entering the first three letters of a desired phonebook entry's name, after which the call is transferred to the desired extension or an outbound call is placed with the desired number.

Field	Content
Name	The name to look the first three letters up from.
Number	The extension or outbound number to be dialed for this entry.
Speed dial code	A speed dial code to associate with this phonebook entry.
Set speed dial	The speed dial entry will be added to the database if this field is enabled.
File	An optional CSV format file for uploading multiple phone book entries. Entries should be in the following format: "Name";number;speeddial <div style="border: 1px solid black; padding: 5px; margin-top: 10px;">Note: if the delimiter is something other than ';' the import will fail. Please see your operating system / application help for configuring ';' as the CSV export delimiter.</div>

4.21 Listening in on calls / Spy feature

The Com.X PBX allows a spy feature to be enabled that allows the listener to listen in on calls in progress. Input from the listener will not be audible to the callers on the call being listened in to.

To listen in on all calls, enter the ChanSpy feature code. The first active channel will be listened in on. To cycle through the channels, use the * key. To increase or decrease the volume of the intercepted audio, use the # key.



5 Monitoring the system

A rich suite of monitoring functions are available from the Com.X GUI. Monitoring domains include Call Detail Records, PBX system monitoring and events, Configuration details, Host system monitoring, Telephony status and events and Gui web server status.

5.1 Detailed logging

The Com.X supports a variety of logging levels, as detailed below:

Log level	Description
debug	Source code level debug reports. Enabling this level results in a large volume of logging data.
notice	Informative messages that report on system, channel, span and call state.
warning	Informational messages requiring attention and evaluation. Warnings may indicate conditions that may limit or in future potentially impede functionality.
error	Error reports indicating items that should be addressed should system functionality be affected.
verbose	Additional informational messages (similar to notices), allowing more effective call tracing.

Logging takes place on the CLI as well as to `/var/log/asterisk/messages`

To include additional logging on the CLI or the messages file, edit `/etc/asterisk/logger.conf` and add the required keyword to the 'console' or 'messages' entries.

In order to reload the logger configuration, execute:

```
sudo asterisk -rx "logger reload"
```

From the Com.X GUI, two levels of detail can be selected, namely "brief" or "verbose". These may be accessed using Monitor -> Asterisk -> Set log level <Brief/Verbose>

5.2 Call Detail Records (CDR)

5.2.1 Reviewing Call Detail Records

The Com.X records call detail records for all calls. To access these records, select "Call Log" from the Report menu. Specify the period for which records should be retrieved and select "Search".

For each record, the date, channel, call source, caller ID, destination, the disposition and duration of the call is logged.



Note that, if the outbound CID on a trunk has been configured with an override (or as 'hidden'), the override CID will be stored in the caller ID field, not the original CID, as well as in the Source field (empty if the override is 'hidden').

If the extension number or a specific account code should rather be stored in the CDR when the trunk CID has been over-ridden, configure the "Account code" field on each extension to an appropriate value. The account code is then stored in the CDR, while the upstream telco is presented with the CID as configured on the trunk.

A result summary is displayed at the bottom of the list, where navigation of the list is also facilitated. CDR records are stored in a database on the Com.X and can be backed up. Please see 2.3.1.3 for more detail on backing up the CDR database. Call reports can also be exported to Portable Document Format or in Comma Separated Values format for further processing.

5.2.2 Enabling logging to CSV

To enable logging of all CDRs to `/var/log/asterisk/cdr-csv/Master.csv`, uncomment the 'enable=yes' line in `/etc/asterisk/cdr.conf`. Please see section 6.1.2.3 for details re: editing system files.

5.2.3 Accessing the CDR database

To access the records in the MySQL CDR database remotely, the Com.X needs to be configured to allow an external user access.

1. Configure the Com.X to allow remote connections to the MySQL server by editing `/etc/mysql/my.cnf` and changing the `bind-address` field to `0.0.0.0`

Please note that this poses a security risk and that remote access should be restricted by a proper firewall.

2. Restart the MySQL server:

```
sudo /etc/init.d/mysql restart
```
3. Restart the comma-gui service:

```
sudo /etc/init.d/commagui restart
```
4. Log into the mysql server as root from the Com.X console:

```
mysql -u root
```
5. Configure a cdruser with read access to the CDR database:

```
GRANT SELECT ON asteriskcdrdb.* TO cdruser@%' IDENTIFIED BY 'password';  
FLUSH PRIVILEGES;
```
6. Configure your SQL client to point to the server, with username cdruser and the password as configured, on the default port 3306, and with database schema asteriskcdrdb.

5.3 Retrieving monitoring information

To access monitoring functions, select Monitoring from the main menu bar, and select the category and function of interest. Results are displayed in tabular format, with the ability to page to more results. In some cases the information is colour coded to facilitate easier analysis. The reload button re-issues the monitoring query, facilitating ad-hoc snapshots of monitoring data.



Note: Due to a known issue with the tabular representation of results, if you are only seeing one page, manually enter "2" in the page number at the bottom of the table and press enter. All pages of information should then become visible.

5.4 Monitoring functions

Category	Function	Description
Asterisk		
	Asterisk Events	Asterisk event log (/var/log/asterisk/event_log)
	Asterisk Messages	Asterisk messages log (/var/log/messages)
	Asterisk Queues	Asterisk queue log (/var/log/queue_log)
	Channels	Asterisk CLI (show channels). Lists channels in use and active calls.
	Database	Displays the PBX configuration database
	Set log level brief	Enable logging of notices, warnings and errors only
	Set log level verbose	Enable logging of DTMF, verbose messages, notices, warnings and errors
Configuration		
	DHCP Configuration	
	Dahdi system	Dahdi span configuration (/etc/dahdi/system.conf)
	Dahdi channels	Dahdi channel configuration (/etc/asterisk/chan_dahdi.conf)
	Extensions	Extension configuration file (/etc/asterisk/extensions.conf)
	IAX peers	IAX peer configuration file (/etc/asterisk/iax_additional.conf)
	IAX registrations	IAX registrations configuration file (/etc/asterisk/iax_registrations.conf)



Category	Function	Description
	Network interfaces	System network interface configuration file (/etc/network/interfaces)
	SIP peers	SIP peer configuration file /etc/asterisk/sip_additional.conf
	SIP registrations	SIP registrations configuration file (/etc/asterisk/sip_registrations.conf)
	mISDN ports	mISDN configuration file (/etc/asterisk/misdn.conf)
Reload		
	Reload Asterisk	Reloads the Asterisk PBX. Results in reload of SIP, IAX, dahdi and mISDN. Complete service loss during reload.
	Reload IAX	Reloads the IAX module.
	Reload SIP	Reloads the SIP module. Causes SIP trunks and extensions to re-register.
	Reload mISDN	Reloads the mISDN module. Service to BRI channels will be lost while reloading.
	Reload Dahdi	Reloads the dahdi module. Service to analogue and PRI channels will be lost while reloading.
System		
	Disk usage	Displays system mount points, their capacity and usage statistics.
	Network status	System network interface status (ifconfig) Displays interfaces, whether they are Up and their Rx, Tx and error statistics. Also shows IP configuration for each interface.
	Performance	Displays system performance snapshot, including CPU



Category	Function	Description
		performance, active processes, memory usage and page file usage.
	System log	Displays the system-wide log. This includes information, warning and error messages from system-wide applications and services (/var/log/syslog)
Telephony		
	Dahdi channels	Asterisk CLI (dahdi show channels) Displays the list of channels visible to the telephony driver.
	Dahdi spans status	Asterisk CLI (dahdi show status) Displays the current dahdi span status. If spans are in error here, there may be a configuration problem, or a Comma device might not be reachable.
	Dahdi statistics	Displays statistic related to the dahdi dynamic spans, including slips and skips. If these numbers do not remain constant, it might indicate network performance issues that could result in a decrease in call quality.
	Device discovery	Displays the list of Comma devices visible to the Comma Manager on all networks, including their ethernet and IP configuration status.
	IAX peers	Asterisk CLI (iax2 show peers). Displays IAX peers that have registered with the PBX and their status.
	IAX registry	Asterisk CLI (iax2 show registry). Displays the PBX IAX registry. IAX trunks should be listed here when registered with IAX providers.



Category	Function	Description
	Manager log	Displays the Comma Manager log. This is a useful resource for finding problems with Comma devices booting and channel events.
	SIP peers	Asterisk CLI (sip show peers). Displays SIP peers that have registered with the PBX and their status.
	SIP registry	Asterisk CLI (sip show registry). Displays the PBX SIP registry. SIP trunks should be listed here when registered with SIP providers.
	misdn channels	Asterisk CLI (misdh show channels) Displays available BRI channels
	misdn stack status	Asterisk CLI (misdn show stacks) Displays the ISDN BRI software stack status, including link enumeration, TE/NT configuration, point to point and multipoint configuration and status.
Web Server		
	Access	Log of web server access events (/var/log/apache2/access.log)
	Errors	Log of web server error events (/var/log/apache2/error.log)
GUI	Client Log	Debugging information, warnings and errors generated by the GWT-based GUI running in the browser.



6 Advanced administration

6.1 Console access

6.1.1 Authentication

Default username: comma

Default password: farsouth

6.1.2 Accessing the unit

On power-on the unit can be accessed either via serial cable or via a secure shell using ssh if the IP address is known from a DHCP server.

A serial connection can also be used to obtain the IP address allocated by a DHCP server.

6.1.2.1 Serial access

Connect a serial cable to the unit's serial port and configure a terminal emulator such as putty with the following settings: 115200,8,N,1

The cable should be a "null modem" serial cable, with pin 2 (RxD) and pin 3 (TxD) crossed and pin 5 (ground signal) connected to pin 5 (ground signal).

The DHCP-assigned IP address can be obtained from the serial console using the ifconfig command:

6.1.2.2 Secure shell access

Secure shell login is supported on the default SSH port 22:

The unit can be accessed using ssh on eth0, eth1 and eth2.

6.1.2.3 Editing system files

To edit a file, use the 'jed <filename>' file editor. If superuser privileges are required to edit the file, use 'sudo jed <filename>'.

6.1.2.4 Changing the shell login password

Log in as the comma user. To change the password, type: 'passwd'. Enter and confirm the new password.

6.2 Rebooting the system

Log into the system and issue the following command:

6.3 Configuring email services

6.3.1 Configuring voicemail to email

When a user has voicemail to email enabled, the Com.X needs to know which address to point the user to in the email, as well as the email address to send the voicemail message form.

Log into the Com.X and edit the `/etc/asterisk/vm_email.inc` file



Edit the email line to read: `http://<ip>/freepbx/recordings/index.php`

Note that 'freepbx' needs to be added and AMPWEBADDRESS replaced with the IP or FQDN of the Com.X

Edit `/etc/asterisk/vm_general.inc` and specify the correct email address voicemail emails should appear to come from in the 'serveremail' field.

The Com.X PBX software then needs to be restarted with

```
sudo /etc/init.d/asterisk restart
```

6.3.2 Configuring sendmail for mail relay

Note: Mail relay configuration is not backed up by the Com.X backup algorithm.

The Com.X product range makes use of sendmail for outbound SMTP mail delivery. The sendmail server will attempt to directly resolve destination mail server addresses and contact these servers directly for mail delivery.

If your network design requires the use of a mail relay server for mail delivery, sendmail can be configured to deliver outbound mail into the relay server's queue as follows:

Open an ssh shell to the system as described in section 6.1.2.2 and perform the following steps:

```
sudo su
```

You should now have a root prompt. All actions following are executed as the root user. Please use caution and use 'exit' to return to a normal user after completing the steps below.

```
/etc/init.d/sendmail stop
```

This will stop mail delivery while the configuration is underway. Edit the `/etc/mail/sendmail.mc` file as described in section 6.1.2.3 and add the mail relay entry near the bottom (directly after the includes).

```
define(`SMART_HOST', `mailrelay.somewhere.com')dnl
```

Note that the relay server's hostname must be entered, not its IP. If the hostname is not known, create an entry called 'mailrelay' in `/etc/hosts` with the IP and use 'mailrelay' as the hostname.

Ensure that the `FEATURE` definitions are listed before the `MAILER` definitions. The default sendmail configuration has these in the wrong order, resulting in an error (`*** ERROR: FEATURE() should be before MAILER()`) when compiling. For example:

```
include(`/etc/mail/m4/dialup.m4')dnl
include(`/etc/mail/m4/provider.m4')dnl
define(`SMART_HOST',`mailrelay')dnl
dnl # Masquerading options

FEATURE(`always_add_domain')dnl
MASQUERADE_AS(`-e false')dnl
FEATURE(`allmasquerade')dnl
FEATURE(`masquerade_envelope')dnl
dnl #
```



```
dnl # Default Mailer setup
MAILER_DEFINITIONS
MAILER(`local')dnl
MAILER(`smtp')dnl
```

The mail relay server may be configured to allow emails through only from a specific domain, in which case the sendmail server could need to be configured to masquerade as originating email from that domain. If this is the case, replace the default `MASQUERADE_AS` and `FEATURE` entries above with:

```
MASQUERADE_AS(required_domain)dnl
MASQUERADE_DOMAIN(current_domain)dnl
FEATURE(masquerade_entire_domain)dnl
FEATURE(masquerade_envelope)dnl
```

For example, if the Com.X hostname is `comx1` on domain `mydomain.com`, `current_domain` would be `comx1.mydomain.com` and `required_domain` would be `mydomain.com`

Re-generate the sendmail configuration and start the mail server.

```
m4 /etc/mail/sendmail.mc > /etc/mail/sendmail.cf
/etc/init.d/sendmail start
```

Exit to a normal user.

```
exit
```

6.3.3 Sending a test email

You can send a test email by using the 'mail' command. The first parameter of the mail command is the destination email address. After entering a subject, enter the message body. A '.' on an empty line signals the end of the message.

For example:

```
comma@comx1:~$ mail info@farsouthnet.com
Subject: Test email
This is a test email.
This is line 2.
.
Cc: test-user@farsouthnet.com
```

6.3.4 Monitoring outgoing email

Each outgoing email attempt and the result of the delivery attempt is recorded in the sendmail log files. To view, from the command line, use one of the following commands:

```
less /var/log/mail.info (navigate through the log)
tail /var/log/mail.info (list the last few lines)
tail -f /var/log/mail.info (real-time monitoring)
```

6.4 Asterisk PBX software management

6.4.1 Accessing the Asterisk PBX console

Log in using ssh as described in 6.1.2.2. From the console, type:



```
sudo asterisk -vvvr
```

'sudo' executes the command with root privileges.

'asterisk' launches the CLI (command line interface) for the PBX software subsystem.

'-vvvr' indicates three levels of verbosity (v) enabled, and remote console (r) access.

6.4.2 Stopping and restarting the Asterisk PBX software

For certain maintenance tasks it may be necessary to temporarily stop or restart the AsterisPBX software.

6.4.2.1 Restarting Asterisk

To restart the Asterisk PBX software:

```
sudo /etc/init.d/asterisk restart
```

6.4.2.2 Stopping Asterisk

To stop the Asterisk PBX software, waiting until there are no longer any active calls, allowing new calls in the interim:

```
sudo asterisk -rx "stop when convenient"
```

To stop the Asterisk PBX software, waiting until there are no longer any active calls, without allowing new calls in the interim:

```
sudo asterisk -rx "stop gracefully"
```

To stop the Asterisk PBX software immediately:

```
sudo /etc/init.d/asterisk stop
```

6.4.2.3 Starting Asterisk

To start the Asterisk PBX software:

```
sudo /etc/init.d/asterisk start
```

6.4.2.4 Notify when no calls

In some situations you may want to wait until there are no calls on the system before performing some action. The script below will print the message "0 active channels" when no calls are active:

```
#!/bin/bash
while [ 1 -lt 2 ]; do
    printf "."
    sudo asterisk -rx "core show channels" | grep -x "0 active calls"
    sleep 1
done
```

6.5 FreePBX access

Though we strongly recommend using the Com.X GUI for all configuration tasks, occasional advanced configuration might be required that necessitate the use of FreePBX.

To access the FreePBX configuration GUI, point your browser at:



`http://<ip-of-pbx>/freepbx`

The login username is 'admin' and the password is the same as the password configured for the Com.X GUI.

6.6 PXE booting

The Com.X supports PXE LAN booting, provided a 32-bit kernel is used and console redirection enabled. This section includes examples of PXE boot configurations for the Com.X

6.6.1 Architecture

Please ensure that the boot kernel image provided is a 32-bit architecture image. 64 bit Server images are not supported on the Com.X servers.

6.6.2 Example PXE configuration

Please ensure that the console is set to 115200.

6.6.2.1 NFS

```
DEFAULT nfs
LABEL nfs
  kernel ubuntu-installer/i386/vmlinuz-2.6.24-24-generic
  append vga=normal initrd=ubuntu-installer/i386/initrd-nfs.gz root=/dev/nfs
  nfsroot=192.168.0.75:/hardy-chroot ip=dhcp
  interface=eth0 -- console=ttyS0,115200
PROMPT 0
TIMEOUT 0
```

6.6.2.2 Auto seeded installer

```
SERIAL 0 115200 0
DISPLAY ubuntu-installer/i386/boot-screens/boot.txt
DEFAULT install
LABEL install
  kernel ubuntu-installer/i386/linux
  append debian-installer/locale=en_ZA console-setup/ask_detect=false
  console-setup/layoutcode=us netcfg/get_hostname=fsn-commaX
  interface=eth0 auto url=http://192.168.0.75/boot/auto.seed vga=normal
  initrd=ubuntu-installer/i386/initrd.gz
  -- console=ttyS0,115200n8
PROMPT 0
TIMEOUT 0
```

6.7 Upgrading the software

Com.X software updates, including Dahdi, mISDN, Comma Manager, Comma Firmware, Asterisk and Comma-GUI updates are published from time to time on the Far South Networks repository.

6.7.1 Updating the PBX software

Each Com.X unit ships with the Far South Networks repository configured in `/etc/apt/sources.list`:

```
deb http://update.commanet.co.za/repo/comx-1.1.x hardy main
```

To update your software, follow the following steps:



1. **Important:** Back up your current configuration. Please see 2.3 for more detail.
2. Log into the system as described in 6.1.
3. Stop asterisk as described in 6.4.2
4. If you are required to use a proxy for HTTP access, configure the proxy as follows:

```
sudo su
export http_proxy= http://<ip-of-proxy>:<port>/
```
5. Perform the update as follows:

```
sudo aptitude update
sudo aptitude install comx-base comx-gui
```
6. **Important:** Answer Yes or Y to all prompts
7. Check the installation output for any errors
8. Reboot the system as described in 6.2
9. Confirm system operation and configuration integrity

Confirm the correct (updated) package versions by logging into the GUI and selecting Start – Configuration – Setup – Modules.

6.7.2 Updating the Com.X iTA application image

The iTA application image is responsible for hardware detection, card configuration and TDM stream setup, processing and management. To update:

```
sudo aptitude update
sudo aptitude install commaimg
```

Application images have filenames of the form comma_<major>.<minor>.img and are around 300K in size.

Restart the comma manager to restart all iTAs connected to the network and initiate download of the updated application image:

```
sudo /etc/init.d/commamgr restart
```

6.7.3 Updating the Com.X iTA factory boot image

The iTA factory boot image is responsible for initializing the iTA media processor and retrieving the iTA application image from the Com.X server, and then loading the downloaded image.

Download the latest iTA factory image from:

<http://archive.farsouthnet.com/comma/firmware/>

Factory images have filenames of the form comma_<major>.<minor>_upgrade.img and are around 650K in size.

To install a factory image, copy it to /var/lib/comma/ on the Com.X unit and use comma-console to log into the unit. Use comma-ls to determine the IP address of the unit:

```
sudo cp comma_<major>.<minor>_upgrade.img /var/lib/comma/
comma-ls
```



```
comma-console <ip>
From the comma-console:
update comma_<major>.<minor>_upgrade <major.minor>
E.g.: update comma_1.42_upgrade.img 1.42
hreset
```

6.7.4 Confirming software image versions

PBX software versions can be listed on the Modules page by logging into the GUI and selecting Start – Configuration – Setup – Modules.

Use `comma-ls -v` to confirm that the unit is using the correct **factory** and **application** image:

```
comma@fsn-pbx:$ comma-ls -v
Probing for devices...
MAC Address      = 00:50:c2:97:91:bb
  Host Name      = ita
  IP Address     = 192.168.10.2
  Subnet Mask    = 255.255.255.0
  Default Gateway = Unassigned
  Server IP Addr = 192.168.10.1
  Boot File      = comma.img
  Hardware Config = cards=sn--b-
p,bootstate=app,hw=MkII(4),fpga=u/1.41,sw=MkII(1.49),ser=FSNPBX,state=ready
```

The `fpga=u/<version>` field indicates the factory image version. If this version indicates 'f', the iTA is using the original factory image. If this version indicates 'u', the iTA is using an upgraded image.

The `sw=Mk<model>/<version>` field indicates the iTA media processor card model and application image version.

6.8 Setting the system's date, time and timezone

The Com.X unit tries to obtain date and time from an NTP server on the internet. Failing this, or in the case where the date / time discrepancy on a unit is large, the date and time can be configured as follows:

1. Log into the system as described in 6.1.
2. If the system is unable to contact an internet NTP server to derive time from, date and time might be presented using UTC (GMT+0) which may not be desirable.
3. To reconfigure the system's timezone, issue the following command and select the correct settings:

```
sudo dpkg-reconfigure tzdata
```

4. Set the date / time using the 'sudo hwclock' command and the format "mm/dd/ccyy hh:mm:ss"



5. For example, to set the date to 10 Mar 2011 and the time to 18:12:

```
sudo hwclock --set --date="3/10/2011 18:12"
```

6. Propagate the hardware clock setting to the running operating system:

```
sudo hwclock --hctosys
```

7. Confirm that both the hardware clock and the operating system time matches:

```
sudo hwclock
sudo date
```

8. Ensure that the commamgr is logging using the correct time. **Warning:** this will disrupt service on all iTAs and analogue interfaces:

```
sudo /etc/init.d/commamgr restart
```

6.9 NTP server configuration

The Com.X platform ships with an NTP server installed. It can be configured to serve NTP to local devices by editing `/etc/ntp.conf`.

Add a restrict entry as in the example below, which configures the Com.X to serve NTP on LAN 2:

```
restrict 192.168.101.0 mask 255.255.255.0 nomodify notrap
```

Restart the NTP server for the changes to take effect:

```
sudo /etc/init.d/ntp restart
```

6.10 TFTP server configuration

The Com.X platform may be configured to provide TFTP services for phone auto-provisioning purposes.

Obtain the necessary packages from the Far South Networks repository:

http://archive.farsouthnet.com/support/tftp/tftpd_0.17-15ubuntu1_i386.deb.deb

http://archive.farsouthnet.com/support/tftp/xinetd_2.3.14-5_i386.deb.deb

Copy the files to the Com.X and install the packages:

```
sudo dpkg -i ./xinetd_2.3.14-5_i386.deb.deb ./tftpd_0.17-15ubuntu1_i386.deb.deb
```

Configure the TFTP service as follows:

Create the file `/etc/xinetd.d/tftp` and put this entry:

```
service tftp
{
protocol      = udp
port         = 69
socket_type  = dgram
wait        = yes
user        = nobody
server       = /usr/sbin/in.tftpd
server_args  = -s /tftpboot
```



```
disable      = no
}
```

Create a /tftpboot directory and assign the correct permissions:

```
sudo mkdir /tftpboot
sudo chmod -R 777 /tftpboot
sudo chown -R nobody /tftpboot
```

Restart the service:

```
sudo /etc/init.d/xinetd restart
```

6.11 Video call support (H.263 and H.263+)

The Com.X platform may be configured to support video calls. Care should be taken to evaluate potential performance impacts of video codec translation on the platform in question.

To enable video support:

Add the following to the [general] section of /etc/asterisk/sip.conf:

```
videosupport=yes
```

In /etc/asterisk/sip_additional.conf, on extensions supporting video, add h263 and h263p to the supported codec list.

Restart the PBX.

Ensure that all phones that take part in video calls have these codecs included in their codec list.

6.12 Recovering / resetting the Com.X comma user password

While GRUB is loading, press escape and select recovery mode and select “Drop to the root shell prompt”.

If the system was booted from a USB disk, mount the hard drive as follows:

```
mount /dev/sda1 /mnt
chroot /mnt
```

Change the comma user password as below:

```
passwd comma
```

Enter the desired password.

```
reboot
```

6.13 Mounting a USB drive

Note: This section is informational only. Far South Networks do not provide hardware, software or functional support for non-Far South Networks devices connected to the Com.X

In order to mount a USB drive on the Com.X, first determine where the operating system is recognizing the drive device:



```
sudo fdisk -l
```

Create a mount point for the device in order to access files:

```
mkdir /mnt/usbdrive
```

Place an entry for the device in `/etc/fstab` as follows, using the device name as determined previously (`/dev/sdb1` in this example), specifying the device type (`ntfs` in the example):

```
/dev/sdb1 /mnt/usbdrive ntfs defaults 0 0
```

Mount the device (the device should also automatically mount on next system restart):

```
sudo mount -a
```

If the UUID of the drive is available, it might be preferable to mount the device by UUID instead of device name:

```
sudo hdparm -I /dev/sdb1
```

Confirm that the device has been mounted using:

```
mount
```

6.14 Adding static routes to the network

It may be desirable to add static routes for routing of specific subnets via specific gateways or over specific interfaces.

Applying networking changes from the Com.X GUI over-writes the `/etc/network/interfaces` file. Any changes made to this file may be lost after an apply.

Rather, static routes should be placed in `/etc/network/if-up.d/static-routes`. This script will be run when-ever an IP interface comes up. The interface that triggered the script is available via the `IFACE` environment variable.

For example:

```
#!/bin/bash
if [ "$IFACE" == "eth0" ]; then
    sudo ip route add x.x.x.x via y.y.y.y
fi
if [ "$IFACE" == "eth1" ]; then
    sudo ip route add x.x.x.x/24 dev eth1
fi
```

Ensure the script is executable:

```
sudo chmod +x /etc/network/if-up.d/static-routes
```



Restart the interface or alternatively the Com.X:

```
sudo ifdown eth<N> && sudo ifup eth<N>
```

Note: It is not recommended to have more than one default route in the Com.X routing table.

6.15 Booting the Com.X in recovery mode

6.15.1 Creating a USB boot disk

It may be necessary to create a USB boot disk in order to enter recovery mode (e.g. if hard drive errors occur and the hard drive boot sector and applications have been compromised.)

Create a bootable USB recovery disk by downloading UNetBootin from <http://unetbootin.sourceforge.net/>

Download an Ubuntu 8.04 LTS alternative CD image (approximately 390 Mb)

Create the USB boot disk using UnetBootin and the CD image

Insert the USB boot disk in to the USB port

Power on the Com.X and press TAB to access the BIOS

Configure the BIOS to boot from USB by setting:

```
Advanced -> USB Configuration -> USB 2.0 Controller Mode = "Full speed"
```

```
Boot -> Hard Disk Drives -> 1st Drive = USB
```

```
Boot -> Boot Device Priority -> 1st Boot Device = USB
```

Boot from the USB disk. While booting, select recovery mode and (if necessary to enable serial port interaction) edit the boot like to include: `-- console=ttyS0,115200`

If required, mount the hard drive as follows:

```
mount /dev/sda1 /mnt
```

```
chroot /mnt
```

6.15.2 Booting in recovery mode

While GRUB is loading, press escape and select recovery mode and select "Drop to the root shell prompt".

From the root recovery shell, perform the necessary tasks, such as scanning and fixing hard drive problems, or resetting the comma user password.

Once completed, reboot the system, enter the BIOS and remove the USB drive from the boot priority list.

6.16 Reinstalling the Com.X

Warning: Reinstalling the Com.X except when under instruction of Far South Networks will void the software warranty on the system!



6.16.1 Configuring the installation media

Installation required an Ubuntu 8.04.4 LTS Alternative CD. The media needs to be pre-configured as follows:

Edit the `/isolinux/isolinux.cfg` file in the ISO image, and ensure that the Label install section contains the following:

```
file=/cdrom/preseed/ubuntu.seed initrd=/install/initrd.gz quiet -- console=ttyS0,115200
```

Edit `/cdrom/preseed/ubuntu.seed` and add the following:

```
d-i base-installer/kernel/image select linux-image-2.6.24-27-generic
tasksel tasksel/first multiselect standard
d-i pkgssel/include string openssh-server
```

6.16.2 Installing the operating system

Connect a CD drive to the Com.X's second SATA port, or use a CD image in the Com.X's USB port.

Configure the BIOS (press tab on boot, using serial as described in section 6.1.2.1)

Configure the hostname to be the same as the serial number printed on the bottom of the unit, e.g. FSN1X000001

Accept the defaults for partitioning

6.16.3 PBX software installation

Edit `/etc/apt/sources.list` and add the following:

```
deb http://update.commanet.co.za/repo/comx-1.1.x hardy main
```

Then install the PBX software stack:

```
sudo aptitude update
sudo aptitude install misd-modules-2.6.24-27-generic dahdi-modules-2.6.24-27-generic
comx-base commaman3000 asteriskcomma-g729-eval
sudo aptitude install comx-gui
```

When prompted for a MySQL password, always leave the password empty.

If you have erroneously set MySQL password, you can remove it after login:

```
mysql -u root
use mysql;
UPDATE user SET Password=PASSWORD('newrootpassword') WHERE User='root';
FLUSH PRIVILEGES;
```

After clearing the MySQL password you will need to restart the CommaGUI:

```
sudo /etc/init.d/commagui restart
```



7 Third Party Modules

Please note that Far South Networks does not actively support third party modules and installations.

Note: third party configurations and data may not be backed up by the Com.X backup algorithm.

7.1 Telephone Management Systems (TMS)

7.1.1 MAN 3000

7.1.1.1 Installing MAN 3000 support

The Com.X supports logging of call records to a MAN 3000 buffer across ethernet.

To enable support for MAN 3000 on a Com.X unit, from the shell, install the MAN3000 support package:

```
sudo aptitude install commaman3000
```

For Man3000 releases prior to 11.02:

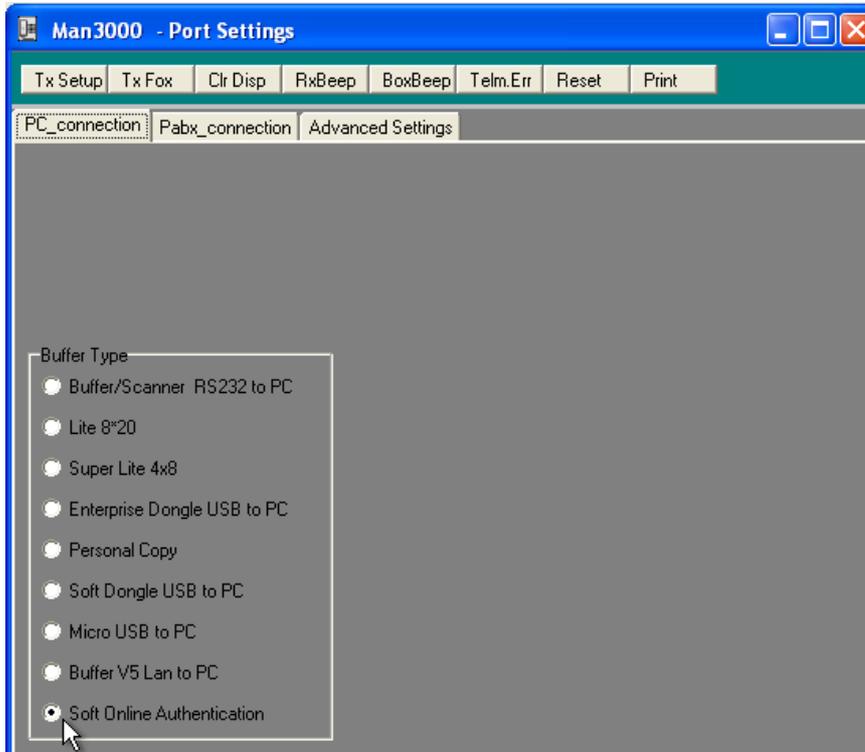
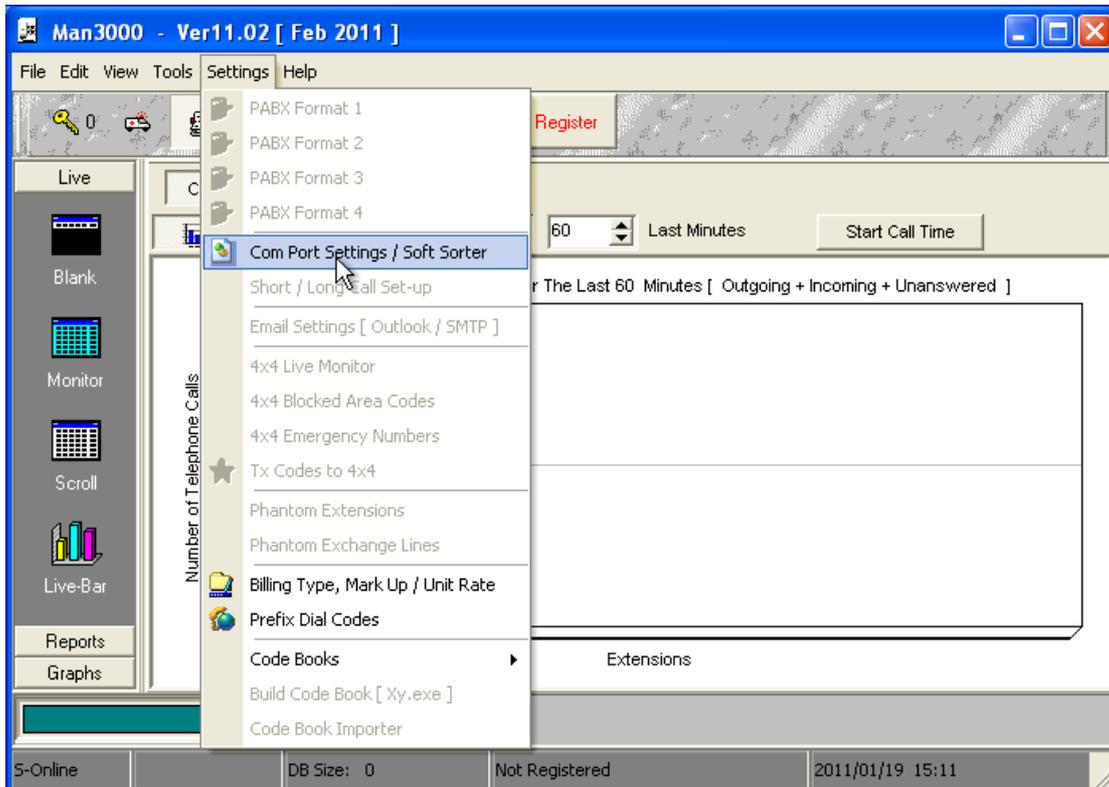
Install the TCPIP2MAN3000 software on the MAN3000 Windows platform from:

```
C:\Program files\Man3000\App Feature Installation\Install_TCPIP2MAN3000.exe
```

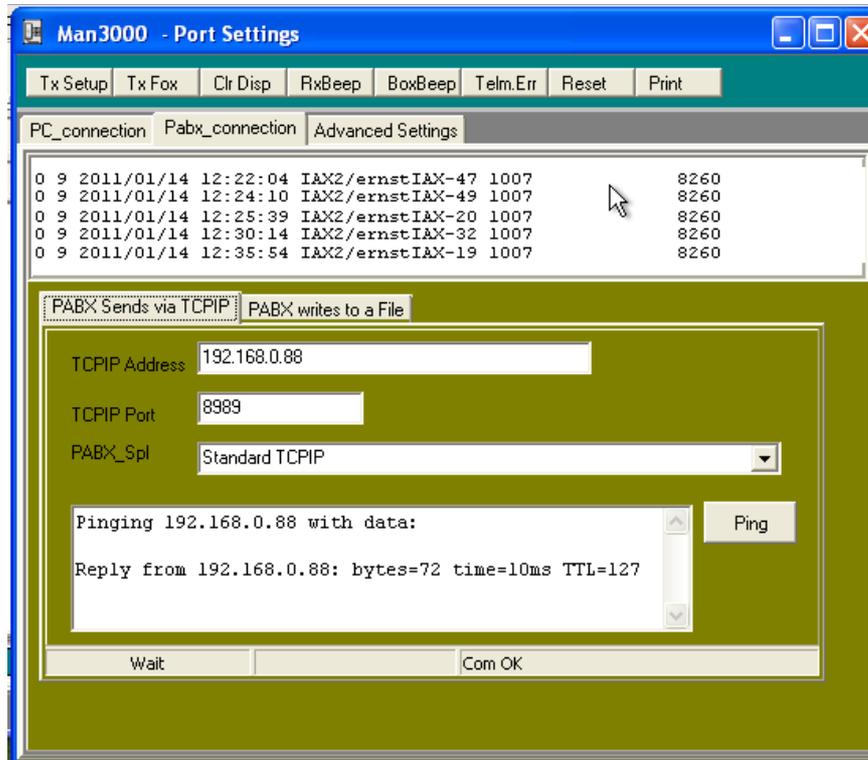
Run the TCPIP2MAN3000 software and configure it with the IP address of the Com.X PBX, using port 8989

For Man3000 releases post 11.02:

Activate the Man3000 TCP/IP configuration by enabling Soft Online Authentication, or your Man3000 USB software dongle:



Configure the TCP/IP settings with the IP address of the Com.X PBX:



Installing the PBX templates:

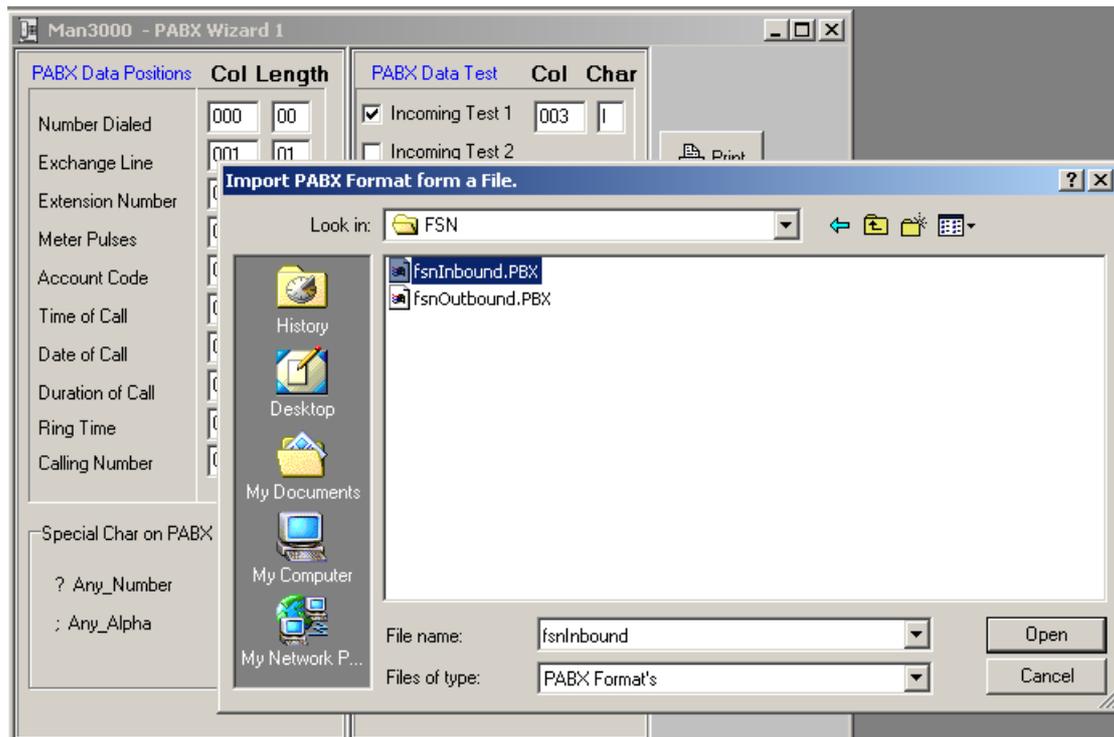
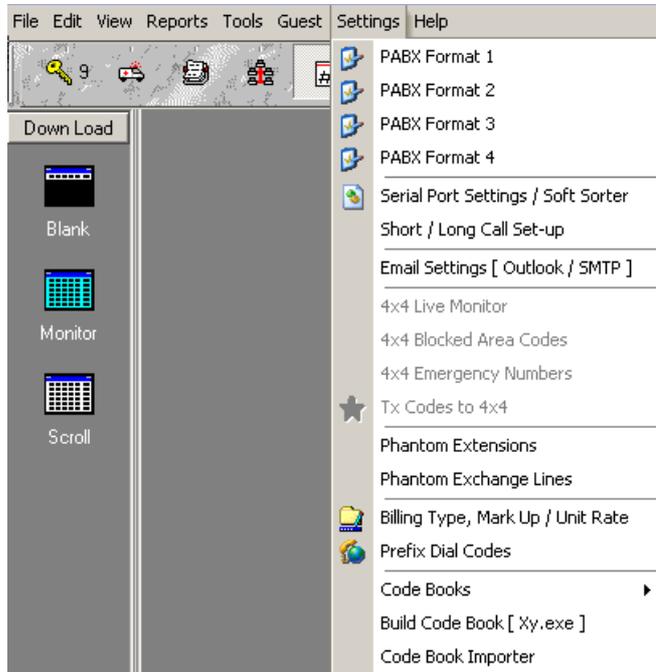
Download the Far South Networks MAN3000 PBX format template from the Far South Networks website at <http://www.farsouthnet.com/resource-center/> (right-click Save Target As, or Save Link As)

Alternatively, download the formats directly (using a downloader such as wget) from:

<http://archive.farsouthnet.com/support/fsnInbound.PBX>

<http://archive.farsouthnet.com/support/fsnOutbound.PBX>

Configure the MAN3000 software to use the Far South Networks Inbound PBX format for PABX format 1 and Outbound PBX format for PABX format 2:





7.1.1.2 CDR format

Records are streamed to MAN 3000 in the following format:

Column: 1	2	3	4	5	6	7	8	9
Type / Line	Type / Line	Date	Time	Channel	Source	Destination	Bill	Account Code

Field	Content
Type / Line	In the first column, a value of 'I' indicates that the record is for an incoming call. Any other value indicates the line number as defined in /etc/commagui/man3000.patterns for an outgoing call, or 9 in case the channel for the outgoing call was not pattern matched.
Type / Line	<p>In the second column, a value of 'O' indicates that the record is for an outbound call or 9 as the line number for an incoming call.</p> <p>This process cannot correctly cater for all conditions, since the specific configuration details of any given system is variable. The outbound pattern match requires the following conditions in order to match a call as outbound:</p> <ul style="list-style-type: none"> • The call context must be “from-internal” • The destination may not be a local extension • The destination may not include “Local” • It may not be a follow-me call or internal transfer. <p>If any of these conditions are not met, the call is flagged as internal.</p> <p>Configuring a line number pattern match as described in section 7.1.1.3 overrides the checks above and is the recommended way of ensuring that outbound calls are correctly classified.</p>
Date	The date of the start of call (including channel access / call setup / ringing) in the following format: ccy/mm/dd
Time	The time of the start of call (including channel access / call setup / ringing) in the following format: hh:mm:ss
Channel	The channel from which the call originated in the inbound case or over which the call was placed in the outbound case. The values of this field will be pattern matched against the entries in /etc/commagui/man3000.patterns to determine the line number. If a match cannot be made, the line number is set to 9.



Field	Content
Source	<p>The originating number for the call. This can be the CID of the caller for an incoming call, or the CID (extension number) of the caller for an outgoing call.</p> <div style="border: 1px solid black; padding: 5px; background-color: #f0f0f0;"> <p>Note: If the CID on the trunk used to place the outgoing call overrides the caller's CID (e.g. extension number), the trunk's override CID will be logged here.</p> </div>
Destination	The destination number of the call in the outbound case, or the destination extension or service of the call in the inbound case.
Bill	The billable duration of the call in seconds, excluding call setup (ring) time.
Account Code	The account code assigned to the Com.X, Flex path or extension (defaults to 0000).

Example CDR output:

```

9 I 2010/05/07 06:36:23 n/a          9905          *97           37  0000
9 I 2010/05/07 06:56:11 n/a          9906          *98           13  0000
9 I 2010/05/07 06:56:38 n/a          9906          *98           3   0000
9 I 2010/05/07 06:56:43 n/a          9906          *98           3   0000
2 I 2010/05/07 06:59:15 SIP/9905-0000e9 9902          9905          18  0000
9 I 2010/05/07 07:12:39 n/a          9930          *97           20  0000
2 I 2010/05/07 07:20:46 SIP/9905-0000e9 9906          9905          5   0000
2 I 2010/05/07 07:39:04 SIP/9901-0000e9 9921          9901          22  0000
0 3 2010/05/07 07:39:52 mISDN/1-u3809  01154XXXXX   01266YYYYY   26  0000
2 I 2010/05/07 07:40:23 SIP/9901-0000e9 9921          9901          12  0000
3 I 2010/05/07 07:42:08 mISDN/1-u3813  01154XXXXX   02159ZZZZZ   37  0000
3 I 2010/05/07 07:42:24 mISDN/2-u3818  01154XXXXX   01231AAAAA   25  0000

```

7.1.1.3 Defining outbound line numbers

By default all outbound calls will be labeled with line number 9 to indicate that the outbound line is unknown to the Com.X. One-digit line numbers (0-8) can be assigned to outbound channels for recognition in the MAN 3000 system.

This assignment is done by creating a `/etc/commagui/man3000.patterns` file in the Com.X file system. This file can be created on a separate system or laptop and copied onto the Com.X as follows:

```
scp man3000.patterns comma@<ip-of-com.x>:/etc/commagui/
```

The `man3000.patterns` file contains entries of the following format:

```
pattern=line
```

For example:

```
mISDN/=1
```

```
Zap/=2
```



SIP/hookd=3

Pattern matching matches the <pattern> if it is part of the outgoing call record channel field. If more than one entry matches, the last entry matched determines the line number. The same line number may be assigned to multiple patterns.

7.1.1.4 Logging records from a specific date

The Com.X remembers the last record logged and will only log records once new records have become available. To configure the Com.X to start logging records again at some date in the past, modify the `/var/lib/commaman3000/lastdate` file located in the Com.X file system. This file can be created on a separate system or laptop and copied onto the Com.X as follows:

```
scp lastdate comma@<ip-of-com.x>:/var/lib/commaman3000/
```

The format of lastdate is as follows: `ccyy/mm/dd hh:mm:ss`

For example:

```
2010/05/04 10:15:34
```

In this example, all calls placed after 10:15 and 35 seconds on 4 May 2010 will be streamed to MAN3000.

7.1.2 TNG

The TNG system supports both importing of Asterisk CSV call records as well as real-time CDR interaction using the AMI interface.

7.1.2.1 Configuring the Com.X for TNG support

- Enable Master.csv support as detailed in section 5.2.2
- From the SSH command line (please see section 6.1.2.2), create a new manager configuration file in `/etc/asterisk/manager.d` (e.g. `comma.conf` in the example below):

```
[comma]
secret = farsouth
deny=0.0.0.0/0.0.0.0
permit=192.168.0.75/255.255.255.0
read = system,call,log,verbose,command,agent,user
write = system,call,log,verbose,command,agent,user
writetimeout=10000
```

- Reload the Asterisk Manager in order to recognize the new user:

```
sudo asterisk -rx "reload manager"
```

- If required, set up a script to periodically update the local Master.csv file from the one generated by the Com.X (located in `/var/log/asterisk/cdr-csv/Master.csv`).

7.1.2.2 Installing TNG

- To install TNG, download the latest TNG for Asterisk application from <http://www.datatex.co.za/>



- To configure TNG to use the Master.csv as the primary source of CDR record, when installing TNG, select the Interface Method as Asterisk CSV File Import.
- To configure TNG to use real-time AMI interaction as the primary source of CDR records, when installing TNG, select the Interface Method as Asterisk Manager Interface
- Enter the relevant Com.X PBX information, including the manager username and password configured above.

7.1.3 Asterisk TMS systems

The Com.X may support TMS systems that are compatible with Asterisk.

We strongly recommend installing third party TMS software on a separate machine, accessing the Com.X CDR files and database with read-only access.

Third party TMS systems have access to Com.X CDR records via one of the following tools:

- After installation of the commaman3000 package, CDRs are served on the port described in section 7.1.1.
- By default the commaman3000 server module will only output new CDR entries every 10 seconds. To disable this back-off and deliver data on every request, edit the file `/usr/bin/commaman3000srv` and remove the `sleep 10` statement at the end of the file
- Downloading and parsing the Master.csv file (please see section 5.2.2)
- Accessing the CDR database (please see section 5.2.3)

7.1.4 P\$X (PBillX) hotel management system

The Com.X has been successfully tested and integrated with PBillX Inc's hotel management software suite. For installation instructions and contact details, please see PBillX Inc at <http://www.pbillx.org>

7.2 Asterisk Flash Operator Panel

For detailed information on configuring the Asterisk Flash Operator Panel, please see <http://www.asternic.org/documentation.html>

A graphical, flash operator panel can be installed for the Com.X, enabling operators to see extension status and activity, and to perform a number of useful actions on extensions.

To install the additional panel, gain access to the Com.X unit as described in 6.1.2.2 and install the additional panel:

```
sudo aptitude install freepbx-panel
```

Once installed, configure the operator panel according to your preferences by editing the `/etc/op-panel/op_server.cfg`. A minimal configuration would entail enabling call events:

```
event_mask=call
```



Configure the extensions to be monitored in the `/etc/op-panel/op_buttons.cfg` files. A typical extension configuration would be:

```
[SIP/300]
Position=1
Label="Reception"
Extension=300
Context=local
Icon=2
Mailbox=300@localvm
```

Once configured, restart the operator panel:

```
sudo /etc/init.d/op-panel restart
```

Access the panel by entering the following URL in a browser:

```
http://<comX1-ip-address>/freepbx/panel.php
```

For each extension configured, the extension should appear. The extension should change colour depending on its status. Selecting the down arrow for an extension presents an extension menu. Actioning an extension during a call will hang up the call. Dragging one extension onto another attempts to establish a call.

7.3 Hylafax fax server

Hylafax offers the functionality to route received incoming fax calls and forward the faxes to email. To install hylafax on the Com.X platform, follow the instructions below:

7.3.1 Installing the necessary packages

Log on to the Com.X server as described in section 6.1.2 and install the following packages:

```
sudo aptitude install libspandsp6 iaxmodem hylafax-server
```

7.3.2 Configuring hylafax receivers

Create a number of fax receivers to receive your faxes. Create as many receivers as you anticipate you will have simultaneous incoming faxes.

For each receiver, perform the steps below. `<index>` indicates the index of the receiver, e.g. 0,1,2, etc. The peer name `210<index>` is illustrative and should match the peer names you configure your IAX trunks with:

```
sudo vi /etc/iaxmodem/ttyIAX<index>
```

File contents:

```
device /dev/ttyIAX<index>
owner uucp:uucp
mode 660
port 457<index>
refresh 300
server 127.0.0.1
peername 210<index>
secret 210<index>
cidname Fax<index>
cidnumber 210<index>
codec alaw
```

```
sudo /etc/init.d/iaxmodem restart
```

Check that each device has been created:

```
ls -la /dev/ttyIAX<index>
```



Edit each receiver's configuration file. A sample template receiver configuration file can be found on the Far South Networks archive at <http://archive.farsouthnet.com/support/config.ttyIAX0>

```
sudo vi /var/spool/hylafax/etc/config.ttyIAX<index>
```

Configure the CountryCode, AreaCode, FAXNumber, LongDistancePrefix and InternationalPrefix fields.

Disable token requirements for the local IAX fax trunk clients:

```
sudo vi /etc/asterisk/iax_general_custom.conf
```

Add the following entry and reload IAX:

```
calltokenoptional = 127.0.0.1/255.255.255.0
sudo asterisk -rx "iax2 reload"
```

7.3.3 Configure the fax DID to email mapping:

Faxes received by Hylafax can be mapped to email addresses based on the name of the IAX trunk the fax arrived via or the DID dialed. If the Flex Path is configured as set out in this chapter, the incoming DID number should be passed on to the Hylafax IAX trunk and should be available for mapping in the FaxDispatch configuration.

```
sudo vi /var/spool/hylafax/etc/FaxDispatch
```

An example:

```
case "$CALLID4" in
2101)
SENDTO="matchedontrunkid@somewhere.com"
;;
3174)
SENDTO="matechedonincomingDID@somewhere.com"

*)
SENDTO="generalreception@somewhere.com"
;;
esac
```

Restart hylafax:

```
sudo /etc/init.d/hylafax restart
```

7.3.4 Incoming routing configuration:

In order for hylafax to receive incoming calls, you need to create IAX trunks, add them to an outbound route and route your incoming fax calls by DID to the outbound route. As a fax call is processed, one of the IAX trunks should route the call to the hylafax receiver. Subsequent calls would fail over to the next available IAX fax trunk, allowing handling of multiple simultaneous incoming fax calls.

In the Com.X gui (<http://<ip-of-your-comx>>) select Start - Configuration -

Add a number of IAX trunks according to the number of simultaneous inbound calls you wish to route. Use the names of the fax receiver accounts configured in section 7.3.2.

To perform incoming-trunk to iax-fax-trunk routing based on the incoming DIDs, use the Flex Path panel on the Com.X GUI:



- Start - Configuration - Outbound panel.
- Create an outbound route called 'incoming-fax-handler' and place all your IAX fax trunks in it. The pattern for the outbound route should match the pattern for DIDs on which fax originate, e.g. '31XX'
- Next select Flex Path. Add a Flex Path called 'incoming-fax-handler'.
- Add a Match route to this path that matches the DIDs you need mapped (e.g. '31XX' and then point the destination to the 'incoming-fax-handler' outbound route.

To intercept incoming calls and route them to fax if they match, yet still pass calls through to normal inbound DID if they do not match the fax dids:

- Add another Match route to the 'incoming-fax-handler' path. Match all calls '.' and point the destination to Inbound DIDs.
- Also add an include path that includes 'Internal Direct'.
- Make sure the Match route that matches the fax DIDs is listed before the general '.' match route.
- Now select your incoming trunks on which you expect to receive incoming faxes and point them to the 'incoming-fax-handler' flex path.
- Apply.

7.3.5 Receiving PDF faxes

By default, Hylafax stores faxes as PostScript (.ps) files. If you would like HylaFax to store in PDF format, edit `/var/spool/hylafax/etc/FaxDispatch` to include the line:

```
FILETYPE=pdf;
```

7.3.6 Sending faxes

A number of Hylafax clients (commercial as well as free and open-source) for sending email from a variety of operating system platforms are available for download.

In the simplest case, sending a fax involves using `scp` to copy a PostScript file to the Com.X, and then executing the 'sendfax' command as set out below to queue the fax for transmission.

Most printer drivers support printing documents to .ps files, which can then be sent in this manner. A script could, for example, be constructed to monitor a directory shared on the Com.X server, or an NFS or SAMBA directory on a different server, and faxes queued for transmission as they appear in this directory.

```
sendfax -f "Test user <notifyme@farsouthnet.com>" -R -r "Fax title / subject" -c  
"Cover page line 1\nLine 2\nLine 3" -x "Destination company name" -d "user@phone-  
number#subaddress" filename.ps
```

Please see 'man sendfax' for more information.



7.3.7 Viewing fax logs

Individual fax logs are recorded in `/var/spool/hylafax/log` and contains detailed information regarding the fax session. This is an excellent place to start troubleshooting.



7.4 Setting up a streaming music server

Though setting up a streaming server is dependent on the software and hardware platform selected, we include an example configuration here that is compatible with Ubuntu linux, using the free GNU mp3 daemon for your convenience.

Download the GNU MP3 streamer from <http://www.gnu.org/software/gnump3d/>

The .tar.gz file can be downloaded from:

<http://ftp.cc.uoc.gr/mirrors/nongnu.org/gnump3d/gnump3d-3.0.tar.gz>

Alternatively download from:

<http://archive.farsouthnet.com/support/tools/gnump3d-3.0.tar.gz>

Extract the archive and install the software:

```
tar xvfz ../gnump3d-3.0.tar.gz
sudo make install
```

Configure the music location, port and other options by editing the following configuration file, then run the server:

```
sudo vi /etc/gnump3d/gnump3d.conf
sudo /usr/bin/gnump3d
```

Configure the music collection on the streaming server by pointing your browser to:

`http://<ip>:<port>`

Configure a streaming category as in section 4.8.1:

```
/usr/bin/mpg123 -q -s --mono -r 8000 -f 8192 -b 100 -@ http://<ip>:<port>/recurse.m3u
```

Warning: If you incorrectly configure the application entry above, applying might result in the apply freezing up. To fix this, restart the Com.X, correct the application entry, or remove the streaming music on hold category and apply.



7.5 PPPoE support

The Com.X product range uses the Ubuntu 8.04 LTS open source operating system and as such supports a variety of modules and services, including PPPoE for use, e.g. with DSL routers.

7.5.1 Installing the necessary packages

rp-pppoe is released as an RPM. To install on the Com.X, install the necessary build environment and build from the rp-pppoe sources:

```
apt-get install build-essential
apt-get install gcc
wget http://www.roaringpenguin.com/files/download/rp-pppoe-3.10.tar.gz
```

Alternatively use the Far South Networks mirror at:

<http://archive.farsouthnet.com/support/rp-pppoe-3.10.tar.gz>

```
tar xvfz rp-pppoe-3.10.tar.gz
cd rp-pppoe-3.10
./go
```

7.5.2 Configure rp-pppoe

Run the PPPoE setup script:

```
pppoe-setup
```

Configure your connection. Confirm that the default route makes use of the PPPoE connection:

```
route
```

If the default route is not via PPPoE, add or replace it as follows:

```
sudo ip route [add/replace] default via <pppoe-ip>
```

Add the pppoe-start script to your `/etc/rc.local` to ensure that rp-pppoe starts on reboot.



7.6 Microsoft Outlook integration

The Com.X product range is compatible with Microsoft Outlook by means of the Outcall or asttapi plugins.

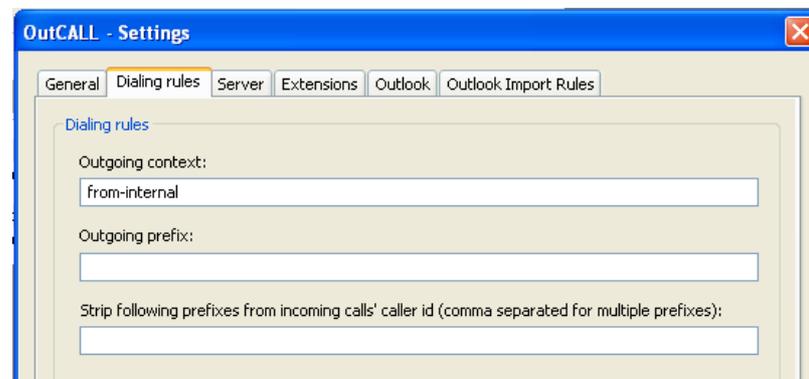
7.6.1 Outcall

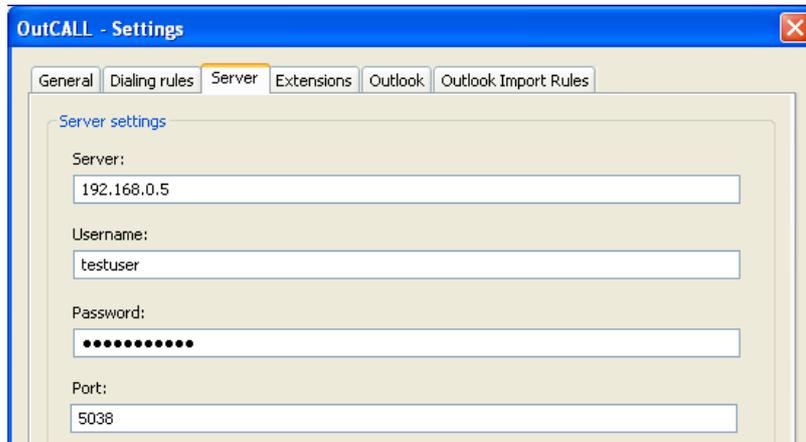
Outcall supports only SIP extensions, but in addition to making outbound calls also pops up contact details for incoming calls. Outcall has been tested with the Com.X using Outlook 2000 and Outlook 2007.

Download the Outcall installer from <http://code.google.com/p/outcall/downloads/list>

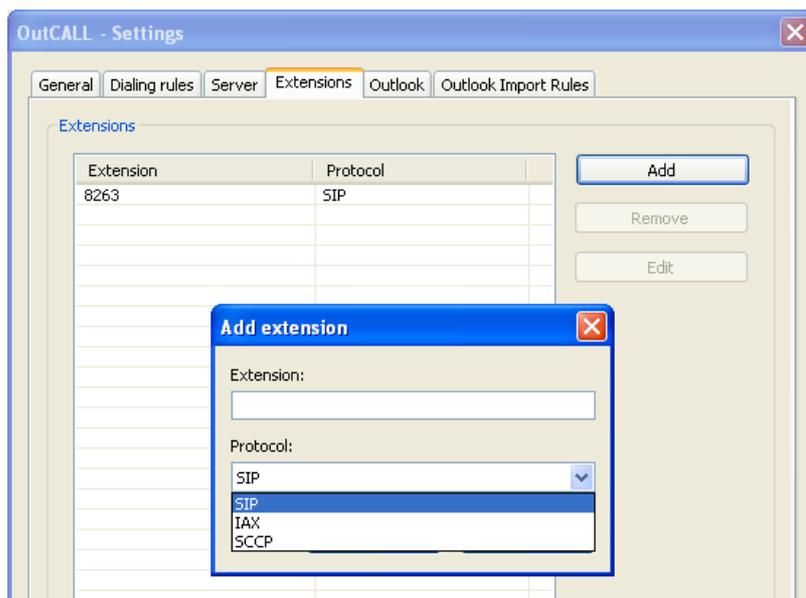
Install the application and configure as follows:

Tab	Field	Value
Server	Server	IP address of Com.X
Server	Username	AMI username, see below
Server	Password	AMI password, see below
Server	Port	5038
Dialing Rules	Outgoing context	from-internal (or the flexpath context for class of service configurations)





Add the extension to be used for the PC running outlook. This must be a SIP extension:



Log in via the ssh shell onto the Com.X and create an AMI user account as follows in `/etc/asterisk/manager.d/outcall.conf`:

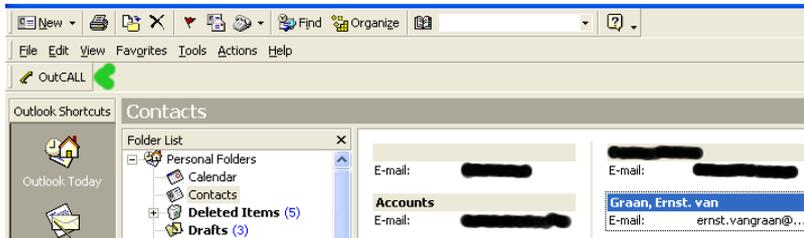
```
[testuser]
secret = fsntestuser
deny=0.0.0.0/0.0.0.0
permit=192.168.0.0/255.255.255.0
read = call,user
write = call,user
```

Restart the manager module:

```
sudo asterisk -rx "reload manager"
```



In Outlook, select the contact to call and select the “Outcall” button. When a call is received by the Com.X, the calling contact, or a new contact window should appear in Outlook.



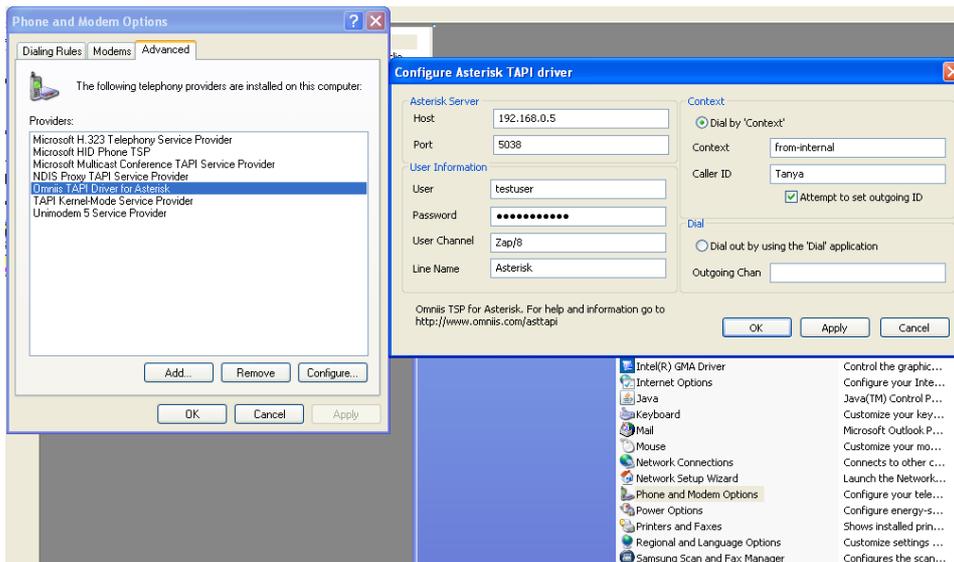
7.6.2 asttapi

asttapi allows dialing from both analogue and SIP extensions, but does not pop up contact details on incoming calls like Outcall.

Download and install the asttapi setup program from <http://sourceforge.net/projects/asttapi/files/asttapi/>

Select Start – Control panel – Phone and modem options

Select the Advanced tab and select 'Omnis TAPI Driver for Asterisk', then Configure

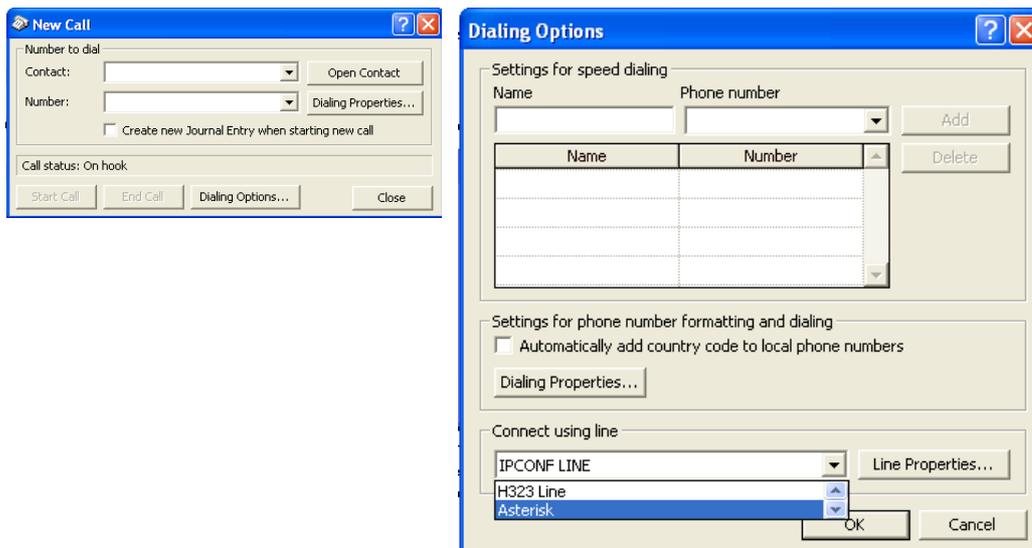




Field	Value
Host	IP address of Com.X
Port	5038
User	AMI username, see below
Password	AMI password, see below
User Channel	The extension channel (e.g. SIP/8263 or Zap/8)
Line name	The name of the line as displayed in Outlook (e.g. Far South Networks Com.X)
Context	from-internal (or the flexpath context for class of service configurations)
Called ID	The caller ID for the extension

Apply the configuration. You may be required to restart Windows.

To place a call, right-click a contact and select “Call contact” and select “Dialing options”.



Select the correct line name. You may have to scroll down the list if the line is not visible.



7.7 fail2ban integration

fail2ban provides a comprehensive and flexible mechanism for preventing unauthorized access to Com.X systems. The fail2ban application acts as a security monitor and firewall manager, detecting multiple failed attempts to access a service (such as ssh or SIP) from the same IP address and then blocks the address for a configured period of time.

7.7.1 Installation

From the Com.X shell:

```
sudo aptitude install fail2ban
```

7.7.2 Configuration

The configuration below blocks the source IP address of 4 consecutive SSH access failures for 10 minutes and 10 consecutive SIP failures for 1 hour:

7.7.2.1 fail2ban service configuration

```
sudo jed /etc/fail2ban/filter.d/asterisk.conf
```

```
[Definition]
```

```
failregex = NOTICE.* *: Registration from '.*' failed for '<HOST>' - Wrong password
            NOTICE.* *: Registration from '.*' failed for '<HOST>' - No matching peer found
            NOTICE.* *: Registration from '.*' failed for '<HOST>' - Username/auth name mismatch
            NOTICE.* *: Registration from '.*' failed for '<HOST>' - Device does not match ACL
            NOTICE.* *: Registration from '.*' failed for '<HOST>' - Peer is not supposed to register
            NOTICE.* *: <HOST> failed to authenticate as '.*'
            NOTICE.* *: No registration for peer '.*' \ (from <HOST>\)
            NOTICE.* *: Host <HOST> failed MD5 authentication for '.*' (.*)
            NOTICE.* *: Failed to authenticate user .*@<HOST>.*
```

```
ignoreregex =
```

Note: Ensure that fail2ban successfully starts after updating the configuration! This can be done by entering `sudo /etc/init.d/fail2ban start` and correcting any configuration errors that may be reported.

7.7.2.2 Jail configuration

```
sudo jed /etc/fail2ban/jail.conf
```

Find and configure:

```
backend = gamin
```

Append to the file:

```
[asterisk-iptables]
```

```
enabled = true
```

```
port = 5060
```

```
protocol = udp
```

```
filter = asterisk
```

```
logpath = /var/log/asterisk/messages
```

```
maxretry = 10
```



```
bantime = 3600
```

7.7.2.3 PBX Logger configuration

```
sudo vi /etc/asterisk/logger.conf
```

Add to the top of this file:

```
[general]
dateformat=%F %T
```

Restart the logger:

```
sudo asterisk -rx "logger reload"
```

7.7.3 Managing the service

To stop fail2ban and allow access to all source IPs:

```
sudo /etc/init.d/fail2ban stop
```

To start fail2ban:

```
sudo /etc/init.d/fail2ban start
```

To restart fail2ban (e.g. after a configuration change):

```
sudo /etc/init.d/fail2ban restart
```

Viewing current blocked IPs:

```
sudo iptables -list
```



8 Troubleshooting

Please refer to the Com.X Troubleshooting Guide for trouble-shooting information.
This guide can be accessed from <http://farsouthnet.com/support/>