Linux-Call-Router

Software based ISDN Private Branch Exchange for Linux

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(http://www.linux-call-router.de)

Documentation for Version 1.1

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Introduction

1.1 What is Linux-Call Router and PBX4Linux?

PBX4Linux started as a pure software based ISDN PBX, that connects external lines, internal telephones, and optionally voice over IP. It is designed to run with Linux only because it uses the kernel's mISDN passive driver by Karsten Keil[1]. It worked together with OpenH323[2], that is a voice over IP implementation complied to the H.323 ITU-T standard. The primary goal is a working ISDN PBX that connects BRI and PRI lines using call routing. Basic applications, like voice box and conferences are implemented. The hardest part was a stable and conform ISDN solution. Allot of work was also done on mISDN[3] driver. Beside protocol improvements and card implementation, I wrote a special audio processor called DSP to make real-time audio processing.

At the same time, Asterisk[4] PBX became popular, due to VoIP switching capability. I found out that ISDN call processing and VoIP call processing is both complex and very different. Since Asterisk lacks in ISDN connectivity and features, I decided to make an Interface between PBX4Linux and Asterisk. Also I removed all VoIP code and simplified code structure and audio processing and concentrate on good TDM call routing and features. I met a company called Beronet[5].

1.2 Philosophy behind

At the end of 2001, I found out, that my ISDN card has a chipset capable of connecting telephones to it. This is called the NT-Mode. So my card can be used to transfer information between telephones and my Linux box. Normally ISDN cards transfer information between a

[1] Karsten Keil is programmer of the famous HiSax kernel driver. It provides ISDN stacks to kernel, mainly for PPPoE connectivity. He is working for SuSe and supports ISDN implementation for SuSe distribution. Later he developed mISDN kernel driver. (www.mISDN.org)
[2] OpenH323 is an open source telefonie stack for VoIP applications using H323 standard. (www.openh323.org)
[3] mISDN is a software ISDN driver for Linux Kernel. It consists of Kernel space drivers and user space stacks. (www.mISDN.org)
[4] Asterisk is a software PBX that is strong in using VoIP protocols. It also supports ISDN and POTS interfaces using Zaptel driver. The core programmer is Digium. (www.asterisk.org)
[5] Beronet provides software solutions and support for Asterisk and mISDN based PBX systems. Also Beronet developes for mISDN. (www.beronet.com)

computer and the public telephone system. But at this time there was no protocol for Linux kernel that could talk to ISDN telephones.

My first idea was to use a telephone to make voice over IP calls. Instead of messing with headsets, I wanted to just pick up the phone, get a dial tone, and dial the IP number. On the other side, I wanted to have another phone, that rings and shows me the IP number of the caller. I know, that there are IP phone already but they are more expensive than ISDN phones. So I started to expand the Linux kernel to be able to handle telephones connected to my ISDN card. The patch was written for the old 'HiSax' driver. I hated that patch because it was another dirty addition. I was glad that Karsten Keil wrote the new modular ISDN driver that gave me a well defined API and real multipoint NT-mode. I added some features to the new driver, so real time cross connections, conferences, DTMF decoding and tone generation is possible.

While writing the expansion of the kernel driver, I designed the PBX4Linux, that is now also capable of connecting calls between connected telephones and external ISDN lines connected to the public telephone system, as well as voice over IP using OpenH323 library.

I wanted to have a PBX, that provides features I am missing in standard products, like letting a call ring on my telephone at home and at the same time on my mobile via external call. The called phone, that picks up first, gets connected, the other gets released. Another idea was callback, which helps me to reduce the costs of mobile calls.

Many features followed: Deletion of digits while dialing after pick-up, an answering machine that records during the announcement to trick the caller, and conferences with no member limit but the resources of available channels.

During this book, you will learn about the features it has, how it works and, how to set up your own software based ISDN PBX.

1.3 Who wrote it?

My name is Andreas Eversberg and I live in the northern part of Germany, called 'Schleswig-Holstein'. I currently work for a telephone company, called Versatel[6], which is a full service provider for telephone lines, internet and leased lines in Schleswig-Holstein and other parts of Germany. I do administration of the network elements like subscriber line cards and ADSL routers. I started writing the PBX and the kernel driver, six months before I got the job at Versatel. I see this job as a start of managing life and getting used to "work" and of course to learn and earn money.

My email address is 'andreas@eversberg.eu'. My homepage is 'www.eversberg.eu'. The PBX4Linux page is 'www.linux-call-router.de'.

1.4 How much does it cost?

It costs nothing at all. It is **GPL**! As long as you use it for non commercial purpose, I will not charge you. Using this software in your company to handle the daily business is not a commercial purpose in this context. See the appendix for copyright information.

[6] Versatel Nord GmbH and Versatel Holding (www.versatel.de)

1.5 Some definitions of terms in this document

➢PBX

Private Branch Exchange (PBX), also called PABX, is a small private telephone system. It features internal calls without getting changed, because calls are routed internally. It has connectivity to the telephone network in most cases. It provides more features than the telephone network provides.

➢exchange / public exchange / local exchange / remote exchange / international exchange / transit exchange

The "exchange" or "public exchange" is the great "PBX" of your telephone company. It can be something like a very large type of "PBX" with thousands of ISDN and analog ports in several racks, or it can be one of these old mechanical types with relays that fill up a complete building. There are different types of exchanges:

- Local exchange means the public exchange connected to the local PBX or party. We talk about the exchange connected to the local user. Local exchanges are equipped with many interfaces to serve many customers.
- Remote exchange means the public exchange connected to the remote PBX . We talk about the exchange connected the remote user.
- International exchange is used to route calls between countries. They are equipped with many interface trunks and use fiber or satellite links to other countries.
- Transit exchange is used to interconnect local and international exchanges together. They have trunks to local exchanges as well as to other transit exchanges. Today there is no need for a dedicated transit exchange, because local exchanges are also capable of routing national and international calls.

➢H.323

H.323 is a standard for making calls via packet switched IP networks. It specifies, how voice (and call control) is transferred over IP. H.323 is defined by the International Telecommunication Union (ITU). To be able to interconnect two persons using H.323, both must have an IP connection. If one party is reachable via public telephone network, an H.323 gateway is required.

➢SIP

SIP is not only a standard for making calls via packet switched IP networks, but it is mainly used for it. There are many other protocols required for making calls over the internet. Since H.323 is complex, heavy and limited to classic voice services, SIP becomes more and more popular. SIP is defined in many RFCs.

➢**Extension**

Extensions are teephones connected to a PBX. Linux-Call-Router supports individual configuration of extension. Calls from non-extensions are just routed, extensions have special applications, like individual voice boxes and call logs. Many telephones can have the same extension's number. Extensions may also ring on external phones. The extension is identified by the outgoing caller ID of the telephone and/or by interface it is connected to. On incoming calls, the internal interface(s) is/are defined by the extension's configuration, to what internal interface to route the call to.

➢**Internal port / interface**

An interface with one or more ports can have an 'extension' flag set, so incoming calls are treated as calls from an extension. Normally telephones connected directly via ISDN card to the PBX will have an extension flag set.

➢**External port / interface**

An external telephone line (from a local telecommunication provider) is connected to the PBX. Multiple external lines may be used for more channels. Since 'extension' flag is not set, calls are treated as external calls.

➢**Interface**

One or a group of one or may internal/external port(s) with a given name. Multiple ports may be grouped to a multi link PBX line. Each port must be assigned to an interface.

➢**a-law / mu-law**

Audio samples are not directly transmitted via B-channel. First a 12 bit sample is sampled from the audio input and then converted into 8 bits. This is done by reducing the dynamics for high amplitude samples, to increase the number of samples available for low amplitude. There are two different 'laws', describing the amplitude ranges. The conversion can easily be done by a table of 256 entries (to decode) and 65536 samples (to encode from a 16 bit sample). In Europe, we use "a-law". In America they use "mu-law". The mu-law also allows to drop the lowest significant bit. This is used in America for telephone exchanges that provides only 56 bits. To read more about the coding, refer to any "how-ISDN-works" manual.

➢**PRI / BRI / G.703 / s2m / S/T / S0 / PMX / up0 / u2m / uk0 / HDSL / SDSL**

There are two different types of ISDN interfaces:

- Primary Rate Interfaces
- Basic Rate Interfaces

The Primary Rate Interface has one 64 bit D-channel and 30 B-channel (channel 0 is used for synchronization). The electric interface is also called "s2m" or "PRI" and is defined in the ITU-Standard "G.703". The bit rate is 2048Kbits/s. The electrical version of a "G.703"

interface is limited to several meters of cable length, so a modem must be used. “HDSL” and “SDSL” provide transmission via none insulated twisted pair copper wires over some kilometers. An older standard is called “u2m”. PRI interfaces are also called PMX (Primary Multiplex).

The Basic Rate Interface has one 16 bit D-channel and two B-channel. The electrical interface is also called “S/T” or “S0”. It supports point to multipoint connectivity. Also this interface has only a limited range of cable length, so “uk0” or “up0” is used to transfer the signal over several kilometers using two wires.

➢**B-channel / D-channel**

An ISDN interface has several types of channels. The D-channel is used to transfer signaling information (data channel). The audio data in transferred vi B-channel (binary channel). Also any data transmission like fax or internet is transferred via B-channel. Some interfaces have overhead, monitor and echo channels. These are only relevant between interfaces and do not transfer information between subscribers.

2

Hardware Requirements

2.1 ISDN cards

The first thing needed for the PBX, is of cause at least one ISDN card. At least two cards are required to interconnect internal telephones with external telephone lines. Any card can be used that is supported by the kernel's **mISDN driver**:

- HFC-S PCI based cards (tested)
- Fritzcard PCI (tested)
- Sedlbaur FAX
- Winbond
- HFC-4s / HFC-8s based cards (tested)
- HFC-E1 based cards (tested)
- HFC-USB
- …

Please mail your experience with untested cards, when using the kernel's mISDN driver.

In order to connect telephones directly to an ISDN card, it must support **NT-Mode**. Then it is possible to use it as internal ISDN port. There currently there are these type of chip sets, that support NT-Mode hardware layer:

- HFC-S PCI based cards
- HFC-4s / HFC-8s chip sets
- HFC-E1 chip set
- HFC-USB based adapters

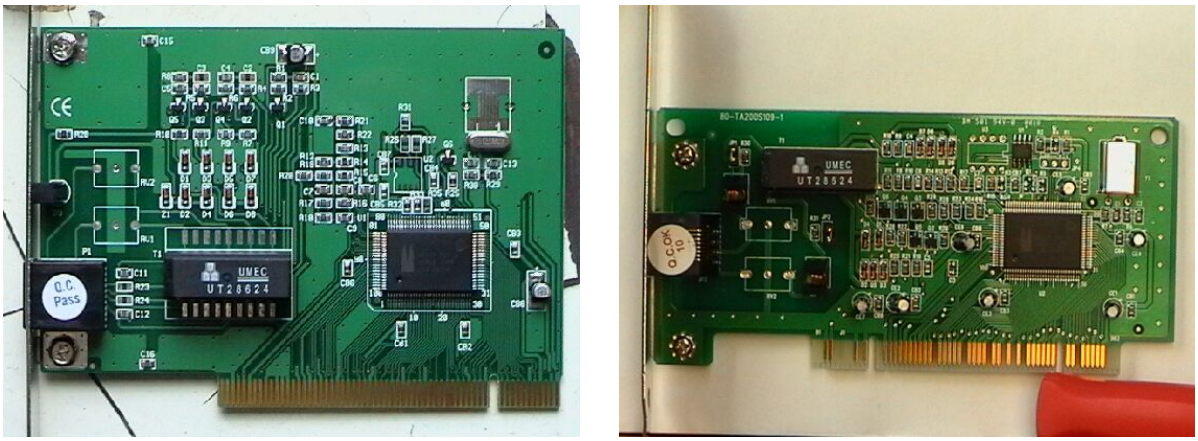
All of these chips have a small 'Dom' of Cologne on the top:

Cologne Chip

The following cards **do** have the required chip set capable of NT-Mode:

- Creatix ISDN-S0/PCI
- Trust PCI-Modem
- Acer ISDN 128 Surf PCI
- D-Link DMI-128I+
- Billion/Asuscom (Asuscom/Askey) (Beware of "BIPAC PCI SE", it has a W6692 chip that DOESN'T WORK)
- HFC cards from .Conrad Elektronik. (Bestellnummer 95 50 78)
- Neolec FREEWAY ISDNPCI (available at Reichelt.de)
- ISDN 128Kbs TA Card (TAS106H)
- HFC-4S / 8S / E1 OEM
- Junghanns Asterisk boards (HFC-4S / 8S / E1) www.junghanns.net
- Beronet Asterisk boards (HFC-4S / 8S / E1) www.beronet.com
- PBX4Linux boards (HFC-4S / 8S / E1)
- Typhoon ISDN Quick Com 128 PCI
- Arowana ISDN 128k
- Conceptronic C128i(R) (snogard.de)
- Conceptronic B128P Bulk ISDN
- Telekom Teledat 220 PCI
- Arowana 128PBS PCI
- mps Software ISLINEuni/ISLINEpro/ISLINEnote
- Micronet 128K ISDN TA Card
- Sitecom DC-105 - PCI ISDN Card
- Microcom ISDN Porte Internal (Czech site, I think)
- E-Tech PCTA128 PCI ISDN board (sold in Holland)
- Dynalink 128P PCI ISDN
- Longshine LCS-8051A (Reichelt)

More cards are listed at www.mISDN.org. Please mail me other cards you know, that have an NT-Mode capable chip sets. The card will look like:



(Figure: Cards with HFC-PCI, thanx to Ulrich for the pics)

2.2 Connect ISDN telephones to your ISDN card.

Normally, cards run in **TE-Mode**, which means '**Terminal Equipment'**. Terminal equipment is e.g. an ISDN telephone, an ISDN card, a fax machine, or any other terminal to places calls, dial into the internet, or send and receive faxes. If the card runs in **NT-Mode**, it is capable of connecting terminal equipment to it. The **NT** is the small box where all ISDN terminals are connected, which means '**Network Termination'**. It is also known as 'NTBA' (Germany) or 'NT1' (everywhere else).

What must be done, to connect telephones to an ISDN card? Don't be shocked when you read the list. It is much easier, as you will find out later in the text. Here is a list of things to do:

- Of course, the card must support NT-Mode hardware layer. (all HFC chips do.)
- A driver capable of NT-mode must be installed. (part of mISDN)
- The ISDN telephone must be connected cross-over to the card.
- Power must be supplied to the ISDN bus, in case the telephone(s) has/have no own power supply.
- The ISDN bus must be terminated with resistors of 100 Ohm. (better 50 Ohm)

In order to bring an HFC card into NT-mode, the following module parameter is used:

```
$ modprobe hfcpci protocol=0x12 layermask=3
```

But this will be explained later…

It is **not** possible to use just a cross-over cable for Ethernet. ISDN uses different wires in the cable than Ethernet and Fast Ethernet. Additionally, terminals and cards have no termination, because termination is normally done by NT1 or ISDN wall plug. ISDN cross connection is done by connecting the inner twisted wires (pin 4 and 5) with the twisted wires around them (pin 3 and 6). Pins 1, 2, 7 and 8 are not used. Some special PBX telephones can use these pins for extra power supply, but I don't know any telephone that supports it. Termination is essential, even if the ISDN cable would have almost zero length. All this can be done by using an old (even broken) NT1 (NTBA). Telephone companies throw them away when they are broken or customers have problems with it. Most times the power supply still works (about 9 out of 10). If you measure about 40 volts between pin 3 and 4, as well as pin 5 and 6, you know that your NT1 is good enough for using it as a power supply. Be sure that the terminator switches inside are turned on. (This is the factory default.)

An NT1 (NTBA) has three types of connectors. The first connects to the underground wire that is connected to the telephone network.[7] The second connects to the ISDN telephone(s). The third is connected to the power line. Follow the steps to get an NT1 connected to an ISDN card instead of the telephone network:

The VERY SIMPLE way:

Take an ISDN or Ethernet cable and cut it in the middle. Reconnect the inner pins (4 and 5) of one end with the pins around them (3 and 6) of the other end. Do it vice versa. (Connect 3 with 4, 4 with 3, 5 with 6, 6 with 5) Now you have an ISDN cross over cable, that is different to an Ethernet cable. Just connect the ISDN card with the cross over to one plug of your NT1, and a telephone with a normal cable (not crossed) to the other plug. Connect the power line of your NT1 or telephone and **you are done**. You will be able to connect only one telephone, because both plugs are used. Don't use the cross-over cable to directly connect your card to your telephone, unless your card or your cable has termination and your telephone has own power supply.

The COMFORTABLE way:

Step 1: Open the small door to get access to the wire clips. If it has no door, just take a screw driver and open the case. Be aware of high voltage at the switched power supply, even if it is not connected to the power line. Capacitors may hold high voltage for a long time. Inside are 4 clips to connect an ISDN cable directly to it, rather than connecting it to one of the RJ45 plugs. Each NT on the picture has six yellow clips: One pair (to the left) are used to connect the U-Bus (underground wire), the two pairs (to the right) connect to the S-Bus (telephone). The S-Bus is directly connected in parallel to the two rj45 plugs.

[7] On the other end of the phone line, the telephone company uses something called LT (Line Termination). It works almost like the NT1, but it is connected together with many other LTs to one or more 2-mbit-interfaces. This is called the 'Access Network'. This network is connected to the public exchange.

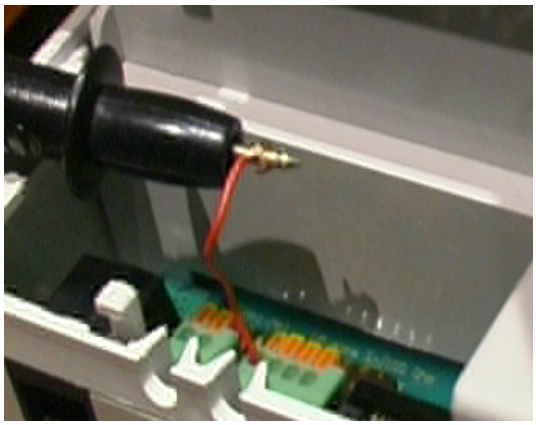
(Figure: NT1 with door open)

Step 2: Find out which clip inside the NT1 is connected to which pin on the rj45 plug. Therefore connect an ISDN cable to one of the two ISDN jacks. Take an Ohm-Meter and find out which clip is connected to which pin of the ISDN cable. Use a small peace of wire, to help the meter's probe get into the hole of the clip. Pin 4 and 5 as well as 3 and 6 of the ISDN cable, is connected through the coil inside the NT1. The coil has low resistance, so your Ohm-Meter will show that they are connected (almost 0 Ohms). So you cannot determine between pin 4 and 5 as well as between pin 3 and 6. There are two clips connected to pin 4 and 5, and another two connected to 3 and 6. It doesn't matter which clip is connected to pin 4 and which is connected to pin 5, since the **polarity doesn't matter**. (pin 3 and 6 respectively)

Most NT1 in Europe name the 4 pins: A1 and B1, A2 and B2[8]. Do the following connections for crossing send and receive pair:

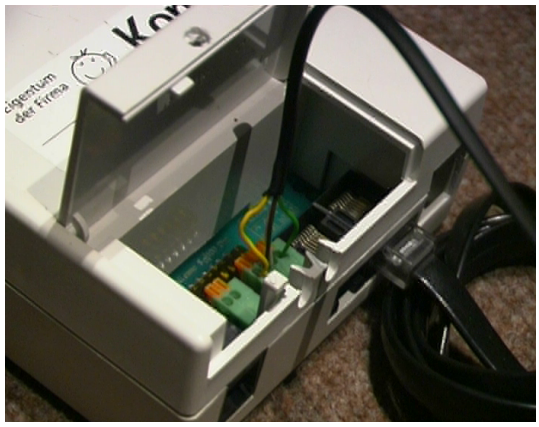
fA1 -> pin 3
AB1 -> pin 6
BA2 -> pin 4
AB2 -> pin 5

[8] German Telekom (former German state-owned telephone company) counted their twisted pairs in their cables. A1 mens, the wire A of twisted pair 1. The white or the cable with less black rings is wire A, the other is wire B.

(Figure: Find clips)

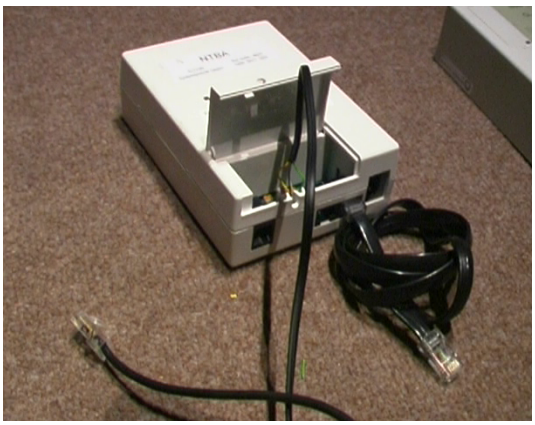
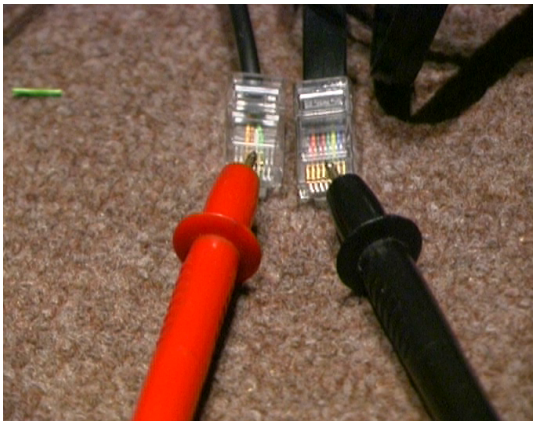
Step 3: Take another ISDN cable or Ethernet cable and cut off one end. Take the inner pair of the cable (pin 4 and 5) and connect them to the clips, that are connected to the outer pins of the ISDN jacks (pin 3 and 6). Do this with the inner pins of the cable respectively. Again, polarity of a cable pair doesn't matter for the ISDN bus.

9

(Figure: cable connected to the clips)

Step 4: Use the Ohm-Meter again and verify the connection between the cable, that is connected to the clips, and the cable connected to one of the jacks. Pin 4 and 5 should now connect to pin 3 and 6 and vice versa.

(Figure: Verify cross connection)

Step 5: Connect the cable, which is connected to the clips, to the ISDN card, that should run in NT-Mode. Be sure that termination is switched on. Inside the NT1 are two switches, that must be ON. This is the default.

(Figure: cable connected to the ISDN card)

Step 6: Connect an ISDN telephone to the ISDN cable, that is connected to the ISDN jack. Also connect the power cable.

(Figure: complete setup with one phone)

Since there is nothing connected to the U-Bus (underground wire), the internal electronics is disabled[9], because it gets its power from the U-Bus (not from the power line). Only the power supply is connected to the coils (transformer) inside the NT1 providing power for telephones, as well as termination. This is why the NT1 can be broken. Only power supply must work.

NOTE: If you like to connect more than one telephone, you might experience clicks and other audio problems. Use 50 ohm instead of 100 ohm termination. This can be done by adding

[9] Some new NT1 will also run their electronics with the integrated power supply, so no power feeding is required. (I found this on Sphairon's NT1 + Split boxes.) In this case, I would suggest to disconnect the internal coil from the NT1 controller. If you are not into electronics like that, just use an old NT1, it will work.

another 100 ohm resistors in parallel with the ones in the NT1. Two 100 ohm resistors in parallel become 50 ohm! One resistor must be connected between pin 3 and 6, and the other between 4 and 5. You must figure out yourself how to connect the cable plus the resistors. Maybe you use a soldering iron or stick both into the holes of the clips.

Alternatively you may get an ISDN termination plug. Use the plug between one jack of the NT1 and the isdn phone or between NT1 and ISDN card. It will also lower the ISDN termination down to 50 ohm.

2.3 Interrupt problems

One serious problem of installing HFC cards are interrupts. Some PCI slots share the same IRQ. Be sure that every HFC card has its individual IRQ. When booting the PC, the IRQ is shown of each PCI card installed. If two cards share the same IRQ the HFC-PCI driver may fail. I experienced this as I installed three cards in one PC. This problem may be obsolete due to Kernel improvements.

Another problem might occur with PC main boards using IDE controllers. The IDE controller blocks CPU during transfer. Since IDE drives require time to calibrate and find sectors, they can block IDE bus even multiple seconds. This causes loss of frames coming from the ISDN cards. This results in gaps of the audio during hard drive operation. I used an ISA Fritz-Card, which causes losses of interrupts during IDE hard driver access. The HFC-PCI card worked better. This problem doesn't occur with my PCI Adaptec SCSI controller in the same machine. With serial ATA, this problem should be no matter.

The audio processing of mISDN and LCR is designed to handle these problems. This should not lead to an increasing delay.

2.4 Hard disk

The size of a hard drive doesn't matter, as long as no audio recording is done. Since the LCR has a call recording feature (and answering machine), smaller hard drives get filled quickly during long calls. A recoding with 8 bit mono causes a recorded file of 8 kilobytes per second or 8000 bytes. The best quality of recoding is 16 bit stereo. This causes 32 kilobytes to be written down during one second. If the call lasts one hour, LCR would record a file of 115.2 megabytes or 115200000 bytes. Even with 8 bits mono (or law codec) is used, the recording would have still a size of 28.8 megabytes or 28800000 bytes. If the LCR would support the LPC-10 codec to record audio, the size would be 1.44 megabytes for recording one hour. A year of audio data could be compressed down to about 12.6 gigabytes. In this case, don't worry about size of your hard drive anymore. Any volunteers are welcome to implement the LPC-10 codec… Note that the call recording is just a feature that is not turned on by default.

There are no requirements for hard drive's speed, since the load will be 800-1600 kilobytes per seconds for 100 recording calls at a time. Hard drives are much faster these days.

2.5 Memory

Almost every Linux box uses swap memory. LCR disables swapping of it's code and data memory. Other programs still use swap. If the LCR allocates new memory, that is not

available, the memory of other processes must be written to disk first. This may cause a high delay of call processing, which also causes audio gaps and jitter. Also if other programs on the Linux box are swapped (like the telnet daemon), the LCR will stop for a short period of time, caused by hard disk traffic. If execution stops for several seconds, it may also cause ISDN protocol to fail. To reduce the risk, you may turn off swap memory when running the LCR. If LCR runs out of memory, it will terminate, because if a call that runs out of resources, LCR requires memory to process releasing of that call, but memory is not available anymore. Swap can be a good idea to prevent this, even if it causes gaps and jitter. Telephone calls are a real-time task, that don't like delays of any undefined time. Therefore increase physical memory, so the Linux box would run without swapping memory.

It depends on what's running on the Linux box. Just boot your Linux, and after starting LCR look how much of memory is left. Turn off swap, and check what's left:

```
$ swapoff -a

$ free
             total       used       free     shared    buffers     cached
Mem:         52376      50364       2012          0        868       6064
-/+ buffers/cache:      43432       8944
Swap:            0          0          0
```

In this example from my Linux box, there is about 2 megabytes of free memory. Note that the cached memory of about 6 megabytes will be available any time. The cached memory holds recent data from hard drive, that will be accessed faster when accessed again. In this case there are 8944 kilobytes (almost 9 megabytes) of memory still available without swapping. After a while, the machine without swap may run out of memory. I experienced this sometimes, because I had many other processes running, like web server, PPPoE and proxy.

Conclusion: It is much better to leave swap turned on, have much memory, and no other heavy programs running (except for normal server tools, like Samba, FTP, telnet and SSH daemons).

2.6 CPU

I have no experience with CPU usage. Making an external call, the CPU usage (AMD K6-200) is between 0.5% and 3.5%. If anyone has more experiences, please let me know. I recommend at least 200 megahertz for the LCR without using asterisk.

2.6 PCM

Almost all ISDN controllers support PCM bus to interconnect them. The HFC-4S / HFC-8S / HFC-E1 support a PCM bus of 128 timeslots with two banks. Also these controllers have an integrated timeslot assigner. This is useful to directly connect two controllers without a switching matrix[10]. It is possible to connect up to 128 callers on one bus. This makes large PBXs possible, where 128 calls can be made without CPU load for audio transfer. The driver automatically detects these cards and does hardware cross-connections and conferencing.

[10] A switching matrix is used when interfaces are statically connected to timeslots on a PCM bus. Since all channels of all controllers are connected to an individual timeslot, a switch controller is required to forward data from one timeslot to another. If controllers have timeslot assigners, they can connect to any timeslot they want, no switching is required.

Also PCM is possible with the HFC-PCI A chip. A special connector must be added to the board in order to interconnect them. I tried this and it worked, but I found it too complicated. If only a few ISDN ports are realized with HFC-PCI cards, software bridging is quite enough.

3

Download & Installation

3.1. Install mISDN driver

Why **mISDN** driver and not HiSax driver:

- The first ISDN driver of the Linux Kernel (HiSax) is not able to handle the ISDN protocol for connecting telephones to it (**NT-mode**). mISDN provides a user space library which supports NT-mode stack. The library "i4lnet" is part of the "mISDNuser" package, but not included in the kernel source. (yet)
- The ISDN4Linux is not able to link certain ISDN layers with user space programs. mISDN provides direct access to any layer. Each protocol layer is implemented in modules and can be loaded as required. Access to ISDN layer 3[11] is essential for running a PBX.
- Also b-channels have individual stacks with layers, that can be accessed from user space.
- Linux is not a real-time operating system. A PBX application in user space will have a delay or interruptions, when cross-connecting audio streams. If, for example, a call is made from an internal port to an external ISDN line, the b-channels must be cross-connected. To keep the delay as low as possible, a module called **mISDN_dsp** is included in the mISDN kernel source. It was programmed for the PBX4Linux but is also used in other applications, like a standallone Asterisk channel driver or this LCR. It will not only cross-connect, it is also capable of making conferences with an unlimited number of b-channels (if available), dejittering of audio source, volume control for incoming and outgoing audio data, real time tone generation, DTMF decoding, echo cancellation, even real time encryption, and it will do it in hardware if supported.

In order to run LCR, the kernel with **mISDN** is require, as well as the "**mISDNuser**" package. Download the latest sources from 'www.mISDN.org', the following packages are required:

- mISDN_xxxx.tar.gz
- mISDNuser_xxxx.tar.gz

NOTE: In earlier versions of PBX4Linux, I suggested not to use the CVS version. This was because the official version of mISDN did not work with the PBX4Linux code. One reason

[11] Layer 3 is the protocol layer of ISDN. This layer controls call setup, call modification (e.g. call forwarding) and call release.

was, that the API changed during development. Now you may download the latest release from mISDN.org. Also the latest master branch (default branch) should work.

mISDN can be installed to Kernel source or compiled directly. This is usefull if the Kernel has no mISDN driver or if newer driver with newer features are required. When compiling directly, just untar the mISDN package, make and make install:

```
$ tar xvzf mISDN_xxxx.tar.gz

$ cd mISDN

$ make

$ make install
```

The modules will be installed in "/lib/modules/<kernel version>/extra/". In this case, be sure that no mISDN is in the Kernel tree, because the kernel modules have higher priority than extra installed ones.

If you have a Kernel with mISDN: Here is a short instruction on how select mISDN vial Kernel config. If you like to use a newer Kernel, be sure to save the Kernel config of your current system:

```
$ cd /usr/src/linux

$ cp .config ../current_config
```

Unpack the source of your new Kernel. Do this only if you don't already have a source installed:

```
$ cd /usr/src/

$ tar -xvzf linux-x.x.xx.tar.gz

$ ln -s linux-xx.xx.xx linux

$ cd linux
```

Be sure to restore you saved Kernel config and copy it back to ".config":

```
$ cp ../current_config .config
```

Then configure your Kernel:

```
$ cd /usr/src/linux

$ make menuconfig
```

Enable the following kernel features, using "**`make menuconfig`**":

- **➔Enter Device drivers --->**
- **➔Enter ISDN subsystem --->**
- **• ISDN Support (say 'm')**
- **• CAPI2.0 support (say 'm')**
- **➔Modular ISDN driver --->**
- **• Support modular ISDN driver (MUST say 'm')**

- **Support for HFC PCI cards (say 'y')**
- **<other cards…> (say 'y')**
- **Support for digital audio processor (say 'y')**

A card, that supports NT-Mode, you may want to use. Otherwise you will not be able to connect telephones. If your card is an HFC-S PCI, say yes to the following:

- **HFC-S PCI cards**

Also say yes to any card you may want to use for external line.

Save and exit Kernel configuration, and continue to build the Kernel source:

```
$ make

$ make install

$ make modules

$ make modules_install
```

If everything compiles without errors, **Reboot**! Warnings about unused variables and wrong prototypes can be ignored of course. After reboot, run "**depmod**", in order to create dependencies, so the ISDN modules will be found. In newer kernels the "depmod" is done automatically when installing the modules.

To be able to use the "/dev/mISDN" device, it must be created:

```
$ mknod /dev/mISDN c 46 0
```

It is a character device with major number 46 and minor 0. Don't worry if you forgot, LCR will tell you when the device is missing. But before the device exists, the modules must be loaded. Read on, on how to create an rc-script[12] to load and unload the modules.

3.2. mISDNuser

In order to access mISDN device driver and use the NT-mode, install **mISDNuser** package:

```
$ tar -xvzf mISDNuser-x.xx.tar.gz

$ cd mISDNuser

$ make

$ make install
```

Two libraries will be created:

- **libmISDN.so**
- **libisdnnet.so**

[12] RC Scripts are Unix scripts, that are called during boot and shutdown, to start and stop services and software.

Also some other utils will be part of the package. Just ignore them. Be sure to uncompress mISDNuser the same source directory where you will uncompress LCR. The paths to the libraries are relative in the Makefile of LCR. If you choose different locations, you must edit the Makefile.

The **libmISDN.so** provides direct access to the mISDN device. It will be used to communicated directly with the protocol stacks of mISDN.

The **libisdnnet.so** provides NT-mode protocol. The TE-mode is part of the Kernel modules.

3.3. Linux-Call-Router

Download LCR at 'www.linux-call-router.de'. Uncompress it:

```
$ tar -xvzf lcr_xxxx.tar.gz

$ cd lcr
```

If you have trouble with OpenSSL includes, edit the "Makefile". At the line "WITH-CRYPTO", write "#" in front of it to comment it out. You will only need it, to support some kind of propriety encryption and handshaking

Compile and install the PBX:

```
$ make

$ make install
```

Configuration files, documents and data files will be installed by default at: "/usr/local/lcr". Edit the "Makefile" in order to choose a different location. Don't worry about your old configuration files, you might already have from older version. They will only be copied, if they don't exist already. If they don't exist, default configuration files will be installed. The binaries are installed at "/usr/local/bin" by default.

To see a list of start options of LCR, just enter:

```
$ lcr
```

To run PBX4Linux as background task, enter:

```
$ pbx fork
```

A daemon fork will be done, so closing the shell will not interrupt the PBX. Also use this for startup, if the PBX should be started at boot time.

Note: In order to run PBX4Linux, the mISDN driver must be loaded. Follow the next chapter on how to configure the driver, and creating a start and stop script.

3.4 Asterisk channel interface

Refer to later section about channel interface. If you intend to use LCR with Asterisk, you must install Asterisk first. Edit the Makefile. At the line "WITH-CRYPTO", remove “#” in front of it to enable, or disable by adding “#” in front.

4

Configuration

4.1. Start and stop mISDN

The following example will assume that your have a HFC-PCI card and a Fritz-Card installed. The Fritz-Card is connected to a normal multipoint ISDN line. Multipoint lines are used in most places. It is possible to connect up to eight telephones to it; the LCR may be connected parallel with other telephones. The HFC-PCI card is connected to a telephone, as described above. With this setup, LCR will be able to forward calls between external lines and internal telephones.

Many things must be done to setup ISDN4Linux. Therefore I wrote a small tool, that creates a shell scrip to start and stop kernel's mISDN driver. It comes with the LCR package and will be installed with it. Run it with:

```
$ genrc

This program generates a script, which is used to start/stop/restart mISDN
driver. Please select card only once. Mode and options are given by LCR.

Select driver for cards:

 (1) HFC PCI (Cologne Chip)
 (2) HFC-4S / HFC-8S / HFC-E1 (Cologne Chip)
 (3) HFC-S USB (Cologne Chip)

Select driver number[1-n] (or enter 'done'): 1
```

Enter '1' to select HFC PCI. If your card is not in that list, there is no driver for it or 'genrc' may be out of date.

Select driver number[1-n] (or enter 'done'): **done**

There is no other card type to use, so we use the keyword 'done'. Now we are asked to enter dsp-module's options and debugging values. Just enter '0' for every value.

```
Enter options of mISDN_dsp module. For a-LAW, just enter 0.
For u-LAW enter 1.
```

```
[0..n | 0xn]: 0

Enter debugging flags mISDN core. For no debug, just enter 0.
[0..n | 0xn]: 0

Enter debugging flags of cards. For no debug, just enter 0.
[0..n | 0xn]: 0

Enter dsp debugging flags of driver. For no debug, just enter 0.
[0..n | 0xn]: 0
```

Debugging is explained later in this book. Just enter ‘0’ for the first time. If you experience crashes, read on, on how to use these values.

```
Enter location of the mISDN modules. Enter '0' for kernel's default
location. Enter '1' for binary distribution's location
'/usr/local/pbx/modules' or enter full path to the modules dir.
[0 | 1 | <path>]: 0
```

Depending on the location of the modules, give '0' or '1' here. '0' for default location (source distribution) or '1' for location of the binary distribution.

```
File 'mISDN' is written to the current directory.
$
```

Now the following shell script is written in the current directory with the name “mISDN”:

```
# rc script for mISDN driver

case "$1" in
        start|--start)
                modprobe --ignore-install mISDN_core debug=0x0
                modprobe --ignore-install mISDN_dsp debug=0x0 options=0x0
                modprobe --ignore-install hfcpci debug=0x0
                sleep 1
                ;;

        stop|--stop)
                rmmod hfcpci
                rmmod mISDN_dsp
                rmmod mISDN_core
                ;;

        restart|--restart)
                sh $0 stop
                sleep 2 # some phones will release tei when layer 1 is down
                sh $0 start
                ;;

        help|--help)
                echo "Usage: $0 {start|stop|restart|help}"
                exit 0
                ;;

        *)
                echo "Usage: $0 {start|stop|restart|help}"
                exit 2
                ;;

esac
```

To load ISDN4Linux, just enter:

```
$ sh mISDN start
```

To unload the modules just enter:

```
$ sh mISDN stop
```

I will now explain what the script does, if the start option is given:

`"modprobe --ignore-install mISDN_core debug=0x0"` will load the kernel mISDN driver core.

`"modprobe --ignore-install mISDN_dsp debug=0x0 options=0x0"` will load the real time audio processor.

`"modprobe --ignore-install hfcpci debug=0x0"` will load the card driver for all HFC cards.

`"sleep 1"` will wait some time until all cards are registered.

Look into the system log. You will see some information while loading the modules. Also errors will show up there. In order to see the current configuration, start use 'isdninfo':

```
$ isdninfo

Port  1 name='hfc-pci.1': TE/NT-mode BRI S/T (for phone lines & phones)
  - 2 B-channels
--------
Port  2 name='hfc-pci.2': TE/NT-mode BRI S/T (for phone lines & phones)
  - 2 B-channels
--------

$
```

Connect an ISDN telephone as described above, and pick up the phone. You will not get a dial tone from the kernel. In order to get a dial tone, the application LCR must be running. But before running LCR, you must configure interfaces and "extensions".

4.2 PBX lines

A PBX line, also known as point-to-point line, is only capable to connect one PBX (or one telephone in point-to-point mode) to it. Also it features direct dialing in of extensions[13]. The PBX, connected to this type of ISDN line, is able receive incoming calls, after the first digit behind the main number is dialed. It is controlled by the PBX, when the number is complete. It depends on the provider, how much digits can be dialed after the main number. The international convention of the maximum length of a phone number is 13 digits. If the county code has two digits (49 in Germany) and the area code has three digits (212 for the city Solingen), and the main number has six digits, then there are two more digits that will be routed to the PBX line. Of course the PBX can define by itself when the number is complete. Inside Germany, more digits are allowed. The 'German Telekom' for example forwards up to

[13] DDI is a feature for incoming calls to dial an extension after the main prefix. The PBX will check the number and signal, if it is complete.

17 digits. The company, I work for, forwards up to 24 digits after the area code. The PBX will signal, when the number is complete. This is supported by LCR.

Another advantage of a PBX line is the ability to have more than one ISDN line for the same main number or prefix. One BRI ISDN line can handle two calls only at a time. The number of simultaneous calls can be increased by heaving more than one line. LCR also supports this. There is no know limit, except for the number of cards, that fit into your Linux box.

Note that PBX lines might get fault monitored by the telephone company, because they are more expensive and more important than normal ISDN lines. In this case, talk to your telephone company to ignore alarms because every reboot, crash or ever restarting of the LCR or mISDN driver will cause alarms.

To tell mISDN to use a PBX line, you must set the card into point-to-point mode. This can be done in the “interface.conf” using “ptp” keyword.

This will be included in the “mISDN” shell script, if the option “ptp” is selected when running the “genrc” tool.

PBX lines may also be multipoint. You will be able to connect normal telephones to it. In case of LCR it makes no difference. If telephones would be connected parallel to the PBX line, dialing of extension digits will work until one phone on the bus is ringing.

Another type of PBX line is the PRI line or “primary rate multiplex” line. This type of line provides 30 B-channels in Europe and 22 B-channels in the USA. It is used for really big PBXs. It is also possible to group multiple PRI lines to one big PBX interface. Use “hfcmulti” driver to support the European PRI chipset “HFC-E1”.

4.3 CLIP No Screening

“CLIP No Screening” or “No Screening CLIP” is a special agreement with your telephone company. Normally caller Ids, that are sent on outgoing calls, are checked against the numbers that are assigned to your ISDN line. If you send a caller ID, for which you have a number assigned, the telephone company will forward it to the called user. If the caller ID you specify doesn’t match with any of your assigned numbers, the main number assigned to you, is used. So it is not possible to fake caller IDs, and pretend that you are your ex-girlfriends new boyfriend, or the boss of your neighbor. But sometimes you may want to pretend that you are the boss of your neighbor, especially when she calls your PBX, and it forwards the call to your mobile telephone using a second line. On the second line to your mobile, the caller ID of the caller of the first line (the boss) is used. When your mobile telephone rings, you will see that the boss calls and not just your default number of your telephone line.

Another way to use this, is when you connect two PBXs. Lets assume you make a VoIP call to another PBX, that transfers the call for you to the public telephone system. In this case you want to send the caller ID of the PBX, you are calling from, and not of the other PBX, that transfers the call.

There are lots of things to use it for. Some of them are not very nice, but most of them are quite funny. I have this feature on my PBX line, for the reasons described above. Ask your local telephone company for the agreement. Note, that this feature is part of the basic ISDN standard ITU-T Q.931, so it is conform to make this feature available to you. If your telephone company tells you, that this feature is not legal, tell them about the standard.

The caller ID, sent on a normal ISDN line, is normally the local telephone number of type "unknown". The telephone company screens the ID and adds the national and international prefix in front. When using "CLIP No Screening", the caller ID must be sent by the PBX including all prefixes. There are three common types of caller IDs that can be sent: Type "subscriber" will just send the caller ID without any prefix. The called party that receives this caller ID will just see that number, as you specified, everywhere in the world. Type "national" must be sent with area code and number. The called party's Telephone will automatically add the national long distance prefix in front, so it must be omitted. Type "international" must be sent with country code, area code and number. The called party's Telephone will automatically add the international access code in front, so it must be omitted. Because the PBX doesn't know what access code the called party must dial, it just sends the type "international", and the prefix is added by the called party's telephone.

It the connected ID is transmitted without checking, it is called "COLP No Screening".

When this feature is enabled, two caller IDs are transmitted by you telephone company: the user provided and the network provided caller ID. The is required for billing the right number.

Note: There are also special handling of caller IDs, so you caller ID will be forwarded as it would be network provided. In this case, be very carefull on what you send, because this number might get billed. But don't worry, normal citizens will not get this feature, but your telephone company might do a wrong configuration of your local exchange. You can test it, because user provided calls will show as 'user provided' in the trace. (The trace feature is explaned later.) Just call yourself and see what is coming in.

4.4 CLIR Ignore

The most wonderful feature that an ISDN line can ever have is "CLIR Ignore". CLIR stands for "Caller Line Identification Restriction". The caller is able to hide the caller ID. If a call with an restricted ID is received, the exchange will delete the number. Only the information, that the ID is restricted, is transferred. In Germany most telephone lines restrict caller IDs. It is regulated by the "Bundesnetzagentur" formerly known as "RegTP" formerly known as "BAPT" formerly known as... , that a call which is made anonymously, should not presented to the called party. But the law says, that the called party has a right to know who is calling. This is quite strange.

If "CLIR Ignore" is set on your ISDN line, the caller ID is transmitted even if the caller doesn't want it. You could see every caller's ID in your log file. You could always reply a call if you were absent. This feature is used by the police and emergency lines in order to call back, if the caller hangs up or is unable to tell her location. The number will be transmitted with the information, that it should not be displayed.

The LCR also supports suppression of the restricted caller IDs, if your have the "CLIR Ignore"-feature on your ISDN line. Of course LCR also has the "CLIR Ignore" feature, that is called "anon-ignore". It displays anonymous external calls (if the "CLIR Ignore" feature is provided on your ISDN line) as well as suppressed internal calls.

4.5 Previous through-connect

Maybe you heared of it as “early B3”. Whenever you make a call, you get tones and patterns from your external line: You hear the dial tone of the local exchange, the ringing of the remote exchange and sometimes announcements. All this happens before and after the called party answers. Also you don’t get charged for it.[14] But if you get called, the local exchange will not connect audio until you answer the call. The local exchange generates tones, because your telephone doesn’t.

It is possible to send own tones during ringing or any stage of call setup, instead of the standard tones of your local exchange. This is useful if you forward calls, so announcements also get forwarded. A caller may also hear your ‘music’ or whatever you send him. There are three things required to send own tones before you answer a call:

1. You must have a PBX line (point-to-point), because during incoming call only one endpoint (the PBX) can send tones. A PBX line only talks to one PBX and so an audio channel can be assigned before answering of a call.
2. You must have the special feature enabled on your local exchange.[15] You may ask your telephone company to enable it for your, but don’t expect them to do it in Germany. Tell me if you are successful.
3. You need to enable tones for the port at ‘interface.conf’. This will be explained later. This options will indicate that tones are available at any state of the call.

The procedure of sending own tones is supported by the PBX, and is specified in the ISDN standard Q.931 Annex K. Note that only the audio path to the caller may be connected through, not the path from the caller to the PBX, so this is no ticket to make free calls.

The feature is mainly used for call forwarding through the PBX. If a forwarded call gets answered, it will take a while until the connect message is received from the exchange and forwarded to the PBX. It the called party answers and the connect comes to late, the audio path would also be opened too late, without this feature. This can cause the first second of the answered call to be lost. In case of previous through-connect, the audio path is connected to the forwarded party prior answer, and so the caller will hear as soon as the party answers.

What do you think about an answering machine that makes you charge only after the beep? ☺ This nice feature as well as playing announcements free of charge are implemented in LCR

4.6 Leased lines

There are three types of ISDN basic access leased lines:

- 64Kb one B-channel
- 128Kb two B-channels
- 144Kb two B-channels + D-channel

Both ends of a leased line are terminated by an NT1 (NTBA). On each side, one ISDN card may be connected, both running in TE-Mode. Since leased lines are statically connected, no call setup must be done. The data can be just transferred between both ISDN cards.

[14] Some countries do charge for calls that are not completed.
[15] On Siemens exchanges (EWSD), the feature is called “PTRUCON” and is available since Version 15 of the APS (firmware). (Since Version 16 it is bug free.)

It is possible to connect a card, running in TE-Mode (hardware layer), to one end of the leased line, where the NT-Mode protocol (software layer) is used. On the other end, telephones can be connected. A 144Kb leased line is required, because the D-channel must also be transmitted. It is then possible to use the leased line to connect telephones far, far away from the PBX. I have never tested this setup with a real leased line, but it should work. Any card can be used of course, since the hardware layer of this card is running in TE-Mode. Please contact me if you require this setup.

An alternative to the leased line is the "layer-1-over-IP" (l1oip) driver for mISDN. It is a virtual cal with a virtual leased line over IP. It is not compatible with any other VoIP standard, but since it just tunnels ISDN, it also tunnels all features of ISDN. These virtual cards can be used for LCR. Also the virtual cards can be connected directly to a real ISDN card.

4.7 General options

When the LCR is started, it loads the file "**options.conf**" first. You may skip that section if you are new to LCR, because all defaults are fine for most setups. The default location of the options file is "/usr/local/lcr/options.conf", if the default installation path is used. The file contains many of keywords and comments. Comments start with "#". Here is a description for every keyword used in this file:

➢**debug [<flags>]**

Use this keyword to specify the debug flags. This can be:

- 0 = debugging off
- >=1 = debugging on

The flags can be given decimal or hexadecimal using the "0x" prefix. Possible flags are defined in "main.h" header file. Also they are shown in options.conf. The definitions begin with "DEBUG_*". If debugging is turned on, a file called "/usr/local/lcr/debug.log" is created and an filled with the debug output. It will also contain trace as brief output.

Note that debugging is for a programmer, not a user feature. To see what's going, use traces. Tracing is explained below.

Default: Debugging is tuned off by default.

Example: "`debug 0x0004`" does mISDN stack debugging output.

➢**schedule [<priority>]**

A PBX transfers signaling information as well as audio data. For the audio data it is essential that LCR responds as fast as possible. Therefore the Linux OS provides a scheduling algorithm to provide real time processing. "SCHED_RR" (round robin) is used which allows one task with the highest priority to be processed until it waits for an action. Be sure to have the highest priority, so no other program will use CPU if LCR needs to. The priority must be between 0 and 99. If no priority is given, the scheduler is not changed, real time processing is disabled. This is useful for debugging, because software faults, causing endless loops would cause locking of all processes, the shells, the telnet daemon, even the console.

Default: By default the scheduler has priority 0.

Example: "`schedule 99`" set PBX4Linux process in SCHED_RR with priority 99.

➢**alaw | ulaw**

Specify the coding of samples, which are used in your country. It affects the internal processing of LCR. Note that all tones and announcements are encoded "alaw". If ulaw is used, mISDN must also be set into ulaw mode. Also tones and announcements must be converted or provided as wave audio files.

Default: By default "alaw" is used.

➢**tones_dir <directory>**

Tones and announcements are used to tell the telephone user when to dial, when the call is ringing, why did this call disconnects… All tones are a-law encoded samples. There are two default sets of tones, located at "/usr/local/lcr", if the LCR is installed in the default location:

- tones_american
- tones_german

Both are located at "/usr/local/lcr/," if the default installation directory was used. This directory can be overridden for each extension's settings file.

Default: American tones are used: "tones_american"

Example: "`tones_dir tones_german`" uses German tones.

➢**fetch_tones <directory>[,<directory>[, …]]**

Tones and announcements can be loaded into memory, rather than streamed when they are played. This will prevent jitter, caused by hard disk access. Only the given directories will be loaded. Other tones will still be streamed, such as recordings of the answering machine. Give all tone sets here, separated by a comma. **Don't** put spaces between the directory names. Note that this needs about as much memory as all tones together. If tones are wave files, which are not 16 bit mono, they are converted, so total memory usage may change. The total memory usage will be printed when starting LCR.

Default: No tones are loaded, so they are directly streamed from hard disk.

Example: "`fetch_tones tones_german,tones_english`" will cause the German and English tone sets to be fetched at startup time. Other tone sets like answering machine announcements will still be streamed from hard disk.

➢**extensions_dir <directory>**

Each extension has its own directory. Settings, for example, are stored inside the individual extension's directory. All extensions are stored inside the default installation directory "/usr/local/lcr". Note that each extension is a sub directory that may have files and directories inside.

Default: "extensions" The full path would be (in case of the default installation directory): "/usr/local/lcr/extensions/".

➢**national [<prefix>]**

Depending on your local exchange, long distance calls within your country, may require a prefix. Germany and most countries in the world require "0" to call outside local area. Denmark, for example, doesn't have a national prefix. In this case this prefix must be omitted. This is essential, to convert from national numbers without prefix to numbers with prefix and vice versa. Caller IDs, for example, are always transferred without prefixes, but with a type indicator, indicating the type of number. To convert a caller ID to a human dial able number, LCR must know, what prefixes national type numbers have.

Default: "0"

Examples: "`national 1`" specifies the prefix to dial long distance within the USA. "`national`" uses no prefix that must be used in Denmark.

➢**international [<prefix>]**

Depending on your local telephone system, international calls require a prefix, the "international access code". Germany and many countries in the world require "00" to call international. This is essential to convert from international numbers without access code to numbers with access code and vice versa. Caller IDs, for example, are always given without access code, but with a type indicator, indicating the type of number. To convert a caller ID to a human dial able number, LCR must know, what access code international type numbers have.

Default: "00"

Examples: "`international 011`" specifies the access code to dial international from the USA.

➢**te_if [interface1,interface2,…]**
➢**nt_if [interface1,interface2,…]**
➢**port <number>[ptp]**

These options are obsolete. Configuration of ports and interfaces are done in "interface.conf".

➢**nodtmf**

The mISDN driver features DTMF detection, in conjunction with the mISDN_dsp.o module. DTMF tones are used to control LCR during a connected call. Each ISDN call will use DTMF detection, if this keyword is not given. DTMF detection needs CPU cycles to decode DTMF tones. The 'Goertzel' algorithm is used, which is fast. By default, DTMF detection is enabled.

Example: "`nodtmf`" will disable the DTMF feature.

Default: DTMF detection is enabled.

➢**dummyid [<id>]**

If external calls are forwarded via LCR to external lines, the caller ID may not be available due to restriction of the caller. In this case, no ID is used, causing not to present any ID to the forwarded party. If your telephone line has the "CLIP No Screening" feature, you may specify a number that will override your default telephone number of your ISDN line.

Anyway, unavailable numbers are send suppressed, so even if you set an ID here, it will not be presented to the remote party.

Default: No caller ID is sent if not available. Your telephone company will use your default number instead. (In Deutschland wird die Kopfnummer verwendet.)

Example: "0" will send a '0' on forwarded calls, whenever the caller ID is not available.

➢**inbandpattern [yes|no]**

This option is obsolete, since tones can be configured for each individual interface in "interface.conf".

➢**dsptones [none|amrican|german|oldgerman]**

Tones/announcements are streamed from user space. It is possible to use the module "dsp.o" instead. This module is required for LCR anyway. It provides simple tones with much less CPU usage, because it is streamed in kernel space. Tones are much simpler and announcements are not supported. This make LCR much faster when only used for ISDN telephony. If supported by special hardware, tones are loops that require no bus/CPU load at all, except when the tone changes. It can be overridden by extension's tone set.

Default: Tones are provided from user space. The sample sets are defined via keyword "tones_dir".

Note: The configuration is loaded and parsed during startup. Whenever this file changes, LCR must be restarted.

➢**email [email]**

Source email address of the LCR. E.g. it is used when sending a mail from the voice box. It is not the address the mails are sent to. Most mail servers require an existing domain in order to accept mails.

Note: The configuration is loaded and parsed during startup ONLY. Whenever this file changes, the PBX must be restarted for the changes to take effect.

4.8 Interface configuration

The "**interface.conf**" file is loaded first after starting LCR. Also it can be reloaded during runtime, so changing interfaces does not require to terminate active calls. By default it is locate at "/usr/local/lcr/interface.conf". It will configure all **ports** of **mISDN** driver to use with LCR.

Sometimes interfaces shall have multiple links (ports) to increase number of channels. A **multi link** interface is an interface that uses multiple ports. All incoming calls are treated as they would come from one 'interface'. Outgoing calls are distributed on any free channel of any port of the 'interface'. This is useful to expand external and even internal interfaces to more than two channels (BRI) or 30 channels (PRI). It also offers redundancy if one or more links fail.

For internal interfaces any extension may be allowed. It is detected by caller ID (MSN number on the phone). Most times only a **given set of allowed extensions** shall be allowed on one interface. For each interface it is possible to define a default MSN number, as well as additional MSN numbers that are accepted on this interface. If the MSN number of a phone is not found in the list of allowed extensions, the first (default) extension is used.

The interface's configuration is placed at "**/usr/local/lcr/interface.conf**" by default. Each interface is defined in multiple line, starting with the interface's name:

```
#This is a comment.
[telco]
port 2

[office]
extension
msn 202,203,204,205
port 5
```

This is about everything you need to set up a PBX. The first interface will get the name "telco". Port 2 is assigned to this interface. This interface is used for external calls, since the keyword "extension" is not used. The second interface will get the name "office". Port 5 is assigned to this interface. This interface is an internal interface, because the "extension" keyword is used. Also this extension is limited to four MSN numbers. If calls with different MSN number are incoming, the first MSN is used.[16]

[16] This process is called „screening". The caller ID of the phone is checked against the list of MSN numbers. If the caller ID is not found, the first MSN is used. The caller ID of the phone is then replaced by the extension's caller ID. The local exchange will also do "screening" and adds area code and country code, if required. If "no screening " is done by the local exchange, the extension's caller ID must include area code and country code.

To get a brief list of all keywords, use "**`lcr interface`**". Here is a list of keywords that can be used to specify an interface:

➢**extension**

Specified whether this interface is just treated as an external interface, or if it has extensions. This keyword has only effect on incoming calls. If an outgoing call is made, the extension's configuration file will specify the interface to use.

Example: "`extension`"

Default: The interface does not have extensions.

➢**msn [<default MSN>[,<another MSN>[, …]]]**

In case of extensions, only the given MSN are allowed. The caller ID of a phone is checked against the list of MSN. If the caller ID is not found, the first MSN is used. This makes only sense on internal ports, that are connected to phones.

The list of MSN can also be defined on external port. In this case, only caller IDs that are found in the list, are accepted, others are replaced by the first MSN. There is no general use for it.

Example: "`MSN 5,6`" only allows extension 5 and 6. If other caller ID is received from a phone, the extension 5 will be used.

Default: The interface accepts any caller ID as MSN number.

➢**is_tone yes | no**

Some interfaces send tones and announcements, and others don't. Your local exchange, for example, will send tones, as you pick up the receiver. But your telephone will not do any tone or announcement, if it is busy. If a phone is busy, the tone is generated by the local exchange and transferred to the remote user.

LCR will be able to send tones and announcements on internal, as well as on external ports. Sending tones on external ports make only sense, if local exchange will not do it, or if a connected PBX doesn't provide own tones. If tones are not available, they are generated by LCR.

Also signaling of the tones are done, by sending progress indicator 8 "Inband tones". If a phone or exchange receives progress indicator 8, it will open the B-channel, to receive the tones. If local exchange has the "previous through connect"-feature enabled, it may also allow tones from LCR.

By default, tones are transmitted on NT-mode ports only, since telephones are not expected to generate tones and announcements to the user.

Example: "`is_tone yes`" will send tones and announcements to all ports of the interface.

Default: Tones are only transmitted on NT-mode ports.

➢**is_earlyb yes | no**

As interfaces send tones and announcements, the interfaces connected to them will connect and receive these tones. Imagine you are connected to the local exchange, that sends tones during call setup or after disconnect, you may want to get these tones. In this case you may need to say "yes" to this option. If you connect a phone, that normally doesn't send any tones, you may say "no" and LCR will make own tones.

By default, tones are received on TE-mode ports only, since telephones are not expected to generate tones and announcements to the PBX.

Example: "`is_earyb yes`" will receive tones and announcements from all ports of the interface.

Default: Tones are only transmitted on NT-mode ports.

➢**hunt linear | roundrobin**

An interface may have one or more ports. If multiple ports are given, the interface has more channels available. When an outgoing call is made on an interface, a channel is required. By default, the first available channel on the first free port is used. If all channels of one port are busy, the next port is searched, until free channel is found. If all channels are free, the first port is always used for outgoing call.

To balance load to all ports of an interface, the "roundrobin" value may be used. The channel depends on port of the last call. If a call was made on port 1, the next call is made on port 2 and so on. A port is skipped, if no channel is available. If the last port was used or is busy, the search begins again on the first port.

This feature is quite useful on point-to-multipoint ports. If one of two ports fail, every second call will fail and the next will go through. In case of liner hunting, if port 1 fails, no outgoing call possible, because always port 1 is used first. In case of point-to-point ports, if one port fails, it will be ignored when hunting a free channel.

Example: "`hunt roundrobin`" will cycle to the next free port on every outgoing call.

Default: The free port is searched beginning with the first port on every outgoing call.

➢**screen_in [options] <old caller ID>[%] [options] <new caller ID>[%]**
➢**screen_out [options] <old caller ID>[%] [options] <new caller ID>[%]**

As you have read above, the "msn" parameter is used to check and change caller ID for incoming calls for an internal phone. If the caller ID sent by the phone doesn't exist in the list of MSN numbers, the first MSN is used.

For more complex change of caller IDs, "screen_in" is used. This parameter can be given multiple times. When an incoming call is received (from external line or internal port), the caller ID is checked against the list of "screen_in" parameters. If it is found, it is replaced by

the “new caller ID”. If the rule matches, the search stops. If no rule matches, no change is performed.

For outgoing calls (to external line or internal port), the caller ID can be changed by using “screen_out”. The same operation is performed, as with “screen_in”.

Sometimes it is useful to compare a prefix of a caller ID. Add ‘%’ to the end of the caller ID to compare only the part in front of ‘%’. When the caller ID matches and is replaced by the new caller ID, the suffix may also be added to the “new caller ID” by adding ‘%’ to the new caller ID.

Caller IDs may have different types or may be suppressed, presented or not available. If options are given in front of the “old caller ID”, they must also match in order to match the rule. If options are given in front of the “new caller ID”, the caller ID’s type and presentation is also replaced, if the rule matches.

- **unknown** will define a caller ID of type “unknown”.
- **subscriber** “subscriber” type
- **national** “national” type
- **international** “international” type
- **present** caller ID is presented
- **restrict** caller ID is restricted

Changing caller IDs can be useful to convert any type of caller ID to the number that are assigned to the local exchange. In case of “no screening CLIP” feature, screening is required by LCR. Local caller IDs must be extended with national area code.

Example: “`screen_out national 21250993 unknown 50993`**”** Will change outgoing caller IDs that exactly match national caller ID 21250994. The caller ID will be replaced by type “unknown” with ID 50993.

Example: “`screen_out unknown 5% national 2125%`**”** Will change outgoing caller IDs that begin with “5” to national caller ID with area code 212.

Default: By default, both screen lists are empty, so no change is performed.

➢**filter <chain>**

Filters are used to add one or more filters to audio processing. This can be single volume change, echo cancellation or even encryption.

To be done…

➢**port <number>**

Each interface consists of one ore more ports. The given port number is added and bound to the interface. If more than one port parameter is given, the interface is a multilink interface. Multiple links are used to increase number of B-channels, as well as adding redundancy to the interface. Multiple links are generally used to increase number of possible calls from or to the local exchange.

Ports can be PRI or BRI interfaces. Also ports can run in TE-mode (to local exchange) or NT-mode (to phones or other PBX).

The port number specifies the mISDN port number, starting with 1. Multiple ports can be specified by giving multiple port parameters below the interface name.

Example: "`port 5`" Will bind port 5 to the current interface.

There is no default for that. If no port is specified, the interface doesn't have any port. No incoming or outgoing call is possible in this case.

- **ptp**
- **ptmp**

This parameter is followed by a "port" parameter. It alters port defaults.

"ptp" forces the port to be a point-to-point interface. It overrides the default mode given by mISDN. For BRI NT-mode interfaces, the default is "ptmp", so it can be altered. For PRI NT-mode interfaces, the default is "ptp".

"ptp" is useful, if port connects to another PBX.

By default, LCR will always set PRI ports into point-to-point and BRI into port-to-multipoint.

Example: "`ptp`" Will force to run the port in point-to-multipoint.

- **nodtmf**

This parameter is followed by a "port" parameter. It alters port defaults.

Disables DTMF detection. For most external lines, DTMF is required for voice menus or dialing extensions after answering the call. For performance reason, DTMF can be turned off for the port.

In my opinion, machines are fast enough today to handle DTMF, since the filter quite fast. It may be used on embedded solutions.

Example: "nodtmf" Will disabled DTMF detection.

Default: DTMF detection is turned on by default.

- **channel_out [force,][<number>][,...][,free][,any][,no]**
- **channel_in [<number>][,free]**

This parameter is followed by a "port" parameter. It alters port defaults.

Channel assignment is simple by default: The user (PBX) asks for "any channel", and the network (local exchange) will answer with a free channel, if available. It the network initiates

a call to the user, the channel number is forced. Channel assignment is done by the network only in this case.

In case we are the user (PBX), we request “any channel” for outgoing calls and accept any free channel for incoming calls. We would have the following defaults:

```
channel_out any
channel_in free
```

In case we are the network (PBX connected to a phone or another PBX), we force a free channel for outgoing calls (calls to the user) and accept any free channel (from the user):

```
channel_out force,free
channel_in free
```

In case of a multipoint link to a phone, we also support call waiting. If no channels is available, no channel is used until the phone releases or parks another:

```
channel_out force,free,no
```

Sometimes, it can be quite useful to alter the channel assignment. In case of “**directed**” channels, a primary rate interface (PRI) may have some channels for incoming calls only, some for outgoing calls only, and some for both way. An example is a hotline. Hotlines may be busy due to high amount of calls. By default, all channels would be used, so no call agent could make outgoing calls, not even in an emergency case.

The local exchange may use 20 channels incoming calls only. 8 channels may be used for incoming calls and outgoing (both way) calls for the call agents. The other two are outgoing only and can be used in an emergency. The LCR will use the following parameter:

```
channel_out force,22,23,24,25,26,27,28,29,30,31
```

This forces to local exchange to use channels 22-29 first, if not free, then it forces channels 30 and 31. For incoming calls we accept all channels, but reject channel 30 and 31.

```
channel_in 1,2,3,4,5,6,7,8,9,10,11,12,13,14,15,17,18,19,20,21,
    22,23,24,25,26,27,28,29
```

All values given in the list are checked in order of appearance. If the first given channel is not free, the next channel is checked. If no channel is found at all, the call is rejected.

`force` – This keyword is a special value. It will force all channels given, by signaling "channel as given, no alternative accepted". If not given, the remote may select a different channel than we selected. This must be the very first value in the list. This keyword is only allowed for outgoing channel selection.

`1…15, 17...31` – These numbers give the channels to select in the given order.

`free` – Instead of typing in a list of all channels of the link, the LCR selects a free channel from 1…31 (PRI) or 1..2 (BRI). The lower channel is selected first.

`any` – This doesn't select any channel, it will signal "any channel accepted". The remote must select the channel in this case. This keyword is only allowed for outgoing channel selection.

`no` – If this keyword is reached, the LCR signals "no channel available". This is only allowed on NT-mode point-to-multipoint links. If a phone is busy and no channel is available, it will get a call with no channel. After clearing a previous, it will receive a channel. This is called "call waiting". This keyword is only allowed for outgoing channel selection.

Default: The defaults are different and explained above.

4.9 Routing

Introduction

Whenever an internal telephone is picked up without a number dialed, a dial tone is given, inviting to dial a number. (This is the normal case, that may be modified.) The configuration file "**/usr/local/lcr/routing.conf**" will specify, what **action** will be performed under what **conditions**. One condition can be the dialed number. Also an action may have **parameters** to change the default behavior. This is quite similar to an access list like "iptables"[17]. There are three elements:

tconditions	(In which case the action shall be performed)
(action	(What action shall be performed if all conditions are true.)
(parameters	(Will alter the default behavior of the action.)

All three elements form a "**rule**". The "action" is mandatory, the conditions and parameters are optional. Example:

```
dialing=5 time=900-1700 : intern extension=15
```

This rule will match if the dialed digits start with "5" AND the time is between 9 AM and 5 PM. If **all** of the given conditions are true, the action "intern" is performed. The internal extension is not the dialed digits "5", but it is "15", so we use the parameter "extension" to override the default behavior.

[17] iptables is used by the Linux Kernel's firewall to filter IP traffic.

Multiple rules for a **rule set**. Here is an example of a simple rule set:

```
[main]
extern                      : intern extension=15
dialing=0                   : extern
dialing=15                  : intern
```

Each call will start with the rule set "**main**". The first rule that matches will perform the action. If the call is an external call, the first rule will match, because all conditions are true. If the call is not an external call, the first rule will not match, so the conditions below are checked. Depending on the dialed digit "0" or "15" the second or third rule will match. If nothing is dialed, the rules will not match until something is dialed. Also if something else, beginning with "9" is dialed, the rules will not match and nothing happens.

Sometimes it is usefull to get disconnected if digits are dialed that would never match. In this case a special condition "**default**" can be used. A rule with the condition "default" matches if the rules above would not match, if more digits are dialed:

```
[main]
extern                      : intern extension=15
dialing=0                   : extern
dialing=15                  : intern
default                     : disconnect cause=1 display="Unbekannte Nummer"
```

If something else than "0" or "15" is dialed, the rule above would not match, even if more digits are dialed, so the fourth rule will match. The action "disconnect" is performed. The cause for disconnect is "1" (Unallocated number) and the message on the telephone's display is "Unbekannte Nummer".

Syntax description

"routing.conf" is an ASCII file with **rule sets**. A rule set starts with the following syntax in a separate line:

```
[peter]
```

Between braces is the name of the rule set, in this case the name is "peter". The rule set ends with end of file or if another rule set is defined.

Each rule set has an undefined number of **rules**:

```
<condition>=<value> [<moreconditions>=<value>] : action [parameter=value …]
```

Each rule is written in a separate line. All given **conditions** must match in order to process the **action**. If no condition is given, the rule always matches. Some conditions have no value, like "extern" or "anonymous". Most conditions have string or number values, in this case "=" is used to separate condition and value. No spaces are allowed. Multiple values may be given using "," to separate values. Also ranges for values are allowed: "-" is used in this case. If a value includes spaces, it must be quoted. Multiple conditions are separated with white spaces. Here are some examples for conditions:

```
anonymous
callerid="021250993"
callerid=0212,0202,0231
```

```
dialing=1,40-79
```

If multiple values or ranges are given, only on value must match, in order to match the condition. If multiple conditions are given, all conditions must match. The first condition given above matches if the caller is anonymous. The second matches if the caller ID is "021250993" or begins with it. The third condition matches if the caller id begins with "0212" or "0202" or "0231" or is equal to one of them. The fourth condition is true if at lease "1" or something between "40" and "79" is dialed.

Strings may be quoted:

S"012345"-"012399"
""Quote \"inside\" string"
"Spaces\ without\ using\ quotes

The same rules apply to the **parameters**. In this case it makes no sense to define multiple values. Some parameters expect a string of a comma separated list of values. The action is separated from the conditions using ":":

```
[main]
extern                      : goto ruleset="extern"
intern                      : goto ruleset="intern"
                            : disconnect
```

In this case the call checked to be an external call in the first rule. If it matches, the rule set pointer jumps to the rule set "extern". If the call is a call from an internal extension, the second rule matches. If the call is neither external or internal ISDN, it will match in the third rule, because no condition is given. The caller will be disconnected.

To add a comment, use "#" to start. Quote strings to use "#" as value for a string. Follow this example:

```
dialing=*                   : goto ruleset="service" # all service codes
dialing="#"                 : goto ruleset="service" # disable services
```

Available '*conditions*':

To get a list of available conditions, use the following start option:

```
$ lcr rules
```

The output shows all available conditions first. Also check the default configuration "/usr/local/lcr/interface.conf" for example. All conditions are explained in detail:

➢**extern**

This condition is true if call is from an external interface. It is defined by "interfaces.conf" whether the port is connected external or internal (extension). Actually the call is extern, if the caller is not associated with an internal extension.

➢**intern**

This condition is true if call is from internal interface (extension). It is defined by "interfaces.conf" whether the port is connected external or internal. Actually the call is intern, if the caller is associated with an internal extension.

➢**port=<number>[-<number>][,…]**

This condition is true if call is received from given port(s), not interface. A port number is defined by mISDN and equals the number of mISDN stack. It is the same port that is defined at "interface.conf". Changing mISDN configuration, may cause port numbers to change, so it is not wise to use this condition. Use "interface" condition in stead. It will check the name that is associated with the port.

➢**interface=<interface>[,…]**

This condition is true if call is received from given interface(s), not port. An interface name is defined at "interface.conf".

➢**callerid=<digits>[-<digits>][,...]**

This condition is true if the caller ID is equal to one of the given string of digits, or begins with one of the given string of digits. Also ranges of strings may be given. This condition is used mainly for external calls

➢**extension=<digits>[-<digits>][,...]**

This condition is true if caller calls from given (range(s) of) extension(s). This condition only applies to internal callers. Each internal caller is associated to an 'extension'. Extensions are explained later. In most cases, the extension number equals the internal caller ID. Also a range of extensions may be given.

➢**dialing=<digits>[-<digits>][,...]**

Dialing is the most important condition. It will let the caller define what to do. This condition will be true if one of the given digits or range of digits have been dialed. Also it is true if more digits have been dialed, but if they begin with the digits given. Also a range of digits may be given. It should be used in conjunction with "intern" or "extern" condition.

➢**enblock**

Whenever a call matches after call setup (incoming call without dialing afterwards), this condition is true. If the condition is checked after dialing additional digits after setup, the condition is not true.

➢**overlap**

This is the opposite of “enblock”. The condition is true only if additional digits are dialed after setup.

➢**anonymous**

If a caller ID is sent restricted, this condition is true.

➢**visible**

If a caller ID is sent unrestricted, this condition is true.

➢**unkown**

If no caller ID is available, this condition is true. Sometimes it is possible that a caller ID is available even if it is restricted. (Restricted calls from internal phone may have a caller ID, so this condition is not true.)

➢**available**

If a caller ID is available, this condition is true, even if the caller ID might be restricted.

➢**fake**

If the caller ID has been given by user and not by network, this condition is true. A special agreement with the phone company allows to send user generated caller IDs that may be of someone else or even don’t exist. In this case the screening indicator show that this caller ID may not really be of the caller. Also a second caller ID is sent, but LCR always uses the first (unreal) caller ID.

➢**real**

This is the opposite of “fake”. This condition matches if the caller ID was not unscreened but network provided (approved).

➢**redirected**

This condition is true if the call has been redirected. (forwarded)

➢**direct**

This condition is true if the call has not been redirected.

➢**redirid=<digits>[-<digits>][,...]**

This condition is true if the call has been redirected by the given number or the first digits of the redirecting number matches. It can be used filter calls that have been accidentally redirected to LCR.

➢**time=<minutes>[-<minutes>][,...]**

This condition is true if the given time in minutes is the current local time. The format must be “hmm”, where 9 o’clock AM equals “900” and 8 o’clock PM equals “2000”. Also ranges may be given. It is also possible to make negative ranges like “2000-900” (from 8 o’clock PM until 9 o’clock AM).

➢**mday=<day>[-<day>][,...]**

This condition is true if the given day is the current day of the month. Also ranges and negative ranges are possible.

➢**month=<month>[-<month>][,...]**

This condition is true if the given month is the current month. Also ranges and negative ranges are possible.

➢**year=<year>[-<year>][,...]**

This condition is true if the given year is the current year. Also ranges are possible. The year must be between 1970 and 2106.

➢**wday=<day>[-<day>][,...]**

This condition is true if the given weekday is the current day of week. Also ranges and negative ranges are possible. Note that Monday equals “1” and Sunday “7”.

➢**service=speech|audio|video|digital-restricted|digital-unrestricted|**
digital-unrestricted-tones[,...]

This condition is true if the call has the given type of service. Note that an ordinary telephone call can be speech or sometimes audio.

➢**infolayer1=<value>[,...]**

This condition is true if the call has the given layer 1 info. The value 2 indicates voice coded aLaw, a value of 3 indicates uLaw.

- **hlc=<value>[,…]**

This condition is true if the call has the given high layer capability. Use it only if you know what you are doing.

- **file=<path>[,...]**

This condition is true if the give file exists and if the first character of that file is '1'. Please note that accessing this file may stall the PBX, so please don't use network file systems or slow devices like CD-Roms or floppy disks.

- **execute=<command>[,...]**

This condition matches if the given command or shell script returns a value greater than 0. Please note that LCR will halt for the time the script is running, so please use scripts that will take not more than a 100th of a second. Audio may get interrupted and delayed.

- **default**

This is a special condition that makes a rule only match if all rules above will never match, even is more digits are dialed. This can be used for a fallback if no number given in any rule above would match.

- **timeout=<seconds>**

This is a special condition that will set a timeout if the rule would match. If no digits are dialed for the given number of seconds, the rule matches.

- **free=<interface>:<channel>**

This condition is true if the given more or equal the given number of channels are available and free.

- **notfree=<interface>:<channel>**

This condition is true if at least the given number of channels are available and free.

- **blocked=<interface>[,…]**

This condition is true if one of the given interfaces are blocked. (not available for some reason)

- **idle=<interface>[,…]**

This condition is true if one of the given interfaces are idle.

➢**busy=<extension>[,…]**

This condition is true if one of the given extensions is busy.

➢**notbusy=<extension>[,…]**

This condition is true if one of the given extension is not busy.

➢**remote=<application name>**

This condition is true if the given remote application is running. One application can be "asterisk" channel driver. This is useful to route calls in case the application is running or not.

➢**notremote=<application name>**

This condition is true if the remote application is not running.

Available '*actions*'

If all given conditions of a rule are true, the given action will be executed. To get a list of available actions, use the following start option:

```
$ pbx rules
```

➢**intern** [connect] [extension=<digits>[,…]] [capability=speech|audio|video|digital-restricted| digital-unrestricted|digital-unrestricted-tones] [capability=transparent|hdlc] [infolayer1=<value>] [hlc=<value>] [exthlc=<value>] [present=yes|no] [type=unknown| subscriber|national|international] [timeout=<seconds>]

This action will forward the call to an internal extension. It most cases it is desirable to give an extension number, rather than using the dialed digits. Especially for mapping external numbers to different internal extensions, it is required to give 'extension' parameter. If multiple extensions are given, the call is forwarded to multiple extensions simultaneously.

➢**extern|outdial** [connect] [prefix=<digits>] [capability=speech|audio|video|digital-restricted| digital-unrestricted|digital-unrestricted-tones] [capability=transparent|hdlc] [infolayer1=<value>] [hlc=<value>] [exthlc=<value>] [present=yes|no] [interfaces=<interface>[,<interface>[,...]]] [type=unknown|subscriber|national|international] [complete] [callerid=<digits>] [calleridtype=[unknown|subscriber|national|international]] [timeout=<seconds>]

This action will forward the call to an external interface. The dialed external number matches the digits that are dialed after the rule matches. Let's assume the rule defines a condition "dialing=0" to match this rule, then the dialed external number will be every digit dialed after "0". The dialed number can be overridden by "prefix" parameter. Any interface that is not an

extension, will be used. To specify one or more individual interfaces, use the "interfaces" parameter.

Example: "`dialing=0 : extern prefix=01013`" This rule matches if at least 0 is dialed. The caller will be forwarded to an external available channel. The prefix is put in front of whatever dialed behind "0". If the caller dials "0021250993", the external number will be "01013021250993".

➢**remote** [connect] [timeout=<seconds>] <application>

This action will forward the call to the given remote application. This application must run in order to complete the call.

Example: "`dialing=4 : remote application=asterisk`" This rule will match if "4" was dialed at the beginning. If "4021250993" is dialed, the call will be forwarded to the Asterisk channel driver. Asterisk will receive "021250993" as dialed number.

➢**vbox-record** [connect] [extension=<digits>] [announcement=<file prefix>] [timeout=<seconds>]

Calls can be directly transferred to the answering machine of the given extension. Instead of forwarding the calls with an extension's forward option, this action may be used. The caller can directly call the answering machine. It is essential to give "extension" option in order to make this work. The announcement can be replaced by a different audio file. This allows different announcements for different conditions.

Example: "`dialing=1234 : vbox-record extension=201`" If a caller dials 1234 (let's assume it is an external caller), the voice box of extension answers.

➢**partyline** [room=<digits>]

This is a special feature to make large conferences by just calling a number. A "room" must be given by digits. If multiple party lines shall exist, multiple rules with different rooms may be specified. Any call to a party line gets connected. The first member will receive hold music. If more than one active member is in the party line, they will hear each other. If one member suspends or holds the call, the audio is muted, so the party line is not disturbed by hold music from that member.

Example: "`dialing=1234 : partyline room=42`" If a caller dials 1234, he or she enters the party line number 42.

➢**login** [connect] [extension=<digits>] [nopassword]

The login feature is used to log into a (different) extension. Especially external callers may log into the given extensions from external phones. The caller will be asked to enter a password, if the extension is already specified. If the extension is not specified, the caller must dial the extension's number, and then dial the password. The password is specified within the extension's setting file. By default, no password is used, so no password will work. The number of digits, to be dialed for the password, is defined by the length of the extension's

password. If the length has been reached, the call is disconnected, if the password is wrong. If the password is correct, the dial tone invites to dial a number. If the "nopassword" parameter is given, no password must be entered. Be sure that the conditions will only allow selected callers to match that rule. Note: For **external calls**, the "connect" parameter is required in order to connect the audio path and allow DTMF dialing. All dialing is now done using DTMF or keypad information, if available.

Example: "`dialing=98765 : login connect extension=200`". If the caller dials 98765, she will get connected and asked to dial the password of extension 200. After entering the correct password, a dial tone indicates the correctness. Otherwise the caller gets disconnected.

➢**callerid** [connect] [present=yes|no] [callerid=<digits>] [calleridtype=[unknown|subscriber| national|international]]

This action will change the extension's caller ID. After the rule matches, the caller must dial the new caller ID. After the caller ID, the caller must finish with "#". The caller ID's type is defined by the digits entered. If the national prefix is dialed in front of the caller ID, the ID is of type 'national', and sent without the national prefix of course. The same rule applies to 'international' and 'subscriber' caller ID type. To make current caller ID anonymous, the caller just dials "#" right after the code. The allow the presentation again, a new caller ID must be specified. If the caller ID is of type 'unknown', the new caller ID will always be of type 'unknown'. This is the normal case, if not "No Screening Clip" feature is available to the external line. It is also possible to give the 'callerid' parameter instead of dialing it. This is useful, if a fixed caller ID shall be selected by that rule.

Example: "`91 callerid`" will specify the prefix "91" to be dialed to change the caller ID. If the caller wants to change her caller ID to "1234", she must dial "911234#".

➢**calleridnext** [connect] [present=yes|no] [callerid=<digits>] [calleridtype=[unknown| subscriber|national|international]]

This is similar to "callerid" except for specifying a caller ID, which is only used for the next call. After the next call is made, the regular caller ID is used again. Again, the dialing must be completed with "#" at the end of the caller ID, if no 'callerid' parameter is given. To make the caller ID for the next call anonymous, the caller must dial "#" without a caller ID.

➢**forward** [connect] [diversion=cfu|cfnr|cfb|cfp] [dest=<string>] [delay=<seconds>]

This action changes the current call forwarding for the caller's extension. After the rule matches, the caller must dial the new destination and finish with "#". If no destination is given, but finished with "#", the call forwarding is deactivated.

To forward a call to the voice box, "`dest=vbox`" must be given.

It is essential to give the diversion type:

"`diversion=cfu`" means: "Call Forwarding Unconditional". Every call to the given extension is directly forwarded; the internal phones on the specified ports will not ring.

“`diversion=cfb`” means: “Call Forwarding when Busy”. If the extension is currently calling, every call to the given extension is directly forwarded; the internal phones on the specified ports will not ring. This will be performed if at least one caller uses the extension.

“`diversion=cfnr`” means: “Call Forwarding when No Response”. Every call to the given extension is forwarded after some time of no response; the internal phones on the specified ports will stop ringing.

“`diversion=cfp`” means: “Call Forwarding Parallel”. Every call to the given extension will be forwarded to the given number. The internal phone and the forwarded destination will ring, until someone answered on an internal phone, or on the forwarded destination.

➢**redial** [connect] [select]

This action just redials the last call. It doesn't redo the last action that was performed, just the last call. If the 'select' parameter is given, the history of all stored calls can be selected. In order to do this, the caller must be internal so the display feature is available. After the rule matches, the last call is displayed. By dialing '#' via keypad or DTMF, the next older call is show. To scroll back, dial '*'. Alternatively '1' and '3' can be used to scroll through the history. To call the selected number, dial '0'. This feature only works for internal callers.

➢**reply** [connect] [select]

This action equals to the 'redial' action, but it refers to the incoming calls. In the list of incoming calls, all answered and unansered calls are stored.

➢**powerdial** [connect] [delay=<seconds>] [limit=<retries>] [timeout=<seconds>]

This action redials the last number again and again, until the call will be connected. To prevent timeout of the telephone, use 'connect' parameter. The delay between calls can be selected. If no delay is given, the delay must be dialed and finished with '#'. On every retry. a beep will indicate that the call failed and that it is redialed. The delay may also be given here, so no delay must be entered by the caller. Also the number of retries may be limited, as well as the total time to try.

Warning: This feature may cause high signalling load the PBX as well as telephone equipment connected to it. (local exchange) Also it may cause blocking of telephone line and even failure. Your phone company may be very angry ☹.

Example: “`dialing=67 : powerdial`” will cause dialing “67” to redial the last number using power dial feature. The caller must dial “67#”, in order to start power dialing with a almost no retry delay. If the caller dials “6760#”, the call will be redialed every minute until it connects successfully.

➢**callback** [proceeding] [alerting] [connect] [extension=<digits>] [delay=<seconds>] [callto=<digits>]

This action will do a callback. It can be used to start call back. The specified extension will perform a callback if the rule matches. The number to call back is specified by the caller ID of

the caller. Alternatively a different call back destination can be given. This makes it possible to trigger a call back from a different phone than the one that is called back. Callback is only done to external destinations. Read “Callback” section for detailed information before using this and authentication. The 'delay' parameter may be used to delay the call back.

If the call has been triggered via external PBX line, all further dialing is stored until the caller hangs up. After calling back, the stored digits are dialed. Entering the digits after callback is not required then. In most cases, only a few digits can be dialed behind the external number prefix, because the total length of a suffix number is limited by the local exchange. I use abbreviated dialing to handle the four digits that I can dial behind my external prefix.

If the caller ID is not known to LCR, the “callto” parameter must be given.

If the caller ID is received for the first time, the password must be entered, as it must be done when using the login feature. Note that this done after calling back. The caller ID is stored in the extension's directory for allowing further callbacks without authentication.

I also would suggest to use the “real” condition, so calls with fake caller ID will not be allowed to do a callback.

Example: “`dialing=8765 real : callback 200`” will perform a callback from extension 200, if the external number (MSN) “8765” is dialed and the caller does not have the “No Screening Clip” feature.

➢**abbrev** [connect]

Every extension may dial abbreviations, which are specified in the extension’s phonebook: “/usr/local/lcr/extensions/<extension>/phonebook”. A phonebook entry is a single line, which looks like this:

<abbreviation> <dialing> [<name>]

The abbreviation may be any string of digits. If the abbreviation is dialed, the “dialing” is performed. “dialing” may be any string of digits that may be dialed from the extension. Beware of redialing the rule itself, to prevent recursions.

Abbreviations can be useful, especially to dial digits, that cannot be dialed from a telephone.

Example: “`dialing=7 : abbrev`” will cause dialing “7”, to dial an abbreviation. Let's assume that the following line is stored in the phonebook:
“`123 05551212 Directory Assistance`”
The caller must dial “7123”. This will cause the dialing of “05551212” to be performed. In this example the “0” in front of “5551212” is used to make an external call.

Note: It is required to put external prefix in front of the number. The dialing is performed as it would have been done from the extension.

➢**test** [connect] [prefix=<digits>] [timeout=<seconds>]

This action runs the test mode. Read the “Test Mode” section for suffixes and their functions. It is possible to specify the test by the ‘prefix’ parameter. If it is not specified, the caller must

dial the ‘test code’. On external lines, it requites the dial in feature, that is only available on PBX lines. If a complete ‘prefix’ is given, no extension dialing is required.

Example: “`dialing=99 : test`” will specify the prefix “99” to be dialed to dial test mode. If the caller would dial “995”, she would hear the hold music. “`99 test prefix=5`” would directly play the hold music, if “99” is dialed.

➢**play** [proceeding] [alerting] [connect] [sample=<file prefix>] [timeout=<seconds>]

This action plays the given audio file. The 'sample' parameter must be given. It may be encoded as wave (16bit mono, stereo or 8bit mono) or “law”. The type of “law” codec is specified in “options.conf”. The file may be specified with relative path or with absolute path. If the path is relative, the specified “tones_dir” (in “settings” or “options.conf”) is used. The suffix “.isdn” or “.wav” will be appended to the file, according to what exist. If the file doesn’t exist or if the playing has finished, the suffix “_loop.isdn” or “_loop.wav” will be played afterwards. If this also doesn’t exist, nothing is played.

Example: “dialing=`93 : play /root/alle_meine_entchen`” will play the file “/root/alle_meine_entchen.isdn” or “/root/alle_meine_entchen.wav”, if exist. If the file has been played or if it doesn’t exist, the file “/root/alle_meine_entchen_loop.isdn” or “/root/alle_meine_entchen_loop.wav” is played, if exists, until the caller hangs up or the action times out.

➢**vbox-play** [extension=<digits>]

This actions starts the interactive menu of the extension's answering machine. It is only possible for internal calls. (or calls from extern that logged in) A spoken and displayed menu will be available. Read “Using LCR” for more information on answering machine.

➢**calculator** [connect]

This action starts the built in calculator. This is just a small funny tool. This only works for internal ISDN phones, since they only get the required display information. The calculator is processed by dialing the formula via DTMF or dialed number. The optional “connect” parameter is used to send a connect message to the phone, since ISDN phones may stop dialing after a certain number of digits. In this case keypad or DTMF dialing is used.

The formula is only a simple term. It must have two values and one operand. The values may only be entered as positive float values. After dialing the code, the first value is dialed, then the operand and then the second value. To dial the operand, the ‘*’ key is used. If it is pressed once, the multiplication is used. If it is pressed again, the division is used. If it is pressed again, the addition is used. If it is pressed again, the subtraction is used. Depending on the first value, if it is pressed again, the decimal point “.” is added. If the second value is entered, the ‘*’ key will directly add the decimal point. After entering the formula, the ‘#’ is used to get the result. If the result is displayed, it may be used for another formula. If ‘#’ is pressed again, the formula is cleared. Here are examples of formulas and how to enter them:

Term:	Result:	Dial:
5*8	40	5*8#
5.5*8	44	5*****5*8#
5.5*8.8	48.4	5*****5*8*8#
4/8	0.5	4**8#
4+8	12	4***8#
4-8	-4	4****8#

The term is always displayed during dialing, not the keys that are actually entered.

➢**goto** [connect] [sample=<file prefix>] [ruleset=<name>] [strip]

This action jumps to given ruleset and optionally play sample. Dialed digits are not flushed. After entering the new ruleset, all dialed digits are still available. Alternatively a sample may be played to announce the ruleset. If the ruleset doesn't exist, the call gets disconnected. The existance of the ruleset is checked only at call time.

In order to remove the digits required for this rule to match, use “strip”. After entering the ruleset, the digits are removed, so further conditions (“dialing=xxx”) will only require the digits dialed after the digits that were required to match the rule.

Example: “dialing=`555 : goto ruleset=″servicemenu″ strip`” will cause to jump into the ruleset “servicemenu”, if 555 is dialed. Because “strip” parameter is given, the dialed digits “555” are removed from the dialed number when entering the new ruleset. If “555666” would have been dialed, the new ruleset is entered with the dialed number “666”.

➢**menu** [connect] [sample=<file prefix>] [ruleset=<name>]

Same as 'goto', but flushes all digits dialed so far. The new ruleset is entered with no digits, even if more digits are dialed than required to match this rule. This is very useful to make interactive voice menus. Here is an example for a simple interactive voice menu:

```
...
time=1700-0900 dialing=123456 : menu connect sample=/usr/local/pbx/agent/intro ruleset=agent
...

[agent]
dialing=1 : play sample=/usr/local/pbx/agent/officehours
dialing=2 : vbox-record extension=200
dialing=9 : extern prefix=021250993
timeout=20 : vbox-record extension=200
```

The „intro.wav“ file would have the following anconcement: *„Hello - Our office is closed. Tho hear our opening hours press 1, to leave a message press 2, and in case of an emergency press 9, or wait to leave a message.“*.

If the caller dials 1, the first rule of ruleset “agent” matches, the sample “officehours” is played. “.wav” or “.wave” is appended to the file name.

If the caller dials 2, the voice box of extension 200 is called.

If the caller dials 9, the external call is made, the prefix “021250993” is dialed.

If the caller waits 20 seconds (enough time to hear the announcement and dial a digit), the call is automatically forwarded to the voice box of extension 200.

➢**disconnect** [connect] [sample=<file prefix>] [cause=<cause value>] [location=<location value>] [display=<text>]

This action disconnects the call. Optionally the 'cause', the 'location', a sample and a display text may be given. The 'cause' is the ISDN specific code for disconnect cause. For example, cause=1 would be the code for an unallocated number, as cause=17 would be the code for a busy phone. The 'location ' is the ISDN specific code for the point that generates the cause. Because the causes are generated by LCR, it automatically selects the correct location, so don't give a different location, unless you know what you are doing.

Here is a list of the causes:

1	**Unallocated number**
2	**No route to transit network**
3	**No route to destination**
4	**<Listen to announcement...>**
5	**Misdialed trunk prefix.**
6	**Channel unacceptable**
8	**Preemption**
9	**Preemption - circuit reserved**
16	**Normal call clearing**
17	**User busy**
18	**No user responding**
19	**No answer from user**
20	**Subscriber absent**
21	**Call rejected**
22	**Number changed**
26	**Non-selected user clearing**
27	**Destination out of order**
28	**Invalid number (incomplete)**
29	**Facility rejected**
31	**Normal, unspecified**
34	**No circuit/channel available**
41	**Temporary failure**
42	**Switchting equipment congestion**
43	**Access informationen discarded**
44	**No requested circuit/channel**
46	**Precedence call blocked**
47	**Resource unavailable, unspecified**
49	**Quality of service not available**
50	**Requested facility not subscribed**
53	**Outgoing calls barred within CUG**
55	**Incoming calls barred within CUG**
57	**Bearer capability not authorized**
58	**Bearer capability not present**
63	**Service or option not available**
65	**Bearer capability not implement.**
66	**Channel type not implemented**
69	**Requested facility not implement.**
70	**restricted digital informat. only**
79	**Service or option not implemented**
81	**Invalid call reference value**
82	**Identified channel does not exist**
83	**No suspended call with this id**
84	**Call identity in use**
85	**No call suspended**
86	**Suspended call has been cleared**

87	**User not member of CUG**
88	**Incompatibel destination**
90	**Non-existent CUG**
91	**Invalid transit network selection**
95	**Invalid message, unspecified**
96	**Information element missing**
97	**Message type non-existent**
98	**Message not compatible with state**
99	**Information element not impl.**
100	**Invalid info element contents**
101	**Message not compatible with state**
102	**Recovery on timer expiry**
103	**Parameter non-existent**
111	**Protocol error, unspecified**
127	**Interworking, unspecified**

Here is a list of locations:

0	**User**
1	**Private (Local)**
2	**Public (Local)**
3	**Transit**
4	**Public (Remote)**
5	**Private (Remote)**
7	**International**
10	**Beyond Interworking**

For a detailed description read the ITU Q.850.

➢**execute** [connect] [execute=<full path>] [param=<string>]

Whenever this rule matches, the given command is executed after the caller hangs up. If digits are dialed after the number, the digits are included in the parameter.

The first parameter of the executed command will be the given optional param string. The next will be the dialed digits. If no optional param string is given, the first parameter will be the dialed digits. The next parameters will be the caller ID followed by the caller's extension, followed by the extension's name, followed by the port number.

Note: Use this only for commands that exit quickly, and do not cause high CPU load, because this will slow down LCR and cause delays. I have not tested it well. It will do a daemon fork and keep all current file descriptions open. Don't use it to start programs. Use it to run a tool, which immediately exits, like "echo", "reboot" or any shell scripts, which may turn on your coffee machine.

Example: "dialing=`92 : execute execute="echo $1 >> /tmp/log`" will execute the "echo" command, which will, in this case, append the dialed digits after the prefix "92" to the file "/tmp/log" after the caller hangs up.

Please don't use this, unless you know what you are doing.

➢**file** [connect] [file=<full path>] [content=<string>] [append]

This action writes the given content to given file. If content is not given, the dialed digits are written.

➢**pick** [extensions=<extension>[,<extension>[,...]]]

This action picks up a ringing call. By default any internally ringing call is picked up. Also calls that are already transfered to the answering machine (VBox) are picked. A list of extensions may be specified with the 'externsions' parameter. Ony given extensions may be picked.

Example: "`dialing=3 : pick`" will connect to ringing or knocking call, when the caller dials "3".

➢**nothing** [proceeding] [alerting] [connect] [timeout=<seconds>]

Does nothing. It is useful to wait for calls to be released completely, by giving timeout value. Very useful for hunting groups.

➢**efi** [proceeding] [alerting] [connect]

This is a special feature for announcing the caller ID of the caller. I use this for my company so technicians can check from what phone line they are calling.

Available '*parameters*'

Actions have parameters to give additional information as well as overriding default value. To get a list of available parameters for an action, use the following start option:

```
$ pbx rules <action>
```

Here is a list of all parameters that are available:

➢**proceeding**

Will set the call into 'proceeding' state to prevent dial timeout.The caller will not be able to dial additional digits.

➢**alerting**

Will set the call into 'altering' state. The caller will get the alerting signal that may cause an alerting sound. The caller will not be able to dial additional digits.

➢**connect**

Will complete the call before processing the action. Audio path for external calls will be established. DTMF dialing will be possible due to connected audio path.

➢**extension=<digits>**

Give extension name (digits) to relate this action to.

extensions=<extension>[,<extension>[,...]]

One or more extensions may be given. If multiple extensions are given, the call will ring on multiple extensions simultaneously.

➢**prefix=<digits>**

Add prefix in front of the dialed number. Also a complete number may be given.

➢**service=speech|audio|video|digital-restricted|digital-unrestricted|digital-unrestricted-tones**

Alter the service type of the call. This will override the service given by the caller. It may be usefull to announce data calls as voice calls, because some provider charge voice calls differntly.

➢**bmode=transparent|hdlc**
➢**infolayer1=<value>**
➢**hlc=<value>**
➢**exthlc=<value>**

Alter mode, layer 1 info and high layer capability of the outgoing call. Use it only, if you know what you are doing.

➢**present=yes|no**

Allow or restrict caller ID regardless what the caller wants.

➢**diversion=cfu|cfnr|cfb|cfp**

Set diversion type:

cfu call forward unconditional
cfnr call forward no response
cfb call forward busy
cfp call forward parallel

➢**dest=<string>**

Destination number to divert to. Use 'vbox' to divert to vbox. (cfu,cfnr,cfb only)

➢**select**

Lets the caller select the history of calls using keys '1'/'3' or '*'/'#'. This is only possible if the caller calls from internal, because display feature is required.

➢**delay=<seconds>**

Number of seconds to delay.

➢**limit=<retries>**

Number of maximum retries.

➢**host=<string>**

Name of remote VoIP host.

➢**port=<value>**

Alternate port to use if 'host' is given.

➢**interfaces=<interface>[,<interface>[,...]]**

Give one or a list of Interfaces to select a free channel from.

➢**address=<string>**

Complete VoIP address. ([user@]host[:port])

➢**sample=<file prefix>**

Filename of sample (current tone's dir) or full path to sample. ('.wav'/'.wave'/'.isdn' is added automatically.

➢**announcement=<file prefix>**

Filename of announcement (inside vbox recording dir) or full path to sample. ('.wav'/'.wave'/'.isdn' is added automatically.

➢**ruleset=<name>**

Ruleset to go to.

➢**cause=<cause value>**

Cause value when disconnecting. (21=reject 1=unassigned 63=service not available)

➢**location=<location value>**

Location of cause value when disconnecting. (0=user 1=private network sering local user)

➢**display=<text>**

Message to display on the caller's telephone. (internal only)

➢**ports=<port>[,<port>[,...]]**

ISDN port[s] to use.

➢**tpreset=<seconds>**

Preset of countdown timer.

➢**file=<full path>**

Full path to file name.

➢**content=<string>**

Content to write into file.

➢**append**

Will append to given file, rather than overwriting it.

➢**execute=<full path>**

Full path to script/command name. (Dialed digits are the argument 1.)

➢**param=<string>**

Optionally this parameter can be inserted as argument 1, others are shifted.

➢**type=unknown|subscriber|national|international**

Type of number to dial, default is 'unknown'.

- **complete**

Indicates complete number as given by prefix. Proceeding of long distance calls may be faster.

- **callerid=<digits>**

Given caller ID will be used instead of dialing it.

- **callto=<digits>**

Where to call back. By default the caller ID is used.

- **room=<digits>**

Conference room number, must be greater 0, as in real life.

- **timeout=<seconds>**

Timeout before continue with next action.

- **nopassword**

Set “no password required” for the rule.

- **strip**

Removes all digits dialed so far.

- **application=<name>**

Selects remote application to forward the call to.

If “routing.conf” has been altered, use “lcradmin route” to reload routing at runtime.

4.10 Extensions

An extension is defined by a number. It is less confusing, if this number equals to the number that must be dialed internally or externally. Extensions have all individual files and settings:

- **/usr/local/lcr/extensions/<extension>/settings**

Settings are used to describe features of the extension.

➢**/usr/local/lcr/extensions/<extension>/logs**

All calls from and to the extension will be logged with date, time, duration, source and destination.

➢**/usr/local/lcr/extensions/<extension>/phonebook**

The phonebook is used to store abbreviated numbers and their names. It is also possible to store any dialing function of LCR. (The only thing that cannot be stored, is the recall of an abbreviated number. This avoids recursions.)

➢**/usr/local/lcr/extensions/<extension>/callbackauth**

Everyone in the world can use the callback feature, as long as she sends her caller ID. After callback, the extension's password must be entered. If the password is correct, the caller ID is stored in the file "callbackauth". This is useful, to prevent unauthorized callers from using callback. Next time a callback is done with the same caller ID, the callback is done without authentification.

➢**/usr/local/lcr/extensions/<extension>/recordings**

Because we have a computer, and **not** just a dirty box hanging on the wall with CPU speed of Commodore 64 computers, we are able to record calls also. Hard drives became large these days. If call recording is enabled for an extension, all incoming and outgoing calls are stored as wave or "law" files. The filename consists of the date, time, source and destination of this call.

➢**/usr/local/lcr/extensions/<extension>/vbox**

The integrated answering machine stores all incoming calls in the "vbox" directory. Also the "announcement" file is store here. Additionally, there will be an "index" file of all currently stored messages.

Extensions can be easily created by using the "genextension" command:

```
$ genextension <extension> <interface> <caller ID>
```

The first parameter is the extension number, for example: "200". The second parameter specifies the extension's interface to be used for calls to the extension. "Int" would send a call to the internal interface as defined in "interface.conf". If calls should ring on more then one port, they must be separated by commas without spaces. "1,3,4" would send a call to the internal port 1, 3 and 4. In this case, at least four internal ports must be defined in "options.conf". The third parameter defines the caller ID to be used, whenever the extension makes a call (to external line). The type of caller ID is "undefined", that is the standard type for normal external ISDN lines. If the external ISDN line has the "CLIP No Screening" feature, it must be given with type "subscriber", "national" or "international". The "settings" file must then be altered with an editor. After executing the command the extension will be created at "/usr/local/lcr/extensions/200/".

Also you must make an entry in "routing.conf" at internal ruleset in order to receive calls:

```
dialing=200 : internal
```

or

```
dialing=300 : internal extension=200
```

For external calls, the extension may also be specified as the “default extension”:

```
dialing=0 : internal extension=200
```

If you use a normal (multipoint) telephone LINE with an MSN number “50993”, specify this in your external ruleset:

```
dialing=50993 : internal extension=200
```

Here is a description of all keywords defined in the “/usr/local/lcr/extensions/200/**settings**” file:

➢**name [<string>]**

Each extension can have a name. The name will be processed internally with the caller/called ID. To actually see the name on a telephone, the ‘display_name’ feature must be enabled. In this case the display shows the name with the number. Also the ‘cnip’ feature can be used. In this case it must be enabled for the other party’s extension and it must be supported by her telephone.

➢**prefix [<prefix>]**

This prefix will be dialed, whenever the extension picks up the phone. Instead of hearing a dial tone, the caller will have the given prefix already dialed. This can be useful, if for example, the caller should be able to directly dial an external number, without dialing the code for external dialing first. If the code is “0”, the prefix of “0” would cause directly call external after pickup. If the prefix would be “0<number>”, the number would be dialed just after pick up. This can be useful for emergency calls of handicapped persons or little children, if they are unable to dial a number.

Note: The given prefix will not override the dialing rights.

➢**cfu [<number> | vbox]**

Whenever this number is given, any incoming call to the extension will cause the call to be forwarded to the given number. The call forward is done through LCR, since there is no implementation of deflecting a call. The benefit of it is, that the party forwarded to, may control LCR, answer knocking calls, and even record. To forward to the answering machine, the keyword “vbox” must be used.

➢**cfb [<number> | vbox]**

This call forwarding is performed, only if the called extension is busy. That is, if a phone of the extension is picked up, is connected during a call, receives announcements during

disconnect, or even makes multiple calls. The person, using the extension, is currently doing something. That's why she is busy. To forward to the answering machine, the keyword "vbox" must be used.

- **cfnr [<number> | vbox]**
- **cfnr_delay [seconds]**

This call forwarding is made if the called person does not answer within a given time. If, by default, the call is not answered after 20 seconds, it is forwarded to the given number. The number of seconds can be changed. To forward to the answering machine, the keyword "vbox" must be used.

- **cfp [<number>]**

This call forwarding will not actually forward the call. Calls will not only ring on internal ports, they will also ring on an external line at the same time. It is possible to pick up the call at any ringing phone. If the called party picks up the phone, all other phones will stop ringing. The benefits are: No more running to the ringing phone, before the call gets forwarded. And no more callers, who will hang up before "the call is forwarded, the mobile rings, the mobile take out of the bag, because the caller doesn't wait that long, because she knows that you have an one-room-30m²-apartment".

- **change_forward (yes | no)**

Allows or disallows the user, to change the forwarding for her extension. This is useful, if the extension's user should not be able to change the forwarding.

- **callerid (none | [s | n | i]<caller ID> [anonymous])**

This will specify the caller ID for this extension. Whenever the extension makes a call or receives a call, the caller ID is used. The caller ID is of type "unknown". This is useful for any ISDN line without the "No Screening CLIP" feature. If the caller ID starts with a small "s", it will be of type "subscriber". If the caller ID starts with "n", it will be of type "national". If the caller ID starts with "i", it will be of type "international". Because the types are used, it is not allowed to put national or international prefix in front of the caller ID's number. For example: A caller ID for the number "1-212-555-1212", which is a national number, would be "n2125551212". The international number would be "i12125551212", because the "i" will indicate international type, the "1" will be the country code, and the rest will indicate area code and number.

Caller ID's may also be sent with the information that it should not be displayed to the called party. The called party will not receive the caller ID, unless she has the "CLIR ignore" feature on her line, that is used by the police, for example. In case of an emergency, the caller ID can be important, even if the call is anonymous.

If no caller ID is given, and if the caller ID should be sent anonymously (not to be displayed), just the keyword "none" is used. In this case, no caller ID is transmitted. The telephone company may use the default number instead.

➢**id_next_call [[s | n | i]<caller ID> [anonymous]]**

If this caller ID is given, it will be used for the call made by the extension. After the call, it will be erased from the configuration file.

➢**change_callerid (yes | no)**

Allows or disallows the extension's caller to change the caller ID. This is useful, if the extension should not be able to change her caller ID, not even for the next call.

➢**clip (asis | hide)**

Calls, that are forwarded, use the originating caller ID. An internal telephone will always show the real caller ID. If a caller makes a call to an extension that is forwarded to a mobile, the mobile will receive the caller ID that is given here:

- **asis** = The original ID of the caller, that is forwarded, will be used. (Only works with "CLIP No Screening" feature)
- **hide** = The extension's caller ID "callerid" will be used.

➢**colp (asis | hide | force)**

If an extension is called; and if the called party answers, the caller ID of the extension is presented to the caller, showing her, who she is connected with. On **forwarded** calls, the given "connected to ID" may be used:

- **asis** = The ID of the party "forwarded to", will be presented to the calling party, if available, showing here, that she is connected to a different number. (Only works with "COLP No Screening" feature)
- **Hide** = Always the extension's caller ID "callerid" will be presented to the calling party, even for forwarded calls. No one would notice, if the call has been forwarded.
- **Force** = The ID of the party "forwarded to" is used. If the party "forwarded to" will not present any "connected to ID", the dialed number of the party "forwarded to" will be used instead. (Only works with "COLP No Screening" feature)

➢**clip_prefix [<prefix>]**

When a caller ID is shown on an internal phone, it is show without the out-dial prefix. Most telephones have a feature to add the out-dial prefix in front, so unanswered calls can be dialed out of the list of caller IDs. But some phones don't have this feature. In this case the out-dial prefix can be specified here. This/These is/are the digit(s) that have to be dialed to get an external line.

If no prefix is given, the caller ID is forwarded as is. The type of caller ID is processed by the telephone. If a prefix is given, the caller ID is converted by LCR and sent as „unknown“ type to the telephone. In this case the prefix + access codes are prefixed, as it will be done by the telephone.

Example: A caller ID **+1-809-563-0000** of „international“ type „1809563000“ and a given prefix of „9“ and an international prefix of „00“ (see options.conf) will result in an „unknown“ caller ID on the phone: „90018095630000“

➢**centrex (yes | no)**

Note: Centex might not work in the current version.

“Called Name Identification Presentation” (CNIP) can be used if supported by the telephone or even by the external line. This feature is user -> network only, which means that it can not be transmitted to the external line, only received from it. This special feature must be supported by the telephone. Some telephones get confused by the CNIP feature, so their operation fail. It is also possible to use display information. CNIP might also store the name in the caller’s list of the telephone. Display information may disappear after a few seconds. CNIP can be presented with the ‘centrex’ feature when

- an internal call is made, and the caller has specified a name for the extension.
- a call is received/answered from external, and the external line also features CNIP / CONP.
- the name is found in the phonebook.

The ‘centrex’ option also features “COnnected Name identification Presentation” (CONP). The name of the party that answered the call can be presented.

➢**interfaces [<interface 1>[,<interface 2>[, …]]]**

This is one of the most important setting. If nothing is specified here, the extension cannot be called. Of course the call forwarding is still possible. The interface or interfaces that should be used must be specified here. Multiple interface must be separated by comma and **without spaces**. See the following examples:

- “interfaces”
  This causes no internal phone to ring, when the extension is called.
- “interfaces Int2”
  The interface “Int2” is used.
- “interfaces Int1,Int3”
  Interfaces “Int1” and “Int3” will ring on incoming calls simultaneously. Remember not to use spaces around the comma.

➢**rights (none | internal | local | national | international)**

Each extension has dialing rights. **Use only one of the keywords.** By default, **international** calls are allowed. If the rights are set **national**, any international call attempt will cause the call to be disconnected. If the rights are **local**, any national or international call attempt will be

forbidden. If the rights are **internal**, no external call is allowed at all. If the rights are none, not even internal calls are allowed. The reception of any call will not be restricted by this option.

Example: If you have children, they may not be allowed to make calls outside your local area. You would give local rights only to your kids.

Example: If you have a customer phone in the supermarket, you would give only internal rights to your customers.

Example: If the phone should only receive internal calls, try "none".

➢**delete_ext (yes | no)**

Delete-function when dialing external numbers. '*' will delete the last digit, '#' will delete the complete external number. After pressing '#' the dial tone will be heard again. Also enable 'display_dialing' to see on the display what is actually dialed.

➢**noknocking (yes | no)**

Knocking may be used to receive a call, even if the phone is busy. Knocking must be provided by the ISDN telephone. This is the normal case. If you say "yes" and if the extension is busy, any call to this extension is rejected as being busy. By default, the extension can receive multiple calls, even if busy.

➢**txvol <volume –8…0…8>**

This will adjust the volume of the extension by the power of 2. It only affects internal phones. If the volume is 0, it means that it is not changed. A value of '-1' would cause the audio to be transmitted from the extension to the remote party at half volume. A value of '1' would cause the audio to be transmitted at double volume. Values of '2' would cause the quadruple volume, and a value of '8' would cause 256 times the volume (very loud). Please note, that the sound wave may be clipped when the volume is increased due to limited range.

Increase this value, if the mouth peace of your telephone is too weak, or you speak with very low volume. Decrease this if, you have the telephone in a loud environment.

You may also use this feature to amplify the mouth peace, so it becomes a hand-free microphone.

➢**rxvol <volume>**

This will do the same as "txvol", but for the audio received by the extension. Increase this value, if the ear peace of your telephone is too weak, if you have an ear disease, or if you have the telephone in a loud environment.

➢**tout_setup (off | <seconds>)**
➢**tout_dialing (off | <seconds>)**
➢**tout_proceeding (off | <seconds>)**
➢**tout_alerting (off | <seconds>)**
➢**tout_disconnect (off | <seconds>)**

By default there are timeouts during a call. The timeouts can be turned off. All timeouts are only relevant for internal ISDN telephones. Some phones have own timeouts, that cannot be altered. Here is the explanation of all timeouts:

- tout_setup: after picking up the phone, before dialing anything
- tout_dialing: after dialing the last digit(s)
- tout_proceeding: after received the proceeding message
- tout_altering: during ringing
- tout_disconnect: after the call has been disconnected or failed

Note: The timeouts are equal for outgoing and incoming calls. The ringing tone that can be heard when making a call will timeout, as well as ringing of the phone on incoming calls. If both parties have timeouts, the party with the smaller value will timeout first.

➢**own_setup (yes | no)**
➢**own_proceeding (yes | no)**
➢**own_alerting (yes | no)**
➢**own_cause (yes | no)**

There are some announcements in the public network, that tells the caller the reason, why a call cannot be completed, or why it has been disconnected. Also the dial tone or busy tone is coming from the network to inform the current call state. The ISDN network supports up to 127 possible causes, but only some of them are specified or assigned with tones. Most causes 3result in a busy tone. LCR can give you more detailed announcements, than the public network would do. Read the section “Understanding Causes” for more details about this. If a call gets disconnected, the original announcement, that is coming from the external line can be presented to by the option “own_cause no”. If the LCR announcement should be used instead, the keyword “own_cause yes” must be used.

Tones during setup, proceeding and alerting can also be replaced. It can be useful to make a dialtone or ringing tone sound different.

➢**facility (yes | no)**

Facility information are country specific information during call. Germany for example transfers advice of charge information during a call. The facility is encoded using ASN.1 notation, that is just transparently forwarded to the terminal. To enable the feature, ‘yes’ must be given here. If supported by the telephone, it will display the charge info and even keep track of it, using a counter. The telephone company does not send any money amount, they send ‘Units’. Ask you company about the price of one unit. Also note that the charge for a call depends on the length of a call or even the total calls at the end of a month, so the displayed information may differ from the actual charge on the phone bill. Also some services will not be charged by units, like value added service numbers. This option is required if a payphone should be connected to LCR.

➢**display_cause (none | english | german | english-location | german-location | number)**

Whenever a call cannot be established, or if the call is disconnected, a number from 1-127 is transmitted. This number gives information about why the call cannot be established, or has been released. The number can be displayed to the ISDN phone that is connected to an internal port. If "none" is defined, nothing is displayed. If "number" is defined, only the value, that causes the disconnect, will be displayed. If "english" or "german" is defined, the value, causing the 'disconnect', will be shown on the display of the phone as English or German text. In front of the text, the number is shown. If "english-location" or "german-location" is defined, the value, causing the 'disconnect', will be shown on the display with the location text.

Examle: If an unassigned number has been called, the 'cause' value "1" will be transmitted. The display message in English would be "1 – Unallocated number". The display message in German would be "1 – Nummer nicht vergeben". The message with location ("german-location") could be "1 – Vermittlung (Lokal)".

Displaying 'causes' will give more detailed information, then the telephone would give.

➢**display_ext (yes | no)**

If you say "yes," the caller ID will be indicated to the telephone's display (using display information). Also IDs, that have been sent using "CLIP No Screening", will be discovered. In this case the word "fake" is appended to the caller ID, indicating that this number is made up by the caller and, may not be the real caller ID.

➢**display_int (yes | no)**

If you say "yes," the caller ID from internal extensions will be indicated to the telephone's display (using display information), and also the internal extension's number.

➢**display_anon (yes | no)**

If you say "yes", the display shows if the caller ID is suppressed.

➢**display_fake (yes | no)**

If you say "yes", the display shows if the caller ID is sent with the "No Screening CLIP" feature.

➢**display_name (yes | no)**

If you say "yes", the extension's name is displayed.

➢**display_menu (yes | no)**

It is hard to remember all dialing codes, which are defined in “routing.conf”. If you say “yes” here, the keys “*” and “#” can be used to select all rules that have been defined within a ruleset. If, after pick up of the telephone, the key “*” or “#” is pressed, the first code is displayed on the telephone. If “#” is pressed, the next code is displayed. If “*” is pressed, the previous code is displayed. Additionally the code on the display is followed with the function description for this code. Only rules that are defined by a “dialing” condition are displayed. After selecting the rule, press “0” to dial the rule.

Example: If the first code, defined for a ruleset is “dialing=0 : external”, the display would show after pressing “#” or “*”: “0=External”. If the next code, defined in the ruleset is “dialing=2 : internal extension=200”, the display would show after pressing “#”: “2=Internal extension=200”. After pressing “#”, if the end of the codes is reached, the display will continue to display the first code. After pressing “*”, if the first of the rules are reached, the display continue to display the last rule.

➢**display_dialing (yes | no)**

During dialing, the dialed number may alter. If you say “yes” here, the display would show, what’s actually dialed, instead of what has been entered on the phone. This works for all codes, like internal, external and other dialing functions.

➢**tones_dir [<directory>]**

If this directory is given, the default tones directory, which has been specified in “options.conf”, will be overridden for this extension. This option makes it possible, to define an individual set of tones for each extension.

Example: An extension may use German tones (“tones_german”) while others use the default tones (“tones_american”).
9

➢**record (<type> | off)**

Each call can be recorded. If recording is turned on, the calls will be recorded as wave encoded files. The files will be store at “/usr/local/lcr/extension/<extension>/recordings/<file>”. The file name is specified by the date and time, it has been recorded, and by the duration. The following file types can be generated while recording:

- a-law/u-law: (8 Kbytes/s) (depending on the global mode defined in “options.conf”)
- wave 16 bit mono: (16 Kbytes/s)
- wave 16 bit stereo: (32 Kbytes/s)
- wave 8 bit mono: (8 Kbytes/s)

When using stereo recording, one party will be heard on one channel, and the other party on the other.

➢**password [<pass phrase>]**

The password is needed, to gain access to an extension. When an internal telephone is picked up, no password is needed to make phone calls. For example, if callback is used, this password must be entered after PBX4Linux calls back, in case the phone (identified with the caller ID), from where the callback is done, is not known yet. If the login function is dialed, the password must be entered in order to have access to the extension. In this case the password is used to authenticate external or H.323 calls. Read the section “Internal numbering” and “Callback” for more information.

The following options control the **answering machine (voice box)**. It has nothing to do with the ISDN4Linux “VBox".

➢**vbox_mode (normal | parallel | announcement)**

By default, the **normal** mode is used. If someone gets connected to the answering machine, the announcement is played. After playing the announcement, the call is recorded. When the caller hangs up or the time limit exceeds, the recording is stopped. If the **parallel** mode is used, the call is recorded during play of the announcement. This is useful to get the name of the caller, if you record an announcement, where you pretend that you answer the phone. The caller will tell his name before she recognizes that he talks to an answering machine. Many people I know don’t talk to answering machines. If the caller should not be able to talk to the answering machine, the **announcement** mode is used. After the announcement is played, the call is disconnected.

➢**vbox_codec (mono | stereo | 8bit | law)**

See “record” for codec descriptions. This codec is used to record calls. If the stereo parameter is used in conjunction with the parallel mode, the caller and the announcement are recorded on different channels. Using stereo for ‘normal’ mode makes no sense.

➢**vbox_time (infinite | <seconds>)**

This specifies the limit of recorded calls in seconds. If “infinite” is given, then there is no limit.

➢**vbox_display (brief | detailed | off)**

The display on the ISDN telephone will show the current menu and state, when the playback function of the answering machine is called. The current counter of the message is displayed also. For small displays, use “brief”, for larger displays, use “detailed”. If your telephone gets confused, turn it “off”.

➢**vbox_language (english | german)**

When the playback function of the answering machine is called, all menu information are spoken by a voice. The voice is available in English or in German. Specify the desired language for the voice.

➢**vbox_email [<email>]**
➢**vbox_email_file (yes | no)**

To send an email whenever a message is received, give the email address using the "vbox_email" keyword. It is also possible to attach the recorded audio file to that email by saying "yes" to the file option. The source email address is specified in "options.conf", it must be a valid address, so most servers will only accept valid source and destination addresses.

➢**vbox_free (yes | no)**

If this flag is set, the voice box will only answer the call after playing the announcement. This makes sense if the audio path is connected through prior answering the call. The caller may not get charged during this time.

Note: This only works with a special feature of your external telephone line. To make this feature work, you need to enable "tones" at "interface.conf". This feature requires connecting the B-channel during call setup, and must be enabled by the local exchange.

➢**datacall (yes | no)**

Data calls will not be answered by a telephone. Sometimes it may be useful to answer a data call. In order to answer a data call but tell the telephone it is a voice call, say "yes" here. This is useful to answer calls that should be encrypted. Your telephone company may use compression and conversion for audio and speech calls, so you may need data calls instead or encryption is not possible. It is also possible to change data calls to voice calls, by defining a special rule in "routing.conf".

➢**seconds (yes | no)**

According to the ITU Q.931 standard, the connect message may include date and time. The information element consists of year, month, day, hour and minute. To also include seconds, say yes. Most telephones should support it.

➢**last_out <number>**
➢**last_in <number>**

These keywords may appear multiple times in the configuration file. "last_out" stores the 50 last external, internal or remote application's numbers that has been dialed, including the dialing code. It is used when the redial or power dial function is dialed. "last_in" stores the 50 last incoming calls. It is used when the reply dial function is dialed. The actual number of entries are defined in "main.h" using "MAX_REMEMBER" definition.

If “interface.conf” has been altered, use “lcradmin interface” to reload interfaces.

4.11 Directory

In the default installation directory there is a file:

/usr/local/lcr/directory.list

It contains numbers and their names. For every incoming caller ID (CLIP) and outgoing connected ID (COLP) this directory is parsed. If a number is found, the name can will be presented with the number. Of course the ‘display_name’ or ‘centrex’ feature must be enabled to be able to present a name.

Format: <phone number> <name>

The number must be given as received:

- National number
  The national prefix must be included or 'n' must be used.
  E.g.: n9036681733 or 19036681733 (in case '1' is the national prefix)
  E.g.: n21250993 or 021250993 (in case '0' is the national prefix)

- International number
  The international prefix must be included or 'i'.
  E.g.: i4921256483 or 0114921256493 (in case '011' is the intl. prefix)
  E.g.: i19036681733 or 0019036681733 (in case '00' is the intl. prefix)

- Subscriber number
  No prefix must be included or 's' must be used.
  E.g.: s50993 or 1733

Note: National numbers always require the area code. International numbers always require the country code + area code.

5

Using Linux-Call-Router

5.1 Running

To see a list of commands, just enter:

```
$ lcr
```

This will show a list of available options.

```
$ lcr query
```

Will show all available ISDN ports/cards, and if they are configured right to be used with LCR.

```
$ lcr start
```

The LCR will start and may exit in cause of an error. If no error occurred, LCR will continue to run, until CTRL+C has been pressed. LCR will then exit by freeing all resources.

It is also possible to run LCR as a daemon:

```
$ lcr fork
```

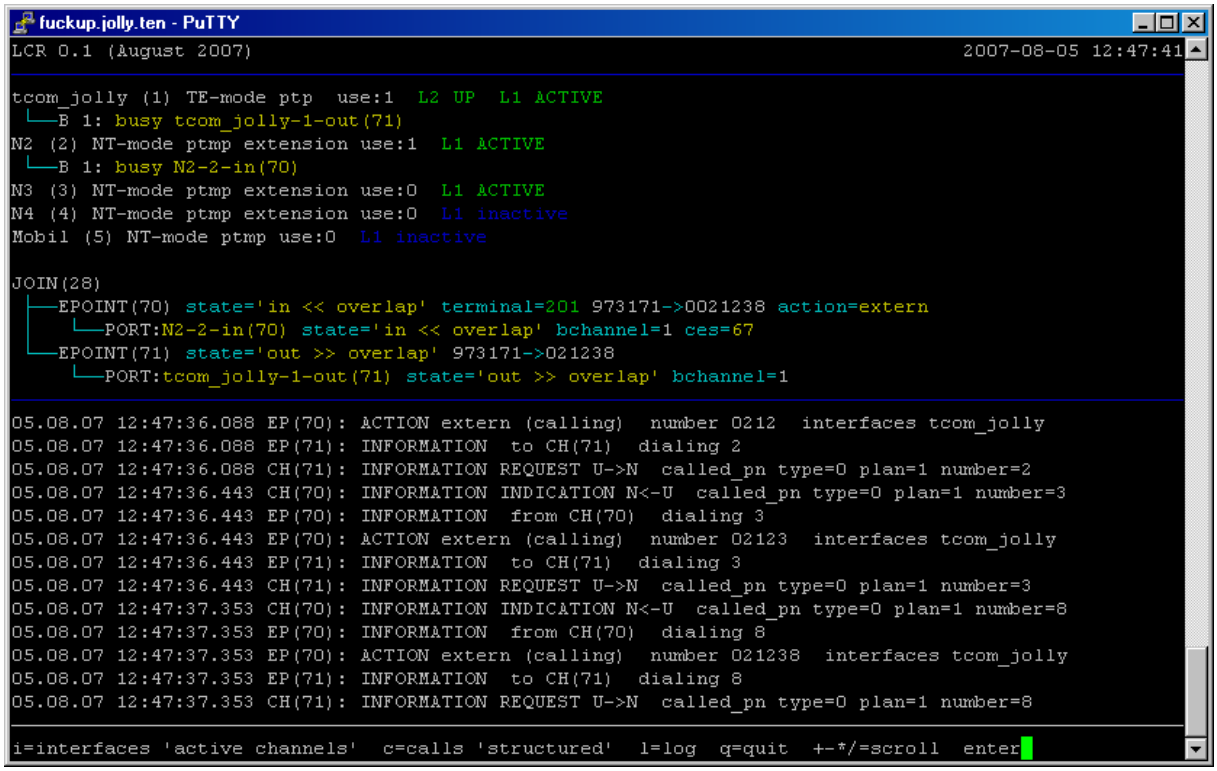
In this case LCR will still do debug output to the current TTY, but will not exit, if the shell is closed. LCR will do a daemon fork, use "**`kill`**" or "**`killall`**" to terminate. Use this to start LCR during system boot. Be sure that mISDN and its drivers are started first.

If LCR is running twice, the second process will recognize the first lock, and exit immediately with a message. The lock is located at "/var/run/lcr.lock".

To get an overview of the current call states, run **lcradmin** with the 'state' option:

```
$ lcradmin state
```

The screen will look like this:



(Figure: state of a call)

On the this example screen a call is made from an internal phone to an external line by dialing "0". The upper block show all ISDN interfaces wit ports and their B-channels. The middle block show the state of each call entity. The 'JOIN' connects two 'ENDPOINT's. The 'endpoints' have a 'PORT' entity each, that connects the 'endpoints' them to the telephone or the phone line. The structure of LCR is explained later in this documentation. The lower block will show the log as found in "/usr/local/lcr/log".

Press enter once and then enter "help", press enter gain. You get a list of commands.

To reload "routing.conf", enter "route" or run:

```
$ lcradmin route
```

To reload "interface.conf", enter "interface" or run:

```
$ lcradmin interface
```

To block a port for further incoming and outgoing calls, just enter "block <port number>" or run:

```
$ lcradmin block <port number>
```

To unblock the port again, enter "unblock <port number>" or run:

```
$ lcradmin unblock <port number>
```

To unload the mISDN driver, all ports of the driver must be unloaded. An unloaded port can the be used for a different application. Also it is useful before removing hot-plug devices. Enter "unload <port number>" or run:

```
$ lcradmin unload <port number>
```

To load the port again, use “unblock” as described above, or use “interface” to reload all interfaces.

To release a pending call, look at the endpoint’s serial ID. It is shown in the status screen, as well as in the log, starting with “EP(<ID>)”. This can be useful, if two members are joined in a conference, but they don’t or even can’t hang up for some reason. Also lost parked calls can be released. Enter “release <endpoint’s ID>” or run:

```
$ lcradmin release <endpont's ID>
```

WARNING: Don’t try to release the call by restarting LCR, since active calls will not be terminated in some cases.

5.2 Internal phone (extension)

A phone, that is connected to any internal interface of LCR will be identified by its caller ID. See the “Interface configuration” sections on how to make an interface “internal”. In order to use a phone, the caller ID must be programmed at the phone, that is called MSN number. (Multiple Subscriber Number) Whenever a caller picks up this phone to make a call, the extension, that equals the caller ID, is searched. If the extensions should be “200”, the MSN number must also be 200. If a list of MSN numbers is given for the interface, the first MSN is used if the given caller ID is not in the list. Also the extension must exist. Create an extension first, as described in chapter “Configuration”.

If a call is made to the extension, the call is forwarded to all extensions, that have been configured in the extension’s configuration file. If a phone on the internal extension should ring, it must have the extension number configured as MSN number.

Whenever a call is made from the extension, the **caller ID** is “screened” to whatever configured in the extension’s settings. The MSN is replaces by the caller ID configured. Normally, the local exchange requires the local number as caller ID.

If the internal phone makes an **anonymous** call, the call is forwarded also anonymous by LCR, regardless what is configured in the extension’s setting.

5.3 Calling intern

If the call is made from internal or external to internal, the caller must dial the code, that is specified in “routing.conf” for the extension to be dialed. Also the directory tree of the given extension must exist in “/usr/local/lcr/extansions/”. All phone on all interfaces of the extension will ring, as specified in “/usr/local/pbx/extensions/<extension>/settings”.

Example: If “`dialing=200 : intern`” is specified in “routing.conf”, the caller must dial “200” to reach extension 200. If “`dialing=0 : intern extension=200`” is specified, the caller must dial “0” in order to reach extension 200.

5.4 Calling extern

If a call is made from internal to external, the caller must dial the code, that is specified in "route.conf". It is standard for most PBXs in the world, to use the "0" for external calls. After dialing the "0", or whatever code is specified, the audio of the external line is cross-connected with the phone. The dial tone, and even announcements can be heard now.

Example: If "`dialing=0 : extern`" is specified in "routing.conf", the caller must dial "0" to make an external call.

During dialing, it is not possible to delete the last number, because it is already sent to the telephone network. LCR supports deleting, by terminating the call and making a new one. The delete function must be enabled for this extension. When disabled the digits are dialed as received. Whenever a "*" is received, the last digit is removed from the dial string, and the call is terminated and reestablished by a new one. By dialing "#", the complete number is removed from the external call, an external dial tone is given. Example: "555124*12" will dial the number "5551212", because the "4" is removed by the "*".

Info: In good old Germany, dialing of the digit '0' was called "Amt", which is the short form for "Postamt" (post office). This term is used, because the old telephone system was controlled by the German "Post", and all the switching systems were located in the buildings of post offices. Earlier, the calls were made by operators. The operator was sitting in the post office "Amt".

5.5 Calling remote application (Asterisk)

LCR offer connectivity to remote applications like Asterisk channel driver. Instead of routing calls directly using "intern" or "extern" action, the call can be route to a connected application. This is done by the "remote" rule:

```
dialing=1234 : remote application=asterisk
```

will forward the call to the asterisk, if "1234" is dialed in the current ruleset.

TBD…

5.6 Caller/Connected ID (CLIP/COLP)

The caller ID is transmitted from the caller to the called party. (CLIP) When the called party answers the call, the connected ID is transmitted from the called party to the caller. (COLP) So why is the connected ID transmitted? The called party number is known, because it has been dialed! The called party may have call forwarding enabled or may answer the call on a different telephone than that one that has been called. The caller will see who actually answers.

The ID for internal calls may be specified in the extension's settings. If the caller uses anonymous caller ID, it will be forwarded anonymous. (suppressed) Only the police or any other party with special rights may see even anonymous IDs to call back in an emergency.

5.7 Conference

LCR also supports conferences with an unlimited number of parties. Actually, LCR always use conferences for all calls. If a normal call is made, a conference with two parties is set up. The call/conference can be put on hold, to answer another incoming call, or to set up a new one. The ISDN telephone must be used to put the call on hold. The LCR can join a call on hold with the current active call. This may be done by using DTMF tones. Read the "Keypad" section for information on how to join calls. If calls are joined, the members of the call/conference increase.

Whenever a conference is put on hold, all parties connected in a conference may continue to talk. Inactive parties are also muted, so the conference is not disturbed. If there is only one party in the conference, who is active, she will hear the hold-music and get a notification, that the call is currently inactive, until someone else retrieves the call.

If any party in a conference hangs up, the party is removed from the conference. If only one party is left, the party will also be disconnected.

5.8 Keypad

Keypad dialing is used to control the call, after it has been connected. The keypad may be dialed using DTMF tones, but also using the ISDN keypad facility. The keypad facility is not supported by all ISDN telephones, so DTMF may be used in most cases. DTMF may be recognized even if no DTMF tone is sent. This may happen during analyzing of voice. Sometimes small parts of the human voice equal to very brilliant DTMF tones. Therefore a sequence is used, whenever a call is connected. The sequence must be dialed quickly. If pauses of more than 2-3 seconds is made between the digits, the sequence is not detected. The sequence begins with "*" and ends with "#". In case of keypad facility, "*" and "#" must not be dialed.

Keypad	DTMF	Function
0	*0#	Disconnect the current call and give a dial tone
3	*3#	Join the current call with the call on hold.
7	*7#	Enable encryption using manual keying. (explained later)
8	*8#	Enable encryption using automatic keying. (explained later)
9	*9#	Disable or abort encryption.

When a call is put on hold, or the call has been disconnected using keypad, the caller may dial using DTMF or ISDN keypad facility. If a call has been disconnected, or if the caller wishes to get back to the dial tone because of wrong dialing, the caller must enter "#" twice, within two seconds.

5.9 Callback

The callback makes only sense, if dialed from external telephones. The caller must dial the code for callback, as specified in "numbering_ext.conf". If the caller doesn't transmit a caller ID, callback is not possible at all. If the feature "CLIR Ignore" is available to the ISDN line, that may only available to police or emergency phones, callback is possible, because the caller ID is available, even though it is restricted. If the ID is not available, the call will be

disconnected with an error, indicating that the service is not available. After dialing the callback code, the caller may continue dialing the number, that will be dialed after callback. This feature is only available with PBX lines. The number of digits that may be dialed after the code for callback depends on the maximum number that will be transmitted by the telephone network. It may be enough to dial an abbreviation from the phonebook. If nothing is dialed after the code for callback, the dial tone will be presented after callback.

As soon as the caller hangs up while processing the callback rule, LCR will wait 2-3 seconds, and start calling back. The caller ID of the calling party will be used to indicate the callback. This only works if the external line has the "CLIP No Screening" feature. On normal lines, the default caller ID is given by the local exchange. After the caller answered the call, she must enter the extension's password, in order to use the callback. If the password is correct, it will be stored in the extension's password list "/usr/local/lcr/extensions/<extension>/callbackauth". If the password is already specified, LCR will directly dial, what has been dialed after the callback coded. If nothing has been dialed, the dial tone is presented.

After callback, the DTMF feature is used to input digits. The caller must dial the same number as she would dial when she picks up an internal phone.

If a call gets disconnected, the caller may dial "#" twice, to get a dial tone again, without doing another callback. During a call, the caller must disconnect using "*0#" sequence. To end the callback, the caller must hang up.

Callback gets rejected during trigger call, if

- the caller ID is not available.
- the extension does not exist.
- the password is not given.

The call gets disconnected after callback if

- the password input timed out.
- any digit is wrong after entering the number of digits equal to the password.
- the caller hangs up.

5.10 Test mode

The test mode is used to make certain tests for internal and external telephones. The rule for test mode may be specified in "routing.conf". For external calls using the test mode, the external line must be a PBX line, to support the dialing of more digits after the test mode code. Alternatively the test may be given as parameter "prefix=<code>".If the code is given, the test mode can be used with multipoint lines also, but only one test can be executed. Here is a list of tests that can be made:

➢**1**

The call is proceeding. A 'proceeding' signal is sent to the calling ISDN telephone.

➢**2**

The call is alerting. An ‘alerting’ signal is sent to the calling ISDN telephone.

➢**3**

The call will be connected. The caller ID of the caller is sent back as connected ID. The audio is echoed (looped). This is useful to test the audio roundtrip delay. You can compare hardware bridging with software bridging delay, by just holding your mouth peace to the hand-free speaker of you phone. If you hear a “ringing” echo, you have a delay that you can recognize. If you hear just a smooth “whistle” tone, than your echo is too fast to recognize.

➢**4**

The call will be connected with a continuous test tone. This is good for testing if LCR runs without gaps. High interrupt load may cause buffers in the kernel to underrun. Also hearing some gaps are ok, because they result in de-jittering, because cards may have different clock source and audio delay has to be keep low.

➢**5**

The call will be connected with the hold music.

➢**6<cause>**

After dialing 6 and three more digits, the call will be connected, and the announcement, that will be specified by the ‘cause’ value. The ‘cause’ value is a number between 1 and 127. If the ‘cause’ value has less than three digits, it must be filled with zeros in front. This is useful to test announcement’s audio files.

➢**7<cause>**

This is the same as the previous function, except that the call gets really disconnected. The ‘cause’ value is sent with a disconnect message. This is useful to see how the telephone reacts on different ‘causes’ or how the local exchange interprets and forwards the ‘cause’ value. On external calls the call will be disconnected, and the announcement of the telephone system is used.

➢**8**

Sends back a ‘disconnect’ message with normal call clearing cause (16).

➢**9**

Connects the call and sends back the text “Welcome to Linux” and “12345678” as connected ID. This is useful to test COLP and display facility.

5.11 Understanding Causes

A “cause” value is a number between 1 and 127. It is used to explain why a call cannot be completet. ISDN telephones use them to give text messages on its display. The public telephone system uses them to send announcements, which explain the ‘cause’ value. All

cause values are defined in Q.850 of ITU-T standard. In the source file “cause.c” is a list of all causes.

Example: If the LCR’s routing finds, that a dialed number doesn’t exist, it sends a ‘disconnect’ message with the cause value “1”, that means that the number is not allocated. Someone in the USA would then get the announcement: “The number, you have dialed, is not assigned.” In Germany the announcement would be: “Kein Anschluß unter dieser Nummer.”

Also the disconnect message has a “location” value. The location is used to indicate where the cause in generated. If the cause is generated by the local exchange, the location is “public local”. If the served user (the phone that receives the disconnect) is connected to a different exchange, it will receive the location “public remote”, because for her it is the remote exchange.

5.12 Answering machine (voice box)

The answering machine is divided into two parts. The first part answers a call, plays an announcement and records a message. The second part offers a voice menu to plays the recorded messages and to record the announcement. There are two ways to record messages. The easiest way is to define a rule in “routing.conf”:

dialing=<digits> : vbox-record extension=<extension>

When the given digits are dialed, the answering macine (voice box) of the given extension is called. The other way is to forward the call to the voice box. This can be done in the extension’s “settings” file. Instead of entering an external telephone number at **cfu**, **cfb** or **cfnr**, the key word “vbox” will forward the call to the voice box instead. To forward to the voice box after 30 seconds of ringing without answer, use the following parameters in the settings of the extension:

```
cfnr        vbox
cfnr-delay  30
```

When a call is received on the voice box, it will play the **announcement**, which is stored in “/usr/loca/lcr/extensions/<extension>/vbox/announcement.isdn”. After that, it will record until the maximum recording time is reached, as specified with “vbox_timeout”, or if the caller hangs up. It is also possible to record during the announcement. This is useful to pretend, that the call is not answered by an answering machine. In this case, the caller might begin to talk during the announcement. Then it is useful to have her voice recorded. To use the feature use the following parameter in the settings:

```
vbox_mode   parallel
```

To just play the announcement without recording, the following parameter is used:

```
vbox_mode   announcement
```

After the announcement is played, the call is disconnected.

To **play** the recordings of the answering machine, a special rule must be used in "routing.conf":

dialing=<digits> : vbox-play

Because each extension has its own answering machine, it is required to call from that extension. In order to play back messages from extern, the 'login' code or the callback must be used. It is also possible to just play back the recording of someone else's voice box:

dialing=<digits> : vbox-play extension=<someone else's extension>

After dialing the digits for playing back, a voice will speak the menu. If no recording is stored in the answering machine, the voice will tell this first. During the speech, the following digits may be dialed using DTMF or keypad dialing:

"1" Play previous message.

"2" Play current message. The date, time and caller ID is spoken and written to the display. Then the display shows the total recording time and the current time playing. Pressing "2" during intro will directly playback. Pressing "2" during playback, will stop. Pressing "2" after stop, will continue the playback.

"3" Play the next message.

"4" Rewind the current playback. Pressing "4" again, will double the rewind speed whenever it is pressed. If the beginning is reached, the playback will continue to play forward. Pressing "2" during rewind, cause continue to play at normal speed.

"5" Stop the current playback. The menu is spoken afterwards.

"6" Wind the current playback in forward direction. Pressing "6" again, will double the wind speed whenever it is pressed. Pressing "2" during rewind, cause continue to play at normal speed.

"7" Use to record the announcement. After pressing 7, the announcement can be recorded by pressing "1", played by pressing "2", and aborted by pressing "3".

"8" Use to store the current message in the recordings directory. After pressing "1", the message is stored, and aborted by pressing "3".

"9" Use to delete the current message. After pressing "1", the message is deleted, and aborted by pressing "3". When recording, almost the first second is cut off, to avoid recording the sound of the keypad. After recording, a beep tone is added to inform the caller that the recording will start.

"0" Use to call the caller of the current message. This is only possible, if the caller ID is available and not restricted.

It is also possible, to select one of the functions above, by using "*" (for previous item) and "#" (for next item). On the display, the current function is displayed. To execute, press "0". The display will only be shown on an internal phone, since external lines cannot receive display information.

To send an email whenever a message is received, give the email address using the "vbox_email" keyword within the extension's config file:

```
vbox_email        someone@somewhere.com
```

It is also possible to attach the recorded audio file:

```
vbox_email_file  yes
```

This will only work if "sendmail" package is installed. Also it must be configured correctly. Some mail accounts may recognize your dial-in-IP-address as a spammer's address, so mail should be forwarded by your "sendmail" to your provider's email server. I cannot explain here how to setup "sendmail". Please visit www.sendmail.org for further info.

5.13 Encryption

Encrypted Calls

Just pick up the phone and call somebody who uses also LCR, press a key and the call will be encrypted. - It's easy isn't it?

There are two types of encryption, that can be selected:

- **pre-shared keying**
- **automatic key exchange**

Both have advantages and disadvantages. Pre-shared keying requires a key to be shared. It is essential that nobody can eavesdrop the key when it is exchanged. The best way to exchange a key is to meet the other party. Always note that if the key gets stolen before and even after the call, the call can easily be decrypted in the future. If you use automatic key exchange, the key will be erased whenever the call has been disconnected. Also you don't need to share the key before you make a call. The key exchange procedure does this for you. To be sure, that no man-in-the-middle modifies the keys, the telephone's display facility is required. To exchange an encrypted session key, RSA is used, but this may take some seconds up to calculate the keys.

Preshared keying

Preshared keying requires some security about handling of the keys:

- The key must be good random.
- The key should not be given to other parties (group of people)
- The key must be stored secure.
- The key can be used in the future to decrypt encrypted calls.
- The key must be exchanged in a secure way.

Both parties must be identified by the caller ID or dialed number. In order to select the correct key, the caller must transmit her caller ID or the called party will not know what key to select. The keys are stored in the extension's directory

“/usr/local/lcr/extensions/<extension>/secrets". The first keyword is the number of the remote party. The number must be given as dialed (without the external prefix). For incoming calls the number must be given as shown (without the external prefix). If the dialed number doesn't match the caller ID of the same party, make two entries with the same key, but different numbers. The second keyword is the authentication method. Only "manual" keying is supported at this time. The third keyword is the encryption method. Only "blowfish" encryption is supported at this time. The last keyword is a string that represents the key. The string is converted from ASCII into binary. If the string is given with "0x" prefix, it is converted from HEX into binary. For blowfish use the maximum key length of 448 bits (56 bytes or 112 HEX digits). To enable encryption, both parties will have to press digits '7'. If the telephone sends DTMF tones instead of keypad information, both parties must press '*7#'. The display will show if the key is found or if an error occurred. A special inband audio signal is given for successfull encryption (two bells) or failure (marine horn). This method is very fast, because no calcualtion process is required. If you here noise, the keys or algorithm mismatch.

To turn of encryption, use keypad digit '9' or DTMF sequence '*9#'. A “marine horn” will indicate the deactivation of encryption.

Automatic key exchange

It is very simple to enable this type of key exchange. It works if "crypto" library is enabled during compilation time. Press digit '8' on your keypad. If your telephone sends DTMF tones instead of keypad information press '*8#'. You should now notice the message "Key Exchange" on the display of your telephone. This only works if your telephone supports display messages. Display messages are essential to avoid man in the middle. The called party should now hear some clicks in his telephone. This is data transmission of the identification procedure. Messages are exchanges via audio channel, so you will hear click whenever some message is transmitted.

The automatic key exchange procedure:

Both parties have to press digit '8' or DTMF '*8#' to enable automatic key exchange. This must happen within 10 seconds. Both will send a random number and the CPU power to each other. The first party that selects the automatic key exchange, will get the information about the remote party. The CPU power is used to determine the fastest computer. If both parties have equal speed, the random number is used.
(In case of equal random number, the b-channel seems to be looped.) The party with the faster CPU or the bigger random number is master and will receive "Master" on her display. The other party will get "Slave" on the display. The master calculates and RSA key pair. One of the keys, the public key is sent to the slave. The slave generates a random session key, encrypts it with the public key and sends it back to the master. Only the master is able to decrypt the session key using the private key. The session key is just a random number that is calculated from noise of all active B-channels during key exchange. It can be expected to be "good random".

To avoid man-in-the-middle attack, both parites will see some values of their public key on the display. If they are equal, there is no man in the middle. One party can tell the other about parts of the public key. The voice of the party 'signs the public key'.

Both parties will use the session key to encrypt and decrypt the audio data using "Blowfish" algorithm.

Man-in-the-middle attack is shown now: Imagine a device (the man in the middle) is attached to your phone line. It intercepts the call to the other party. The device pretends to be the other party and will use the automatic key exchange procedure. On the other side it pretends being you and also does the automatic key exchange procedure with the other party. Now the device can listen to the decrypted transmission, because the transmission is only secure between the device and the parties. If the device is master of both connections, both public and private keys may be equal, so you and the other party will receive the same public keys. In this case both parties are slave, so this is an incident of a man in the middle. If one party would be master, the device must generate a key pair for the other party. It is not possible to generate a key pair with a given public key. Both parties would see different public keys in their display.

Again, to avoid man-in-the-middle attacks:

- During negotiation, one party must see "Slave" and the other "Master" in their displays.
- Both parties must see the same public key digits in their display. It is not required to see the complete key.

About the key on the display: Each bit of the key will halve the coincidence of not discovering a man-in-the-middle-attack. If you compare the first block (4 hex digits), you will end up with a chance of 1:65536 not discovering a man-in-the-middle-attack. If your compare all four blocks (16 hex digits), the chance will be 1:18446744073709551616.

"My display shows a display message three seconds, then it is gone"

Don't worry about your display, just press keypad '8' or enter DTMF '*8#' again and you will see the key message again.

Now the blowfish works:

The Blowfish encryption and decryption is done by the kernel module "mISDN_dsp.o", that is also used for real time audio processing. A special command is used to enable and disble the audio transmission.

The audio is processed in the file "dsp_blowfish.h". A block of 9 samples is converted into a seven-bit-sample. The resulting 63 bits plus another random bit is used to encode a block of 64 bits using blowfish. The block is then converted into 8 bytes of seven bits plus one byte of 8 bits. The upper bit of the first 8 bytes are used for sync information. Decoding is done by reversing the process.

Encryption in a conference:

It is not possible to encrypt a conference with three or more parties. But it can be done before the parties are joined within a conference. Let's say you like to have a secure conversation with two persons, both have also LCR. Call the first person and start encryption. Put the person on hold and call the seconds person. Start encryption on the second call. Join the calls as usual (keypad '3' or DTMF '*3#'). Check the public keys with each party before you join, because in a conference it is not possible to redisplay them.

5.14 Other tools

$ gentones

This tool generates tones. When it is called without parameters, it will show the following help:

```
Usage:
gentones wave2alaw <wav file> <alaw file>
gentones wave2ulaw <wav file> <ulaw file>
gentones tone2alaw <frq1> <frq2> <length> <fade in> <fade out> <alaw file>
gentones tone2ulaw <frq1> <frq2> <length> <fade in> <fade out> <ulaw file>
Length and fade lengths must be given in samples (8000 samples are one
second).
Tones will append to existing files, wav files don't.
```

It can be used, to generate law encoded tones with one or two frequencies. This is mainly used to create patterns, like dial tones, busy tone and other audio patterns. Depending on your telephone system, you need to use “tones2alaw” or “tones2ulaw”. There are two frequencies to specify “frq1” and “frq2”. If only one frequency should be used, enter 0 for “frq2”. The “length” specifies the total length of the tone in samples. 8000 samples are one second. 2000 samples are ¼ of a second. “fade in” and “fade out” is used to make the start and stop of the tone soft. If “fade in” is 800, the sound will fade in within 1/10 of a second. This makes tones very smooth and avoids the ‘click’ sound at the beginning. “fade out” respectively. Whatever specified for “fade in” and “fade out”, the tone will be as long as given at “length”. If you don’t want to use fades, set “frq1” and “frq2” to “0”. I suggest at least 50 sample for fade to silence.

The given file will be appended to “alaw file” or “ulaw file”. If you like to create a new one, but the name still exist, delete the file and then start creating it. This is useful to create a sample with more than one tone. Example: A busy tone normally is made out of ½ second of a tone, and one ½ second of silence. To add silence, just enter 0 for “frq1” and “frq2”.

It is also possible to convert a wave file to a-law or mu-law. The wave file **must** have a sampling rate of 8000. It doesn’t matte what bit-resolution or how many channels it has (stereo or mono). The sampling rate will not be converted, so it must be 8000 samples/second. The resolution should be 16 bits for best quality. a-law and mu-law have better resolution than 8 bits. The data of a-law and mu-law is 8 bits sample, but since it is quantised, the quality will be 12 bits. 12 bits sounds almost as good as 16 bits. Wave files are only available with 8 or 16 bits resolution.

The use of a-law or mu-law files, result in a faster processing, since the samples must not be converted into a-law or mu-law for ISDN use. ISDN uses only a-law or mu-law samples.

$ genwave

To convert an a-law or mu-law file into a wave file, “genwave” is used. If it is called without parameters, the following help is given:

```
Usage:
genwave ulaw2wave <alaw file> <wav file>
genwave alaw2wave <alaw file> <wav file>
```

Use “ulaw2wave” or “alaw2wave”, whatever your law-samples are encoded in. The output file must be given with “.wav” extension. The sample is converted into 16 bits, mono, 8000 samples/s. “alaw” or “ulaw” files are always mono with 8000 samples/s.

Use this tool to convert recordings of calls or answering machine, if they are stored as a-law or mu-law samples.

5.15 Parking a call

Using the HOLD feature

A call can be parked by using the “hold” feature of the ISDN telephone. This should be supported by any telephone. The call will not be released, but only the B-channel. The hold feature is useful to answer or setup a new call while holding the other. If a waiting call without a B-channel should be received, the hold feature must be used in order to answer the call. LCR supports this feature. There is no limit for the number of calls that can be placed on hold. Most telephones only support one call to be on hold. If a B-channel is available, a call on hold can be retrieved. If no channel is available, the active call must be put on hold first. All this is done by the telephone, so you just need to switch between the calls.

During a call on hold, the remote party will get a notification, that the call is on hold, and receive the hold-music. If the call is a conference, the call gets removed until retrieved.

Using the SUSPEND feature

A call can be suspended (parked) also. This feature should be supported by all ISDN telephones, but is not used by many people, for some reason. An active call can be parked by using a code. This can have up to 8 digits. Some telephones support only two digits, some use fixed digits and some support no digits at all. The digits are used to identify the call. The call can then be resumed on a different phone. It is even possible to resume a call on any other interface (NT-mode) of the LCR. This allows to transfer a call or to move a telephone to a different port.

Also during suspend, the remote party will get a notification, as described above for the ‘hold’ feature.

Please refer to the telephone’s manual.

NOTE: These features are only supported for NT-mode and do not apply to external lines. It is not possible to hold/suspend an external call. This is also not possible for most PBX-lines.

6

Using with Asterisk (chan_lcr)

6.1 About chan_lcr

Channel drivers are the interfaces between Asterisk and you telephone hardware. Some implement VoIP protocols, others connect physical interfaces to Asterisk. chan_lcr will use the ISDN part of Linux-Call-Router and connect it with Asterisk. Instead of writing a stand-alone driver, chan_lcr will benefit from these features:

- LCR runs indipendently from chan_lcr, so ISDN ports and interfaces will not shutdown when restarting Asterisk.
- Calls are multiplexed by LCR, so the same interface (port) can be used for LCR routed and Asterisk routed calls.
- Incoming calls are first routed through LCR, so if Asterisk is not running, alternative call routing may be applied.
- LCR has a lot experience with handling ISDN protocol. Most work of LCR is done with interfacing ISDN. A stand-alone implementation will do the work twice.
- No extra ISDN configuration has to be done for Asterisk

Even though LCR handles the ISDN interfaces, the B-Channels are directly accessed by chan_lcr. Briding and even hardware briding is possible with chan_lcr.

6.2 Compiling chan_lcr

If 'configure' is available, it should automatically detect Asterisk source. Otherwhise edit the Makefile and enable the switch “WITH-ASTERISK” by removing the '#' sign in front. Be sure to have the latest Asterisk installed. The includes of Asterisk must be available in the include path. Compile and install LCR as usual. The chan_lcr.so will be built and installed with the other channel drivers of Asterisk.

6.3 Running LCR and Asterisk

First run LCR as usual (lcr fork). Run Asterisk using '-v' option:

```
$ lcr fork

$ asterisk -v
...
chan_lcr.so => (Channel driver for Linux-Call-Router Support (ISDN
BRI/PRI))
```

If the chan_lcr driver is installed, the Asterisk will load it. The driver will connect to LCR via unix socket. Be sure that LCR runs with the same user rights as Asterisk. When killing LCR, chan_lcr will try to reestablish connection to LCR until it is restarted.

6.4 Routing incoming calls to Asterisk

Every incoming call from you phone line or phone you pick up is routed by LCR. To route calls directly to Asterisk, rather inside LCR, here are three examples. Edit "routing.conf" which is found in the installation directory ("/usr/local/lcr").

To route calls by MSN number (external calls), insert the following rule into your "[extern]" ruleset or whatever ruleset you use to route external calls.

```
[extern]
...
dialing=5678   : remote application=asterisk context=extern_context exten=5
```

All external calls to MSN number 5678 will be routed to chan_lcr (Asterisk). The context will be "extern_context" and the dialed extension will be "5". (On a PTP interface you may omit the "exten" keyword, so the dialed suffix will be used.)

Now you may add your extension to "/etc/asterisk/extensions.conf".

```
[extern_context]
exten=>5,1,Dial(SIP/phone)
```

In this case, all calls to extension "5" will be forwarded to chan_sip and call whatever you specified for "phone". Please refer to Asteerisk documentation.

Alternatively you can route all calls from one interface. If you have an internal interface for you phone and like to route all calls from it to Asterisk, edit the main rule in "interface.conf" of LCR.

```
[main]
interface=Int   : remote application=asterisk
... other rules
```

If "context" is omitted, the interface's name is used. In this case "Int". Because "exten" is omitted, the dialed number will be the extenion's number. Edit "/etc/asterisk/extensions.conf" and add your extensions for the interface "Int":

```
[Int]
exten=>0,1,Dial(LCR/Ext/${EXTEN:1})
```

```
exten=>_0.,1,Dial(LCR/Ext/${EXTEN:1})
exten=>5,1,Dial(SIP/phone)
exten=>6,1,Dial(SIP/phone2)
exten=>8,1,Dial(LCR/Int/1234)
```

By dialing "0", you will get routed to external line. Dialing "5" or "6" will do a SIP call. Dialing "8" will do an call on the internal interface "Int" and dial your MSN "1234".

If you like to use only chan_lcr for Asterisk and don't care about LCR at all, you can route all calls of all interfaces (internal and external) to chan_lcr:

```
[main]
                : remote application=asterisk
```

The interface name will be the extension's name for Asterisk.

To dial from Asterisk to an ISDN interface of LCR, use the following syntax:

```
extern=><digits>,1,Dial(LCR/<number>)
extern=><digits>,1,Dial(LCR/<interface>/<number>)
extern=><digits>,1,Dial(LCR/<interface>/<number>/<options>)
```

Where "number" is the number to be dialed on ISDN interface. If no "interface" is given, the first free interface is used.

6.5 Options

For both, incoming and outgoing calls, options can be applied. For outgoing calls from Asterisk to LCR, options can be appended to the dial string as fourth parameter. For incoming calls from LCR to Asterisk, a special dial application is used.

To list all options while Asterisk is running, enter CLI. (asterisk -r) Give the following CLI command:

```
fuckup*CLI> show application lcr_config
fuckup*CLI>
  -= Info about application 'lcr_config' =-

[Synopsis]
lcr_config

[Description]
lcr_config(<opt><optarg>:<opt>:...)
Sets LCR opts. and optargs

The available options are:
    d - Send display text on called phone, text is the optarg.
    n - Don't detect dtmf tones on called channel.
    h - Force data call (HDLC).
    t - Disable all audio features (required for fax application).
    c - Make crypted outgoing call, optarg is keyindex.
    e - Perform echo cancelation on this channel.
        Takes mISDN pipeline option as optarg.
   vr - rxgain control
   vt - txgain control
        Volume changes at factor 2 ^ optarg.
```

```
The 'show application' command is deprecated and will be removed in a
future release. Please use 'core show application' instead.
fuckup*CLI>
```

Here is an example of disabling mISDN_dsp.ko audio features for a call from LCR that is routed through Asterisk. (Incoming from LCR and then outgoing to LCR.)

```
[Int]
exten=>3,1,lcr_config(t)
exten=>3,2,Dial(LCR/Int/1234/t)
```

Be aware of the red characters. The characters '1' and '2' are used to enumerate two extensions that match the same extension '3'.The first rule will call lcr_config. The parameter 't' = transparent is given. It disables the kernel audio features like dejittering and hardware briding. This only work for calls that come from LCR, since lcr_config requires an chan_lcr instance to configure. The second rule uses parameter 't' as a call option to LCR. The dial application creates an LCR instance with the options 't'. Now both call instances (incoming and outgoing) have audio features disabled. Briding is done inside chan_lcr.

7

Tuning

7.1 Disable Kernel 'hacking'

Kernel 'hacking' in the Linux kernel provides varous checks to find bugs in the kernel code. Some features track actions and memory allocations using tables or lists. Especially page allocation debugging slows down your system extreamly. Try to disable kernel debugging features you don't need, and disable features that causes significant slowdown. See kernel config help for that.

Here is an example: A Pentium machine with 350 MHz used 100% CPU when trying to make three calls at a time. After disabling page allocation debugging, 15 calls just caused 50% CPU load.

7.2 Load tones into memory

To reduce jitter, caused by hard disk access, edit the file “options.conf” and enable the “**fetch_tones**” option. Specify all tone set that should be loaded into memory. Read the “Configuration” section for option description. This requires more memory, but needs no more hard disk load, when playing tones and announcements.

7.3 Set scheduler to high priority

If may processes run on your server, you may experience high CPU load. Since LCR is a real-time-process that requires steady execution, but not much CPU, it should run on a high priority.

To run LCR with high priority, alter the “**schedule**” option in “**options.conf**” Use a value of 0 for normal scheduling, this is the default. The highest value is “99”. If a process has a higher value than others, it will get CPU, if it requires the CPU. Note that even a value of 1 may cause to lock up your machine, if LCR failed due to an endless loop bug.

7.4 Get rid of IDE

LCR may run on old and slow systems. This is quite ok, since handling some ISDN calls don't require much CPU power. If you experience voice gaps due to hard disc traffic, you might have an IDE drive. If you can use SCSI or serial IDE, you will be lucky. But don't worry of some gaps, they will be recovered by mISDN's DSP code. They will not lead into increase of audio delay.

8

Debugging

8.1 Logging

Logging is enabled by default, because the log file is given at "/usr/local/lcr/log". It will show what's going on inside the deep LCR code. If you pick up the phone and you hear a busy tone, you don't know what happens until you view the log file. Especially when using routing rules, this may help to see if a rule matches and how it's action is processed.

You will see something like this if you pick up the phone:

```
05.08.07 16:55:22.129 CH: DL_ESTABLISH INDICATION N<-U  tei 66
05.08.07 16:55:22.175 CH(5): NEW_CR INDICATION  callref new=0x10042
05.08.07 16:55:22.175 CH(5): SETUP INDICATION N<-U  calling_pn type=4 plan=1 present=0
screen=0 number=201  called_pn type=0 plan=1 number=995  channel_id exclusive=0 channel=-3
hlc coding=0 interpreta=4 presentati=1 hlc=1  bearer coding=0 capability=0 mode=0 rate=16
multi=-1 user=3
05.08.07 16:55:22.175 CH(5): CHANNEL SELECTION (setup)  channel request=any  hunting
channel=free  conclusion 'channel available'  connect channel=1
05.08.07 16:55:22.175 CH(5): BCHANNEL create stack  channel 1  stack id=0x10010200
address=0x50010202
05.08.07 16:55:22.175 CH(5): BCHANNEL activate  channel 1
05.08.07 16:55:22.175 EP(5): SETUP  from CH(5)  caller id number=201 present=allowed  dialing
995
05.08.07 16:55:22.175 EP(5): SCREEN (found in MSN list)  msn 201
05.08.07 16:55:22.175 EP(5): TONE  to CH(5)  directory tones_american  name dialing
05.08.07 16:55:22.175 EP(5): ACTION (match)  action goto  line 12
05.08.07 16:55:22.175 EP(5): ACTION goto/menu (change to)  ruleset intern  dialing 995
05.08.07 16:55:22.175 EP(5): ACTION (match)  action test  line 71
05.08.07 16:55:22.175 EP(5): ACTION test  test 'hold music'
05.08.07 16:55:22.175 EP(5): CONNECT  to CH(5)  connect id number= present='not available'
05.08.07 16:55:22.175 EP(5): TONE  to CH(5)  directory tones_american  name hold
05.08.07 16:55:22.175 CH(5): PROCEEDING REQUEST N->U  channel_id exclusive=1 channel=1
05.08.07 16:55:22.175 CH(5): CONNECT REQUEST N->U  date day=5.8.7 time=16:55:22
05.08.07 16:55:22.194 CH(5): BCHANNEL control  DSP-RXOFF 1
05.08.07 16:55:22.194 CH(5): BCHANNEL control  DSP-DTMF 1
05.08.07 16:55:24.356 CH(5): DISCONNECT INDICATION N<-U  cause location=0 value=16
05.08.07 16:55:24.356 CH(5): RELEASE REQUEST N->U  cause location=1 value=16  reason 'no
remote patterns'
05.08.07 16:55:24.356 EP(5): RELEASE  from CH(5)  cause value=16 location=0-User
05.08.07 16:55:24.405 CH(5): RELEASE_COMP INDICATION N<-U
05.08.07 16:55:24.405 CH(5): RELEASE_CR INDICATION  callref 0x10042
05.08.07 16:55:24.405 CH(5): BCHANNEL deactivate  channel 1
05.08.07 16:55:24.406 CH: BCHANNEL remove stack  channel 1  stack id=0x10010200
address=0x50010202
05.08.07 16:57:04.389 CH: DL_RELEASE INDICATION N<-U  tei 66
```

In this case, someone picks up the phone, here is what happens. First the layer 2 ISDN link is established from phone with terminal ID 66, a new call is created. The dialed number “995” is included in the setup message. The phone requests “any channel acceptable”, so LCR check incoming channel selection list and find channel “1” is free. The b-channel is activated. The ruleset “main” is parsed and rule “goto” matches, because it is a call from an extension. In the ruleset “intern”, the rule “test” matches, because the first digit of the dialed number is “99”, as requested. The test action is executed with the number “5”. This test plays the hold music. The tone name “hold” is played and a connect message is sent to the telephone. First the telephone receives a processing message with the selected channel ID, then it gets a connect message with date and time. The B-channel is now configured for DTMF. After about two seconds, the caller hangs up. A disconnect is received. Also a release is sent, because the phone does not send any disconnect tone. Also the endpoint is released. The ISDN process is removed and also the B-channel is closed. After about two minutes, the layer 2 ISDN lin k is released.

8.2 Doing traces

To get a more detailed output you may want to trace. Also if you have may calls at a time, you will be flooded with log information. You may want to select only special numbers or interfaces. To get a list of possible trace option, use “lcradmin”:

```
$ lcradmin trace help
```

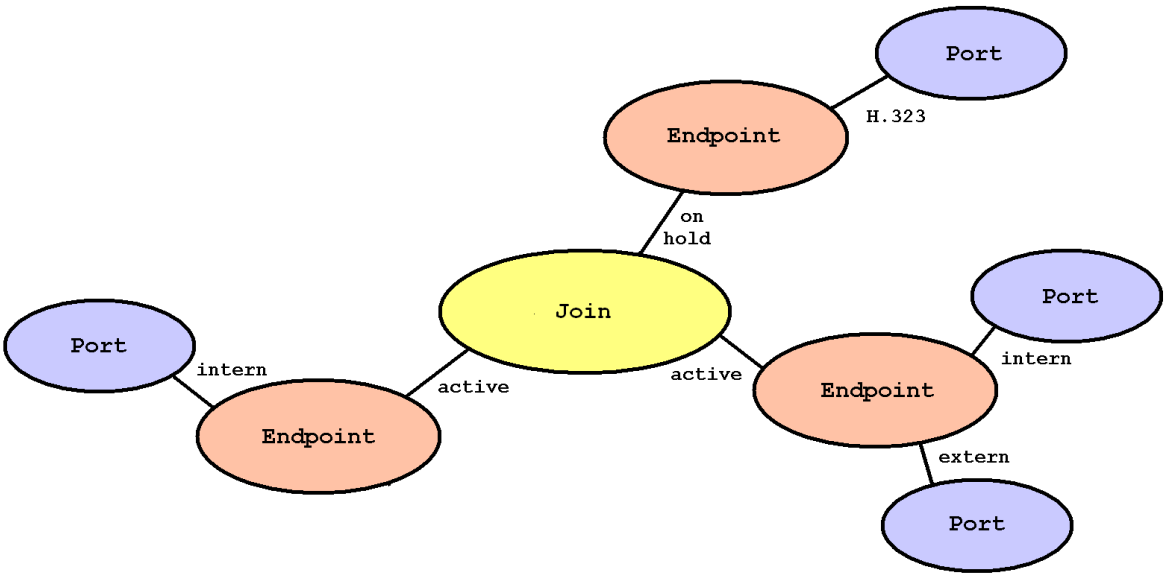
By default everything is traced in detail:

```
$ lcradmin trace

-------------------------------------------------------------------------------
Port: 2 (BRI PTMP NT)  Interface: 'N2'  Caller: ---
Time: 05.08.07 17:14:49.628  Direction: IN  Dialing: ---
-------------------------------------------------------------------------------
CH(6): SETUP INDICATION N<-U
 calling_pn : type       = 4
            : plan       = 1
            : present    = 0
            : screen     = 0
            : number     = 201
 channel_id : exclusive  = 0
            : channel    = -3
 hlc        : coding     = 0
            : interpreta = 4
            : presentati = 1
            : hlc        = 1
 bearer     : coding     = 0
            : capability = 0
            : mode       = 0
            : rate       = 16
            : multi      = -1
            : user       = 3

-------------------------------------------------------------------------------
Port: 2 (BRI PTMP NT)  Interface: 'N2'  Caller: '201'
Time: 05.08.07 17:14:49.628  Direction: ---  Dialing: ---
-------------------------------------------------------------------------------
CH(6): CHANNEL SELECTION (setup)
 channel    : request    = any
 hunting    : channel    = free
 conclusion :              'channel available'
 connect    : channel    = 1
```

9

Architecture

9.1 port, endpoint, join



(Figure: Structure of LCR)

This picture shows the main structure of LCR. There are three classes of objects involved:

➢**Port**

Whenever someone picks up a phone, or a call is received from external line, a **port** object handles the incoming call. Also if someone is called, a port handles the outgoing call. All ports only exist, if a call is made. An incoming ISDN call creates a port of type ISDN. After a port receives an incoming call, it creates an endpoint, and sends messages to it. An endpoint can have multiple ports, so it is possible to ring a call on multiple ISDN ports. The port class is defined in “port.h”, and implemented in “port.cpp”, “mISDN.cpp”, “dss1.cpp”, …. The base class “port.cpp” is incomplete and does not very much. It is used to provide functions, which all classes of ports use. The is inherited by “mISDN.cpp”, “vbox.cpp”, ….

➢**Endpoint**

An **endpoint** is created, whenever a port receives a call. An endpoint acts as a call manager. It controls the dial tone, does the processing of the dialling, and creates a join object, in case of internal, external or remote application calls. An endpoint can have only one active join. The endpoint class is defined in “endpoint.h”, and implemented in “endpoint.cpp”. Also each endpoint has an application. It is defined in “endpointapp.h”, and implemented in “endpointapp.cpp”. The application used for PBX functionality is called “apppbx”. Because the endpoint that controls the PBX is large, it is split into other multiple files: “apppbx.cpp”, “route.cpp”, “action.cpp”, “action_vbox.cpp”.

➢**Join**

A **join** is created by an endpoint, whenever an internal, external or remote call is made. By default a join object is inherited by a JoinPBX object or a JoinRemote object. They are defined and implemented in “join*”. A PBX join may have an unlimited number of endpoints. A remote application join may have exactly one endpoint. A PBX join will normally have two endpoints connected, but in case of may endpoints, the join will form a conference. If an endpoint disconnects from the join, or if it sends any message, no other endpoint will get a note of it, in case of three or more endpoints. If there are only two endpoints left in a join, the messages from one endpoint are transferred to the other endpoint. If one endpoint releases, the join gets released, and the release is transferred to the other endpoint. If a join is created, the join creates another endpoint and transfers set-up information.

➢**Messages**

All entities communicated via messages. The messages are buffered, so that they are sent after the entity is processed. The message contains a serial number, where the message was from, and where it is sent to. If an entity is removed, but there is still a pending message, the message is dropped. An entity will be created, and receives a set-up message afterwards. An entity will be removed, when it receives a release message. If an entity is related to others, it will send a release to the other entities before it is removed. Most messages are closely related to ISDN messages with their information and functions, but expanded for more information than ISDN offers. Messages are stored in a forward linked chain of structures. A message is appended to the end of the chain, and removed from the beginning. All messages are defined in “message.h” Functions to send and retrieve a message are implemented in “message.c”.

➢**A simple call procedure**

An ISDN telephone, which is connected to an internal ISDN port, is picked up. It sends a set-up message to the ISDN port. A **port** entity receives that set-up message, and creates an **endpoint** entity. This endpoint sends back a message, that it needs more digits, and another message to the port, that the dial tone should be streamed. The caller may now enter digits. These digits are sent to the endpoint entity via the port entity and are processed. Let's assume that the caller dials the code for external calls. The endpoint received the digits for external calls, creates a **call** entity and sends a set-up message to the call entity. The call entity receives the set-up message and creates an **endpoint** entity, and sends the set-up message to it. The endpoint entity receives the set-up messages, and selects a **port**, as specified by the type of

number that has to be dialled. In this case it is an external call, so a free port, which connects to an external line is searched. If it is found, the set-up message is sent to that port. From now on, all call-state messages are transferred via this port-endpoint-call-endpoint-port chain. The call entity also interconnects the endpoints, by sending mixer information to the isdn ports, receiving audio data from the endpoints, and sending audio data down to the endpoints.

8.2 Tones and announcements

For the LCR all tones must be defined as samples. These samples may be **a-law** or **u-law**, or even **wave** encoded.

The formats are:

- a-law/u-law: (8 Kbytes/s) (depending on the global mode defined in “options.conf”)
- wave 16 bit mono: (16 Kbytes/s)
- wave 16 bit stereo: (32 Kbytes/s)
- wave 8 bit mono: (8 Kbytes/s)

Each tone can have two files. The first file has the extension ".isdn" or “.wav” and defines the introduction of the tone pattern. The second file has the extension "_loop.isdn" or “_loop.wav” and defines the endless loop of that tone.

If the first file does not exist, the second loop file is played right as the tone starts. If the second loop file does not exist, the first file is played and silence is played afterwards. If no file exist, silence is played.

Tone names and their occurrence:

➢**NULL (null pointer) or “”**

Silence

➢**“busy”**

The called user is busy.

➢**“dialtone” / “dialpbx”**

An internal dial tone is played and LCR is waiting for dialing information. “dialpbx” is used for internal dial tone, “dialtone” if the external dial tone is overridden.

➢**“dialing”**

After receiving ‘dialing’ information, this tone is played.

➢**“proceeding”**

The dialed number is complete.

- **“ringing” / “ringpbx”**

The called destination is ringing. “ringpbx” is used when an internal phone rings. “ringing” is used if the external ring tone is overridden.

- **“error”**

An unspecified error has occurred.

- **“release”**

The call is disconnected and LCR waits for release of the call (hang-up of the phone).

- **“ring”**

Sound of a ringing phone. This is used to indicate an incoming call during connected-mode.

- **“test”**

A sine test tone with 1000 Hz.

- **“hold”**

Hold music.

- **“redial”**

This is an indication during power dialing. A short tone (should not be more than one second), indicates, that the call failed after redialing. It is useful to indicate when the call failed and a retry is done.

- **“cause_xx”**

The call cannot be completed as dialed. The given ‘cause’ with the hex number “xx” is played. If the file for the ‘cause’ doesn't exist, the error file is played. ISDN provides ‘cause’ 1-127 which is "cause_01" - "cause_7f". Some ‘causes’ cannot be played; because they occur when audio transfer is not possible.

- **“password”**

When entering the password, this prompt is given.

- **“activated”**

Function activated.

- **“deactivated”**

Function deactivated.

- **“crypt_on”**

Indicates that encryption is now enabled.

- **“crypt_off”**

Indicates that encryption is now disabled or has failed.

- **“cause_80”**

The given extension (MSN) is not registered. (The cause value “1” is actually sent to with the disconnect message.)

- **“cause_81”**

This indicates unauthorized call from extension. (The cause value “21” is actually sent to with the disconnect message.)

- **“cause_82”**

This indicates unauthorized external call from extension. (The cause value “21” is actually sent to with the disconnect message.)

- **“cause_83”**

This indicates unauthorized national call from extension. (The cause value “21” is actually sent to with the disconnect message.)

- **“cause_84”**

This indicates unauthorized international call from extension. (The cause value “21” is actually sent to with the disconnect message.)

- **“cause_85”**

This indicates unauthorized number or prefix. (The cause value “1” is actually sent to with the disconnect message.)

- **“cause_86”**

The dialed extension doesn't exist. (The cause value “1” is actually sent to with the disconnect message.)

- **“cause_87”**

This selected service is not authorized for the extension. (The cause value “21” is actually sent to with the disconnect message.)

Additional tone sets can be downloaded from ‘http://www.linux-call-router.de/download’.

The **answering machine** (voice box) uses own samples sets. They are installed in “/usr/local/pbx/vbox_english” and “/usr/local/pbx/vbox_german”. I will not describe the tones here. They are self explaining. Note that some tones have short pauses, some have long pauses, and some have no pause at the end.

10

Support

9.1 Mailing list

To be done…

9.2 Buying NT-Mode capable ISDN card

Check out what cards can be used. At http://www.linux-call-router.de you will find a list of supported cards and brand names. Let me tell you that almost every budget card uses the HFC-PCI controller that supports NT-mode.

9.3 Getting an NT1 / NTBA

Every phone company deals with ISDN NTs. Go to the service men of your local telephone company and ask for the trash bin. Old NTs with expired warranty are thrown away. Service men at telephone companies are very nice people. Almost every one might have an old NT for you if you ask gently. They are broken, but 80-90% of them still have a working power supply. Since the internal electronics is driven by the power from the underground wire, the NT will not disturb the S/T interface. To be sure that the ISDN-coils works, take an volt meter and check if you got about 40 volts between pin 3 and 4, as well as between 5 and 6. ISDN-coils almost never break.

Be carefull: NTs out of the trash might be wet. Also I know from our "million customers", that an NT1 can makenoise and even smoke.

Another place is ebay. Many sellers have old (still working) NTs. You may buy one there. Always remember that most phone companies don't sell them, so most offered NTs are still owned by the phone company, even if they are not used anymore. NTs are cheap, so don't feel bad about buying one.

Of course an NT can be bought! Ask your local telephone company to sell one to you.

Pssst…. Or maybe just talk to you telephone's company hotline and tell them "My NT1 makes clicks and noises, if I shake the U-line…." . Find out what happens then….

11

References & Relates Projects

11.1 Asterisk

*Asterisk

Here is a description of the Asterisk project, as it is found at 'http://www.asterisk.org':

Asterisk is a complete PBX in software. It runs on Linux and provides all of the features you would expect from a PBX and more. Asterisk does voice over IP in three protocols, and can interoperate with almost all standards-based telephony equipment using comparatively inexpensive hardware.

Asterisk provides Voicemail services with Directory, Call Conferencing, Interactive Voice Response, Call Queuing. It has support for three-way calling, caller ID services, ADSI, SIP and H.323 (as both client and gateway). Check the 'Features' section for a more complete list.

Asterisk needs no additional hardware for Voice over IP. For interconnection with digital and analog telephony equipment, Asterisk supports a number of hardware devices, most notably all of the hardware manufactured by Asterisk's sponsors, Digium. Digium has single and quad span T1 and E1 interfaces for interconnection to PRI lines and channel banks. In addition, an analog FXO card is available, and more analog interfaces are in the works.

Also supported are the Internet Line Jack and Internet Phone Jack products from Quicknet.

Asterisk supports a wide range of TDM protocols for the handling and transmission of voice over traditional telephony interfaces. Asterisk supports US and European standard signaling types used in standard business phone systems, allowing it to bridge between next generation voice-data integrated networks and existing infrastructure. Asterisk not only supports traditional phone equipment, it enhances them with additional capabilities.

Using the IAX Voice over IP protocol, Asterisk merges voice and data traffic seamlessly across disparate networks. While using Packet Voice, it is possible to send data such as URL information and images in-line with voice traffic, allowing advanced integration of information.

Asterisk provides a central switching core, with four APIs for modular loading of telephony applications, hardware interfaces, file format handling, and codecs. It allows for transparent switching between all supported interfaces, allowing it to tie together a diverse mixture of telephony systems into a single switching network.

Asterisk is primarily developed on GNU/Linux for x/86. It is known to compile and run on GNU/Linux for PPC. Other platforms and standards based UNIX-like operating systems should be reasonably easy to port for anyone with the time and requisite skill to do so. Asterisk is available in the testing and unstable Debian archives, maintained thanks to Mark Purcell.

A mailing list is available where developers and uses discuss bugs and problem.

11.2 OpenH323

Here is a description of OpenH323, as it is found at ‘http://www.openh323.org’:

„The OpenH323 project aims to create a full featured, interoperable, Open Source implementation of the ITU H.323 teleconferencing protocol that can be used by personal developers and commercial users without charge.

OpenH323 development is coordinated by an Australian company, Equivalence Pty Ltd, but is open to any interested party. Commercial and private use of the OpenH323 code, including use in commercial products and resale, is encouraged through use of the MPL (Mozilla Public license).“

"The aim is to 'commoditize the protocol'. By giving the stack away, maybe we can stop everyone obsessing over how to format the bits on the wire, and get them writing applications instead."
Craig Southeren,
co-founder of OpenH323

A mailing list is available where developers and uses discuss bugs and problem.

11.3 ISDN4Linux

The homepage of ISDN4Linux can be found at ‘http://www.isdn4linux.de’. This is the home of the Kernel driver for ISDN support. Additionally the „isdn4k-utils“ can be downloaded here, in order to configure and use ISDN4Linux. Note that ISDN4Linux is old, and LCR uses the new mISDN driver. The mISDN driver now has it’s own page at ‘http://www.misdn.org/’. This page has a FAQ and addresses to the mailing lists.

11.4 ITU

To be done…

11.5 Cologne Chip

To be done…

Appendix

Appendix A. Copyright + License

Refer to http://www.gnu.org/licenses/gpl.html for complete license text.

Appendix B. Telephones

If someone like to do some portraits about telephones with benefits and problems, just mail.

TBD….