



Ultimate Linux Solutions CC

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Server Manual: bender

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Contents

1	System Overview	3
2	Feature Lists	3
2.1	VoIP PBX	3
3	Quick-start guide	4
3.1	Logging in the first time	4
3.2	Creating Your First Group	4
3.3	Creating Your First Extension(s)	5
3.4	Registering Your Phones	5
3.5	Outbound Dialing	6
3.6	Inbound Dialing	6
4	The core system	6
4.1	User Accounts	6
4.2	Personal Details	7
4.3	Server Config Settings	7
4.3.1	VoIP PBX	7
4.3.2	VoIP Virtual PBX	8
4.3.3	Networking	8
4.3.4	Remote API	8
4.3.5	Server Statistics	8
4.3.6	SMS Functionality	9
4.4	System Privileges	9
5	VoIPlets	9
5.1	VoIPlet types	9
5.1.1	Announcement	9
5.1.2	External Call Forward	10
5.1.3	Extension MAP	10
5.1.4	FAX to email	10

Server Manual: bender

5.1.5	IVR	10
5.1.6	Mode Switch	11
5.1.7	Queues	11
5.1.8	Ring Sequence	12
5.1.9	Send SMS	12
5.1.10	Send to Extension	13
5.1.11	Send to Voicemail	13
5.1.12	Time Condition Ladder	13
5.1.13	VoIPlet Sequence	13
6	Outbound Calls	14
7	Special Codes	14

1 System Overview

This section will outline the modules installed on this particular server, and provide a quick description of the purpose of each.

2 Feature Lists

This section will attempt to provide a more comprehensive list of features that each module provides.

2.1 VoIP PBX

The ULS VoIP PBX system is an extensive PBX system with an interface developed internally by Ultimate Linux Solutions. At a cost that is competitive with entry-level analog systems you get the features that compares with the highest end systems out there. Features include:

- **Unlimited SIP extensions:** There is no artificial limit placed on the number of SIP extensions that the ULS VoIP PBX supports.
- **Limited analog + ISDN (BRI) extensions:** The ULS VoIP PBX does support a limited number of analog extensions directly, units like the Grandstream HT502 and other SIP/FXO gateways can be utilized to extend this number.
- **Inter-PBX links:** The ULS VoIP PBX can be configured with the extensions on a peered PBX to enable direct extension to extension dialing on another PBX, either via ISDN or VoIP peerings (both IAX/2 and SIP).
- **Voicemail:** The system supports for per-extension voicemail.
- **Call-Pickup:** The ability for one extension to pick up another ringing extension.
- **PIN codes:** PIN codes can be required to dial out.
- **Call Recording:** The ability to record all calls is built into the system, the capacity of which is limited by available storage only.
- **Extension Groups:** Extensions can be grouped into various groups and/or sub-groups to allow for finer grained control, reporting and managability.
- **Telephone Management System (TMS):** The PBX has a built-in TMS, providing full call details on a per-extension/PIN basis, including call cost estimators (or actual call cost sync when used with Ultimate Linux Solutions CC as Internet-Telephone-Service-Provider), as well as exception reporting (most dialed as well as longest durations).
- **Address-book:** Whilst not fully integrated yet a rudimentary address book allows for providing number to name lookups on outbound calls for the build-in TMS.
- **Full Call Details:** Full Call detail records are available for every group, both inbound and outbound calls.
- **Trunk-balancing:** When utilizing Ultimate Linux Solutions CC as upstream Internet-Telephone-Service-Provider the PBX is able to load balance calls over multiple internet connections, currently up to four but trivially extendable over up to eight. For those with requirements for seriously high call concurrency (in theory a single 4-Mbps DSL line can carry around 40 concurrent calls when utilized correctly).
- **Pattern based call routing:** Outbound calls are matched to a set of patterns, which then determines how the call will be routed, including the ability to configure multiple trunks for a single pattern, resulting in redundant routing (eg, prefer VoIP but fall back to

analog, or vice versa).

- **Various call handling strategies:** Various VoIPlets exist for dealing with inbound calls, including time-conditions, announcements, IVRs, Queues, Ring Sequences (Groups), fax2email and others.

3 Quick-start guide

This section provides a very quick getting started guide. Please note that this really is just intended to get you started and it really is highly recommended to read the rest of this manual in order to more fully understand and be better able to administer the system, the rest of this manual.

3.1 Logging in the first time

By default the system will have a single login available for your use, with username and password as follows:

- **Username:** admin
- **Password:** pleasechangethispassword

In order to log in, please find the link “Login” on the left-hand side navigation and click on it. Once you’ve clicked the link you should be presented with a login prompt requesting your username and password, enter the username and password before clicking on “Log in”.

Once you’ve logged in you should be redirected back to the home page.

It is imperative that you change your password as soon as possible by following the instruction in section 4.2.

3.2 Creating Your First Group

In order to do anything useful with the ULS PBX you need at least one group. For more information please see section ??.

In order to create your first group, on the “system Overview” page, click on “create new” on the “Group Overview” block. Fill in the requested details:

- **Group name:** A name for your group. Generally a company name is used here.
- **Code:** Internally we utilize a group code to handle things. This should be all lower case and consist of alphanumeric characters only.

Once you’ve created a group, you can click on it’s name in the “System Overview” page in order to get to the group’s configuration page.

The group configuration page is from where you will perform the far majority of configuration of the ULS PBX system.

3.3 Creating Your First Extension(s)

You need to be viewing the group configuration page of the group for which you'd like to create an extension, accessed by clicking on it's name from the "System Overview" page.

In the second row of blocks the left-most block contains your "Extensions". By default this list is empty. Just click on "create new". Here you will need to enter the desired extension number and friendly name, and optionally you can override the generated secret.

The only restriction is that the extension needs to be numeric, and not start with a zero, and be at least two characters long.

We of course recommend using three or four digit extensions.

Please ensure that the SIP secrets are secure. Do **NOT** use simple secrets like the extension, 1234, 12345, 12346, variances down the same line, names etc. We cannot stress this enough - there are SIP hacker bots out there, and they are extremely active.

3.4 Registering Your Phones

After you've created extensions you need to register your phones to them. This does vary from phone to phone but the same theme re-occurs. In particular the information you need is as follows:

- **SIP Server:** The IP address of the server.
- **SIP Username** (both register and auth): The extension number.
- **SIP Password:** The secret as per creation above.
- **SIP Transport:** Either UDP or TCP, with different advantages and disadvantages to both. Generally, if you're behind NAT, prefer TCP, if you're not behind NAT, prefer UDP. If you're behind NAT you're welcome to test UDP first, but if you find random call-hangups approximately 30 or 60 seconds into a call your NAT gateway is broken - switch to TCP.
- **STUN Server:** Optional, but you can either use the IP of the server. It runs a basic STUN server by default which you can utilize. In some scenarios you may require a full STUN server, in particular on our Virtual PBX systems the basic variant may not be good enough for your phones, in this case we recommend you try to utilize stun.uls.co.za.
- **CODEC Settings:** Ultimate Linux Solutions CC only allows G.729 for transporting audio data. Thus please make sure your phone is enabled for G.729. And disable "Annex B" or "Voice Activity Detection".

Other options may be required and apply. Mostly the other settings on the phones affect behaviour and should not cause your phone to function or not function.

You can monitor registration status on the group configuration page under the "Extensions" list.

At this point, assuming you've successfully registered at least two extensions they should be able to phone each other at this stage.

3.5 Outbound Dialing

On the Virtual PBX platform outbound dialing is configured automatically and you should be able to dial out from your extensions without any special configuration requirements.

3.6 Inbound Dialing

The short version is that you need to create a VoIPlet and an inbound rule to pass incoming calls to that VoIPlet. The long version is you need to sit down and think about how to best handle incoming calls for your specific situation. You really should browse through section 5 for inspiration.

On the group configuration page, in the second row, far right, click on “create new” in the “VoIPlets” block. Enter a name (alphanumeric only) and a short description. Select “Send to Extension”. “Create” it. You will be redirected to a VoIPlet configuration page, specific to the type of the VoIPlet. For the “Send to Extension” simply select one of your extensions in the dropdown, and leave the “Only if available” on “No” (see section 5.1.10).

Now on the sidenav under “Incoming Calls” we need to direct incoming calls to this VoIPlet. Click on “create new”, and enter the DID (Dialed Number). What exactly needs to go in here varies but usually it’s either the E.164 or nationally formatted number that your ISP has allocated to you. In the case of SIP your best bet is the “Caller ID” that you’ve entered when creating the account. For IAX/2 this is also a good option. In the case of ULS as ITSP this depends on whether “Send Dialed Number” is set to Yes or No. For Yes, use the full 10-digit national number that you’re expecting your callers to dial, otherwise, set to the Caller ID from the “VoIP Providers” page.

At this stage you should hopefully be able to receive incoming calls.

4 The core system

The core system provides the most core elements of the system on top of which Ultimate Linux Solutions CC builds. This includes some user and configuration functionality, amongst others.

4.1 User Accounts

The system allows for any number of user accounts, as long as all user names are unique (they’re not case sensitive). Each account has it’s own set of permissions, groups and possibly some of it’s own user-specific settings.

With the “System Administrator” privilege you’re able to access user administrative functions by utilizing the “User Admin” page (link on the left sidebar navigation).

Once you’ve clicked on the link you will see a list of existing users, giving some basic details regarding the account. To gain more insight into a user, click on the username.

Server Manual: bender

The layout of this page can vary depending on the modules installed. The core of this page however is the administration of the basic user details. Here you're able to alter the "Full name" of a user, enable/disable the account, change the users's password, as well as assign privileges (see section 4.4 for details regarding the privileges), and manage group membership.

The interface should be sufficiently intuitive to not require further explanation.

4.2 Personal Details

Each user is individually allowed to view his settings, as well as edit certain settings for him/herself. This is very similar to the user accounts editing in section 4.1 but is restricted to the user itself, specifically:

1. The user can change his/her "Full name"; and
2. The user can change his/her password by entering a new password.

This can be accessed under the "Personal Details" link on the left hand side.

4.3 Server Config Settings

In order to access the core server settings you require the "System Administrator" privilege. If you have this privilege, there should be a link on the left labeled "Server Config". You can access the server config settings by clicking on this link.

Various modules provides different configuration settings, as per the following.

4.3.1 VoIP PBX

1. **CDR History (months):** Number of historic CDR months to be displayed on sidebar navigation. This does not affect how long CDRs are kept (indefinitely) but merely the searchability thereof.
2. **Agent wrapup time (seconds):** Number of seconds to delay calls to agents that forms part of a Queue directly after that agent has finished a previous call.
3. **Exception Report Size:** Number of line items in exception reports (most dialed numbers, longest calls etc ...)
4. **Preferred SIP transport:** The preferred SIP transport. If set to TCP will go to UDP automatically if TCP is not available for the phone.
5. **Phone VLAN:** If via the auto-provisioning process phones should be assigned to a dedicated VLAN, enter that VLAN number here.
6. **Phone Password:** Some phones has the ability to set an admin password via their auto-provisioning scheme - if this option is set those phones will have their admin passwords set to this.
7. **Phone Syslog:** If you have a syslog server to which phones case log, then you can put it's address here and the phones will send syslog messages there. Only use this for debug use-cases - it's known that some phones runs into problems with this.
8. **SNOM Firmware Series:** Which firmware series to use from SNOM. We provide choices from 7 onwards since older firmware is just too broken to work properly with any modern

Server Manual: bender

version of asterisk, and a 'keep' option which will attempt minor updates only (ie, try and keep each phone in it's major version). In all cases if firmware for the phone cannot be located in the given series it'll be bumped to the next major version.

9. **IVR Option Delay:** In some rare cases upstream providers sends incorrect DTMF start/stop sequences, resulting in duplicate digit presses, this affects IVRs in particular, and adding a delay after reading the appropriate digit results in the follow ups being ignored by asterisk.
10. **Answer before call forward?:** As it turns out in some strange configurations call forwarding fails unless the incoming call channel is already Up. This option allows to explicitly Answer() a call before forwarding. Another option is to play an announcement before forwarding.
11. **Custom Period Days:** Number of days for which custom period should be set by default
12. **Extension Hot Desking:** Whether or not to enable SIP Hot Desking for phones that aren't explicitly configured to an extension. Currently only Yealink will honor this.

4.3.2 VoIP Virtual PBX

1. **Provider Name:** The IAX username that the upstream system will send at us.
2. **Provider Host:** Service name (DNS) to register to (without `.iax.udp` prefix).
3. **DNS Master Host:** The `https://` host to contact for setting up DNS entries.
4. **DNS Zone:** The zone into which to register VPBX systems.
5. **DNS SOAP User:** Username for authenticating SOAP access.
6. **DNS SOAP Password:** Password for authenticating SOAP access.

4.3.3 Networking

1. **Mobile Default Metric:** When new (previously unknown) 3G modems are connected - what should we use as the default metric? Lower values implies higher priorities - use -1 to disable adding a default route by default. A default route will ONLY be added if there are no routes configured for the modem.
2. **Enable IPv6:** EXPERIMENTAL: Please note that this option is considered experimental!

Due mostly to privacy and security concerns we currently avoid IPv6 - if you do want to enable it this is where to do it.

4.3.4 Remote API

1. **WSDL Namespace:** If the default `ULS.$hostname` namespace is not appropriate or you need this to be something specific, please set it here.

4.3.5 Server Statistics

1. **Visibility:** Who should all be able to view the server statistics.

4.3.6 SMS Functionality

1. **Implementation:** Switch between multiple back-end systems.
2. **ULS SMS Username:** Username as per ULS SMS contract
3. **ULS SMS Password:** Username as per ULS SMS contract

4.4 System Privileges

- **VoIP Server Admin:** Administration of the VoIP server component on the server.
- **Incremental CDR Retrieval:** User is allowed to perform CDR retrieval via the SOAP API running on the server.
- **Origination support via SOAP:** The user is allowed to originate calls via the SOAP API running on the server.
- **QA Capturing Agent:** Users with this privilege will have access to perform initial QA on calls, in other words, they will get to fill out the Quality Assurance form and submit it, but not retrieve generated reports, they will simply be provided with calls to perform QA on.
- **QA Reporting Agent:** Users with this privilege will have access to view existing QA reports (and update them should they also have the QA Capturing Agent privilege).
- **VirtualPBX Remote Admin:** This privilege entitles a user to spawn new VirtualPBX instances, and otherwise manage them. It does NOT allow the user to log in etc ...
- **System Administrator:** Full system administrator privileges. With this privilege a user is allowed to do anything that can be done on the system.

5 VoIPlets

The naming for VoIPlets comes from a combination of the acronym “VoIP” and the term “applet”, the function of these components are very specific, and you generally need quite a number of them to achieve complex tasks. You can look at them as small building blocks to build voice applications.

This section will firstly look at the individual VoIPlet types in detail, and will then attempt to explain a few examples of putting them together to facilitate more complex tasks.

It should be noted that many of the types are not terminal, in that they can return to a high-up VoIPlet in order to take alternative action in some cases.

5.1 VoIPlet types

5.1.1 Announcement

This is probably the simplest of all. It plays the selected recording (Section ??) and then returns to whatever called it. If this is used as a first-level VoIPlet the result is that the call will get

hung up, which is in all probability not what you want. Typically used as the first or second step in a VoIPlet sequence (5.1.13).

5.1.2 External Call Forward

A VoIPlet of this type is used to forward a call to an external number, for example, a cell phone. Ultimate Linux Solutions CC utilizes this to forward after-hours calls to a speeddial configured on our core switching infrastructure resulting in a staggered dialing going to the technicians on standby. Very often also used to create “speed dials” by configuring an inbound number of for example “1234” pointing directly to a VoIPlet of this type.

5.1.3 Extension MAP

A VoIPlet of this type attempts to map a sequence of numbers onto internal extensions. A prime example of where to use this is if you for example have a prefix block (eg, 012 345 67XX) and would like to map those last two digits onto your internal extensions. Continuing the example, let’s assume that the internal range is 2XX, then the numbers configured here would be:

Base dialed number: 0123456700
Base extension: 200

There is currently no length set for the sequence, and as can plainly be seen, there can be any random offset between the dialed number and the mapped extension, for example, external number 012 345 6710 could be the base for 100, in which the second-last digit would actually be adjusted, so a person dialing 012 345 6710 would hit extension 100, and a person dialing 012 345 6711 would hit extension 101.

5.1.4 FAX to email

This VoIPlet will instantiate a FAX machine and act as the called station (ie, it will attempt to receive a FAX from the line), and then if receipt is successful, proceed to email it to the configured recipient.

Whilst this application was originally designed and built to function around analog or ISDN infrastructure it will function equally well using SIP permitting that the provider (or source) supports T.38. The only commonly used signalling protocol that does NOT currently work here is IAX/2. Stick your fax numbers on SIP.

5.1.5 IVR

An Integrated Voice Response, or IVR is a voice menu that prompts to user to respond with a key press. Many companies use this as a way to get a caller to dispatch him/herself to the correct department before being dispatched by a switchboard operator dedicated to that department, or even simply to get the caller in the correct call queue.

In spite of the obvious potential for complexity the IVR is surprisingly simple to configure. It

Server Manual: bender

takes a recording (Section ??) to play back to the user (which you can set to repeat a number of times, not recommended to set this to more than two or three).

On special request there was an option added to allow for direct extension dialing, which you can either set global, or restrict to a specific group (subgroups are included). For example, should Ultimate Linux Solutions CC have configured their IVR to utilize this, we would set it to the group “ULS”, and not global, since we share the PBX with other companies, and could potentially divide the extensions internally into departments.

The rest of the config simply lists the various digits, allows you to select an action (Another VoIPlet), as well as to tag the caller id on the call with a prefix (Ultimate Linux Solutions CC utilizes this to tag support calls with SUP, accounting calls with ACC and sales calls with SALES). You’ll note there is also two special options:

- **i**: or invalid. So if the user presses a digit (or dials a non-existent extension) it’ll jump to this voiplet.
- **t**: or timeout. If the user does NOT press a digit it’ll go to this VoIPlet instead of going to the operator.

5.1.6 Mode Switch

More commonly known as a day-night-mode. In a typical day-night switch there are only two possible options, and Ultimate Linux Solutions CC that this is somewhat restrictive, so we allow for up to 10 different modes. Each mode switch gets allocated a number, from 0 onwards, and any person can dial *28X to switch that mode switch, where X is the switch number. This is also stated on the config page for the VoIPlet.

From the config page you’re also able to switch the active mode.

Adding modes, removing and changing the destination VoIPlets should be a trivial task, simply click the “Add Mode” link, which will allow you to add, select the “Del?” tick box, or update by switching the “Cur” radio button. The destination VoIPlets can be changed by selecting an alternative target from the drop-downs.

5.1.7 Queues

A Queue, or more completely, Call Queue, allows for queuing callers into an ordered list whilst they wait for an agent to become available. Whilst conceptually simple there are a few caveats that you need to be aware of.

The first issue is that of static vs dynamic members. A static member is an extension. These cannot be added/removed on the fly by logging in and out (see below). A dynamic member is represented by an agent that can dynamically log in from one phone, or another, and then log back out. Depending on how things work in your organization typically you will only really ever use static members, and then depend on people being at their desks, or properly utilizing do-not-disturb on their phones, or as with many new installs, soft phones, which will register to the system when the user logs in to his desktop, and deregister when the user logs out, resulting in similar functionality to agents.

Server Manual: bender

It will be noted at this point that each member has a penalty assigned to it. The exact meaning of this penalty depends on the queue strategy, but generally the rule of thumb is that a lower penalty results in a higher priority for taking calls.

The different strategies for assigning callers to members are:

- **Ring-All:** Start at the lowest penalty, and ring all available members at that priority, if no-one answers, move on to the next and so forth. Not a widely used strategy in call centers, but it most certainly does have it's use in smaller environments where response-times are critical.
- **Least-Recent:** This will ring the member that has the longest time since last call. Taking penalty into account, so if you have three members, one with a penalty of 5 and two with a penalty of 10, then the first member will receive all calls, unless he/she is not available for taking a call, in which case whichever member from the other two whom had the least recently taken a call will be tried first.
- **Fewest Calls:** A requested strategy, it tries to equalize the number of calls taken. This penalizes people who sits on the phone for a longer duration at a time since the counter goes for number of calls, not when last. This further skewed when adding and/or removing members from a queue.
- **Random:** Randomly select members.
- **Round-Robin:** Within a given penalty group, send the calls in a round-robin manner. Also not a bad scheme but you really should consider least recent.
- **Linear:** ???
- **Weighted Random:** Use the penalty as a weight to select members randomly. A penalty value of 10 is thus 10 times more likely to receive a call than a penalty value of 1.

5.1.8 Ring Sequence

The Ring Sequence provides a way of ringing a set of extensions in a specific sequence, hoping that someone will pick up (or escalating to other people if for some reason the people that usually needs to pick up doesn't). More recently Ultimate Linux Solutions CC has started utilizing Queues more often than not where we need this kind of functionality as the fail-over to backup extensions is typically faster, and we still only ring one extension at a time. The Ring Sequence certainly still has it's place.

Configuring a Ring Sequence is done by adding extensions to a given priority level, where the groups are ordered in ascending order, and then adding extensions to that priority level. You can also specify a time to ring at that group - in seconds.

5.1.9 Send SMS

As an after-hours service some of our customers requested to simply receive an SMS with the details of the person that called, and this VoIPlet was the result. You need an API username and password for this VoIPlet to function, which can be obtained by emailing support@uls.co.za, or requesting one from the organization through whom you've bought the PBX system.

It's very simple, you enter the username and password, the destination number (must be a valid South African cell phone), and the message to send.

Typically this will be utilized as a single step in a VoIPlet Sequence (5.1.13).

5.1.10 Send to Extension

Simply send a call to a specific extension. Useful for non-linear direct inward dialing, but not often used any more. Simply select the extension to send to.

The option “Only if available” will, when set to Yes, result in the VoIPlet declining the call if the destination extension is already busy on another call. This is useful if you have a VoIPlet sequence with fallback options available.

5.1.11 Send to Voicemail

Similar to Send to Extension (5.1.10) but doesn't bother ringing the extension, instead it goes straight to voicemail.

5.1.12 Time Condition Ladder

The Time Condition Ladder is a simple time conditions mechanism. It allows you to select various bits of criteria with respect to the day, and the time of the day, and to then make call routing decisions based on that.

Most often this VoIPlet is used to implement an “office hours” mechanism.

The following criteria can be used:

- **Time of day:** Both a start and stop time.
- **Month days:** If an action needs to start on a specific day of every month, and stop again on another day of the month, eg, if the first week of every month needs to be treated differently, use 1-7 here.
- **Weekdays:** Which days of the week, typically Monday-Friday.
- **Months:** If the action must happen in specific months only.

All options support wrapping, so 23:00 till 1:00 is perfectly valid.

And of course just select the destination VoIPlet for every time.

The ladder is inspected top to bottom, action for first matching row is taken, if that action returns no further checking of the table is done.

5.1.13 VoIPlet Sequence

A VoIPlet sequence is similar to a Ring Sequence, but instead of ringing specific extensions it will simply try a set of VoIPlets, one after another, until one of them answers, or all of them returns, in which case the sequence itself will also return.

6 Outbound Calls

On our Virtual PBX platform outbound calls is automatically configured to allow for the following outbound dialing patterns:

- 00Z. – International calls, blocked on the core switches by default for security reasons.
- 0Z. – National calls (all destinations).
- 857! – ULS special numbers.
- 10[12]. – Telkom special numbers, including emergency numbers 10111.

You do not need to do anything special to get outbound calls working.

7 Special Codes

Code	Function
*8	Call Pickup (Pick up another ringing extension).
*28X	Change the active mode of mode switches. X represents the switch number.
*97	Access to Voicemail for dialing extension.
*98	Access to Voicemail for another extension.
*65	Play back your extension number.
*69	Play back a constant 1004Hz tone at 0dB. (mostly useful for comparing phone volumes - don't dial this with the handset against your ear.)
*56	Record some audio and play it back (debug).
*57	Echos back everything you say (debug).
*78(X)	Listen to the current recording X, if X is not appended it will be prompted for.
*79(X)	Update recording number X, if X is not appended it will be prompted for.
*34	Agent Login (hold on line).
*35	Agent Callback Login (system will call you back on current extension when a call arrives for the Agent).
*36	Agent Callback Logout.
*25	General Conference (conf number will be prompted for).
*25*X	Join conference number X.
*26	Shortcut for *25*EXTEN.
*77	Play back the current time.
*88*EXTEN	listen in on active call for extension EXTEN (only available for admin extensions).
*89*EXTEN	whisper in on active call for extension EXTEN (only available for admin extensions).

=== END ===