

# Design and Construction of a Talking Call Line Identification Unit

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**Abstract**—The most electronic devices convey information to the user by displaying it on a Liquid Crystal Display (LCD). Visually impaired people cannot benefit from using electronic devices with LCD technology. Adapted products exist, but because no market for these products has been established, relatively small numbers are imported. Therefore only a few useful products are available at an affordable cost. We present the design and construction of a talking call line identification (TCLI) unit with integrated sound output to be used on Telkom<sup>1</sup> lines. This unit incorporates audio and visual input/output of all on-board functions. Visually impaired users can navigate the unit settings assisted by prerecorded voice prompts and big input buttons. A graphical LCD unit completes the unit, as it could still be used by the non-impaired. The reproduction of a human voice is accomplished by prerecording on a solid-state Information Storage Device (ISD) that is capable of recording eight minutes of voice data. This enables the unit to announce the incoming callers number or an associated recording of the callers name and comes as a great benefit to not only the visually impaired, but normal users of telephones as well.

**Index Terms**—Calling Line Identity, Frequency Shift Keying, Information Storage Device, Visually Impaired, Telephone Line Terminal Equipment, Audio-output.

## I. INTRODUCTION

**I**n a technologically advanced world, the visually impaired depend more on others to help them function. The reason being that every electronic device uses some form of visual output that renders these "useful" devices useless to the visually impaired. A good example would be the Identical unit distributed by Telkom Ltd<sup>1</sup>. The Identical device uses an LCD display or output medium, through which information about the incoming call (eq. date and time, incoming callers number) is displayed. The Identical unit uses the Calling Line Identity (CLI) service offered by most PSTN operators around the world. When enabled for a monthly fee, the CLI service sends important information, called CLI data, regarding the incoming call to the called users telephone line terminal equipment (TLTE). This identifies the incoming party before the phone is answered, as the CLI data is presented to the TLTE during the long silent period after the first ring signal.

<sup>1</sup>South African Public Switched Telephone Network (PSTN) Operator

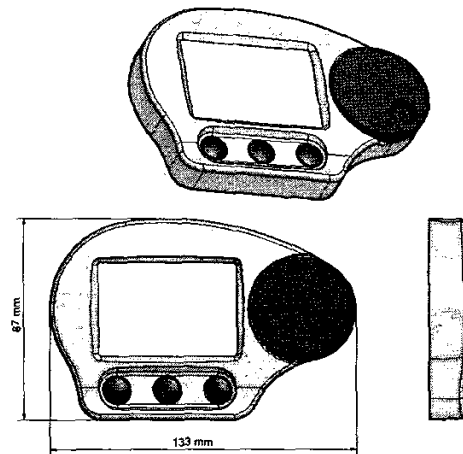


Fig. 1. Proposed design for the Talking Call Line Identity (TCLI) unit

CLI enabled TLTE devices are normally inserted in parallel with a standard telephone line and decode CLI data that originates from the calling party's exchange. CLI data is transmitted using phase consistent frequency shift keying (FSK). The inherent design short-coming encountered in the Identical unit is overcome by incorporating sound output into this newly designed CLI unit. The addition of a voice output gives the impaired person access to data and information regarding the incoming call not previously possible. See Figure 1 for the proposed look of the unit.

Section II explains the fundamentals of FSK modulated signals and summarises the specifications about CLI data structure according to the European and South African standards. The conceptual design, device functions and peripherals are discussed in Section III.

## II. BACKGROUND

CLI data is sent from the calling party's exchange using a FSK modulation scheme. This section gives some background knowledge on FSK modulated signals and the CLI data

TABLE I  
MODULATION RATES AND CHARACTERISTIC FREQUENCIES FOR A  
1200 BAUD FSK DATA-TRANSMISSION CHANNEL

	$F_0$ Mean frequency	$F_Z$ Symbol 1 Mark	$F_A$ Symbol 0 Space
Mode 2: 1200 baud	1700 Hz	1300 Hz $\pm$ 10 Hz	2100 Hz $\pm$ 10Hz

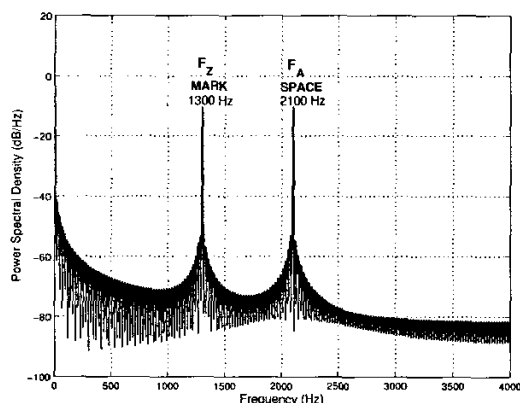


Fig. 2. Super-imposed spectral density of the two symbols used in FSK modulated CLI transmission

structure.

#### A. Frequency Shift Keying

Communications established by switching on the PSTN must presently be able to support data transmission through phase consistent FSK modulation, conforming to the International Telecommunications Union (ITU) [1] standards. Half duplex transmission at 1200 baud involves two symbols or modulation frequencies. The frequencies of interest are presented in Table I. Because only one symbol is present at any given moment, two symbols (space and mark) would have a combined power spectral density with peaks at  $F_A$  and  $F_Z$ . This is depicted graphically in Fig. 2.

The time period per symbol at the given baudrate of 1200 symbols/s is 833.3  $\mu$ s/symbols, that is the number of seconds a signal with frequency  $F_Z$  or  $F_A$  is present. Fig. 3 shows an FSK modulated byte that represents the ASCII letter "U" or binary "01010101"<sup>2</sup>. A logic 0V is represented by  $F_A$  and logic 5V by  $F_Z$ . The FSK signal shown in Fig. 3 has no discontinuities and is phase consistent throughout logic level changes.

#### B. CLI Data Structure

The timing diagram for CLI data reception is presented in Figure 4. The CLI data structure was investigated and

<sup>2</sup>Approximately thirty eight ASCII "U"s represents the channel seizure. See Fig. 4.

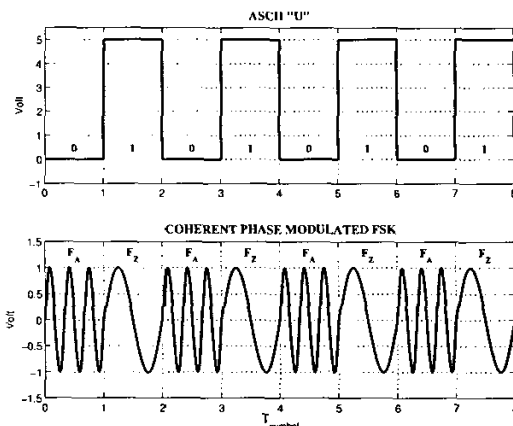


Fig. 3. FSK modulated ASCII "U".

simplified by using the following specifications:

- ETS 300 659-1 [2]
- ICASA TE-010 [3].

Further references, are made to the ETS specification only, as the ICASA specification is derived from the European standard. The CLI data structure is presented graphically in Fig. 6 and contains the data-link packet. The structure is explained as follows.

1) *Data-link Packet*: After the first ring burst the channel seizure signal, mark signal, message type, message length and check sum will be presented as per ETS 300 659-1 specification.

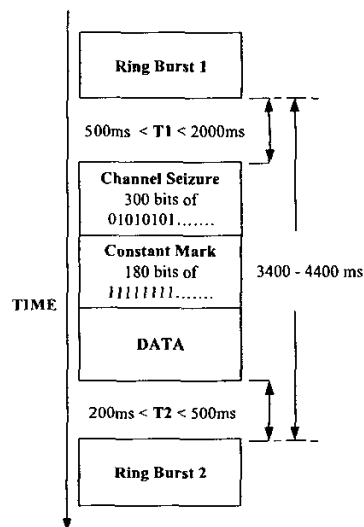


Fig. 4. On-hook FSK transmission as per ETS 300 659-1 specification.

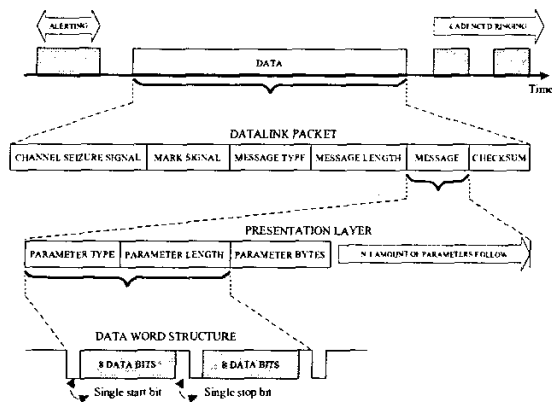


Fig. 5. CLI data structure as per ETS 300 659-1.

TABLE II  
MESSAGE TYPE CODING

Type (Binary) HGFE DCBA	Type (Hexadecimal)	Message name	Status
1000 0000	80H	Call Setup	Mandatory
1000 0010	82H	Message Waiting Indicator	Non-Mandatory

2) *Channel Seizure Signal*: It shall consist of a block of 300 continuous bits of alternating "0"s and "1"s. The first bit to be transmitted shall be a "0". The last bit to be transmitted shall be a "1". It should only start the data transmission in "on-hook" data transmission.

3) *Mark Signal*: The mark signal consists of a block of  $180 \pm 25$  mark bits or, as a network operator option, the mark signal may consist of  $80 \pm 25$  mark bits.

4) *Message Type (1 Octet)*: The message type octet or byte contains a value to identify the message type. Relevant Message type coding parameters are listed in Table II.

5) *Message Length (1 Octet)*: The message length octet or byte contains the number of octets of the Data Link layer message (not including the Message type, Message length and Checksum octets). This allows a presentation-layer message length between 3 and 255 octets.

6) *Checksum Octet (1 Octet)*: The checksum octet contains the two's complement of the modulo 256 sum of all the octets in the message starting from the Message type octet up to the end of the message (excluding the Checksum itself). The protocol does not support error correction or message retransmission<sup>3</sup>. No sequence number or acknowledgement shall be used for the data messages transmitted from the exchange to the TLTE.

7) *Presentation Layer*: Our main interest is in the Call Setup (80h) message type parameter, as it contains the relevant

<sup>3</sup>NOTE: The unit discards any incorrect Data-link messages.

TABLE III  
CALL SETUP PARAMETERS

Message Type	Mandatory Parameters	Non-Mandatory Parameters
Call Setup	Date and Time	Calling Party Name
	Calling Line Identity (CLI)	Reason for absence of Calling Party Name
	Reason for absence of CLI	Complementary CLI
	Called line ID	Network Message system status
	Call type	Extension for operator use
	First Called Line Identity (in case of forwarded call)	
	Type of forwarded call	
	Type of Calling User	
	Redirecting Number (in case of forwarded call)	

information about the incoming call. Fig. 6 sets out the data structure of the presentation layer. The Call Setup data consists of parameters listed in Table III.

8) *Data Word Structure*: Each data word consists of one start bit (space), then eight data bits followed by one stop bit and 0-9 mark bits as shown in Fig. 6.

### III. PROPOSED DESIGN

This section identifies conceptual boundaries and the physical input/output requirements for the prototype. Refer to Fig 6 for the functional block diagram of the unit. By definition all embedded systems contain a micro-controller and residing software. With the exception of a few common features, the embedded hardware components are usually unique. Thus each system adheres to completely different requirements, any or all of which can affect the compromises and tradeoffs made during the development of the product.

#### A. Line Interface

The line interface adheres to standard specifications for TLTE that connects to the PSTN. The regulatory body that enforces these specifications is the Independent Communications Authority of South Africa (ICASA). ICASA-TE-001 Standard Specification for TLTE for Connection to the PSTN [4] was used in the design of the units line interface:

The units ability to dial stored numbers is accomplished by playing prerecorded Dual Tone Multi Frequency (DTMF) tones over its line interface. Therefore the line interface includes a speech circuit that facilitates bi-directional flow of analog signals. The line interface does not loop the line when dialling but requires the user to pick up the connected telephone handset. Play back of prerecorded DTMF tones that reside on the ISD is known as tone dialing. Because DTMF tones should have a maximum frequency tolerance of  $\pm 1.5\%$ ,

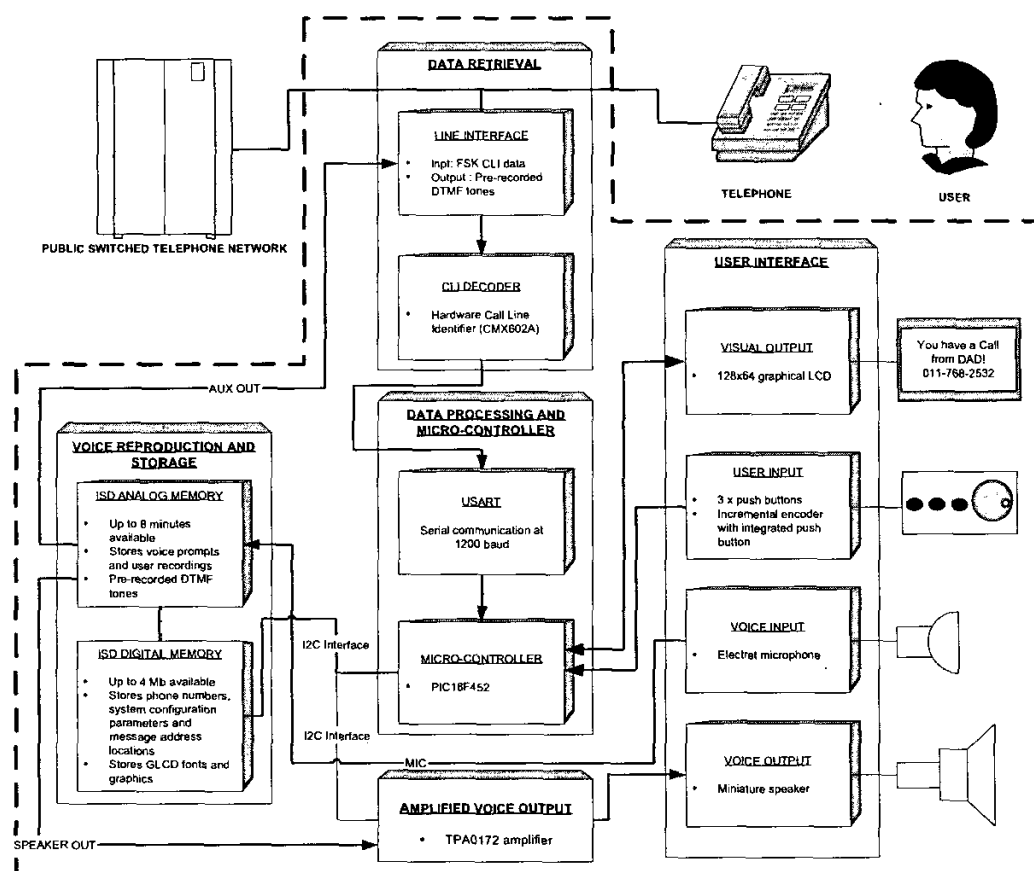


Fig. 6. Functional block diagram for TCL1 unit

the ISD device uses a very stable external clock frequency. This ensures the correct separation between the high and low frequencies of the DTMF tone. The line interface does include first and second stage lightning protection.

#### B. Call Line Identifier - CMX602A

The CMX602B decodes CLI data and presents it to the micro-controller via a serial output line. The identifier has two transmission modes: in mode one data is sent at a constant 1200 baud towards the micro-controller and mode two requires the micro-controller to clock the data out of the hardware identifier via two additional I/O lines. Mode two requires extra embedded code development, but does not require the micro-controller to have an on-board USART. The micro-controller used in the project did fortunately have an on-board USART and mode one was implemented. An external crystal of 3.579545 MHz was used for synchronised 1200 baud transmission.

The hardware identifier proved to be an easy and inexpensive way of decoding FSK modulated CLI data and required

less development time than software decoding techniques.

#### C. Voice Reproduction and Storage - ISD5116

ISD5116 Chiporders provide high quality, fully integrated, single-chip Record / Playback solutions for 1- to - 16 minute durations. Address, control and selection are accomplished through an  $I^2C$  interface to minimise pin count (only two control lines are required).

1) *ISD Digital Memory*: The ISD5116 device is capable of storing digital data, in addition to analog data. This feature is utilised by storing frequently called numbers, received call lists, system configuration parameters and message address locations.

2) *ISD Analog Memory*: A sample frequency of 8 kHz ensures good reproduction/recording quality of analog signals. If the ISD storage is utilised for analog recordings only, a total of 8 minutes of recording time is available. This analog storage space is utilised to record voice prompts, pre-recorded DTMF tones and user recordings.

#### D. User Input

Users navigate the device menus with three buttons, and one multi-function rotating dial button which acts as a menu navigator and volume control. A good quality electret microphone enables the recording of frequent caller names. The user recordings enable the unit to announce an associated recording of the callers name. These recordings are stored in the ISDs analog memory. Recordings of the user's voice at a distance of 0.5 meters deliver acceptable sound quality and playback is audible at a distance of 10 meters.

#### E. Output

1) *Voice output - miniature speaker*: The unit plays back the pre-recorded and user-recordings through a miniature 8 $\Omega$  speaker. ISD output signals are amplified and the output gain or volume is adjusted digitally via the push buttons or incremental encoder. Digital volume control is achieved by implementing the TPA0172 amplifier.

2) *TPA0172 2-W amplifier with I<sup>2</sup>C bus*: The ISD speaker output, SP+ and SP-, are connected to a TPA0172 amplifier available from Texas Instruments. Key features of this amplifier are:

- I<sup>2</sup>C bus compatible
- Delivers 1-W/channel rms output power into 8 $\Omega$  load
- Total harmonic distortion is less than 0.2 percent over the audio spectrum
- Internal thermal shutdown protection mechanism
- Low power consumption when in shutdown mode.

This device utilises the I<sup>2</sup>C bus to control its functionality, which minimises the number of external components needed, simplifies the design, and frees up board space for other features. The overall gain of the amplifier is controlled digitally by the volume control registers which are programmed via the I<sup>2</sup>C interface. Each register contains six bits, that allows 64 gain steps, and two bits that mute the amplifier. Included with the device is integrated depop circuitry that virtually eliminates transients that cause noise in the speakers during power-up, power-down, and while in transition in and out of shutdown mode.

3) *Audio output: ISD recordings*: Recorded voice output consists of the following recordings and prompts:

- Notification prompts: Notifies the user of incoming calls, missed calls, date and time etc
- Spoken numerals and DTMF tones (0-9)
- System navigation prompts: This helps the user navigate the different menus.

4) *Visual output: LCD*: The unit incorporates a 128 x 64 pixel graphical LCD to display the following CLI information:

- Incoming callers number. A list of up to 99 previous calls can be accessed.
- Displays "Private" if CLI number delivery has been blocked.
- Displays "Unavailable" if CLI information is unavailable.
- Date and time.

- Type of calling user. A Graphic image depicts the calling user type and is displayed on the LCD. The different incoming call types are:

- Voice call.
- Data call.
- Mobile phone.
- Payphone.

With every transition of the menu item a voice prompt is played back to assist the visually impaired to identify the features of the unit. The voice output, given when navigating menus, could be muted by disabling the feature in the unit's configuration options.

#### F. Data Processing and Micro-controller

1) *Micro-controller*: The PIC18F452 micro-controller is a single chip, self-contained computer which incorporates all the basic components of a personal computer, just on a smaller scale. The micro-controller is a programmable chip that controls most of the above mentioned peripherals. The choice of micro-controller was influenced by the following criteria:

- Central processing unit (CPU) or speed
- ROM memory
- RAM memory
- Input and Output peripherals
- On-board timers
- Interrupt circuitry
- Available communication buses
- C compiler compatibility

The PIC18F452 micro-controller ranks in the top 10 controllers ever manufactured by Microchip. It scores highly in all the above mentioned criteria and is compatible with Microchips C18 compiler. All embedded C code was developed in Microchips MPLAB IDE and compiled with their C18 compiler. Development time was drastically minimised by using this compiler.

2) *Processing power*: The graphical LCD peripheral requires the highest processing power, but these requirements are met easily by the PIC18F452 that runs on a 40 MHz clock. Because most instructions take four clock cycles to complete, 10 million instructions can be executed in a second.

3) *Memory Requirements*: The memory requirements are influenced by the size of the graphical user interface (fonts and graphics) and the underlining process code. The TCLI unit is able to store 20 frequent caller names and numbers, including a list of 99 previously received call numbers. The total program memory used should not exceed 32 kBytes as external EEPROMs would be required to free up some of the PICs program memory. This is highly unlikely because the ISD could store up to 174 KBytes of digital data when two thirds of its analog memory is used for recordings. This leaves the unit with ample space for the program memory needed for further development and the addition of new features.

4) *Input and output requirements*: The PIC18F452 micro-controller is able to control/interface with the following devices in the system:

- Line interface
- CMX602B call line identifier
- Graphical LCD unit
- ISD5116 information storage device
- 4 push buttons and the incremental encoder
- Texas Instruments TPA0172 2 Watt amplifier
- PC serial port by using its on-board USART.

#### IV. UNIT OPERATION

When a incoming call initiates, CLI data is stored in memory. A checksum computed during reception is compared to the received checksum. If the received checksum differs from the computed checksum, the unit halts, displays "Data error" on the GLCD and a voice prompt informs the visually impaired of the data error. If there is no difference in received and computed checksums the incoming callers number is printed on the GLCD. Memory is checked for previous user recordings that match the received number. If found, the user recording is played back in the following format "You have a call from <user recording>". If a user recorded the name "Fred Flinstone" and associated the recording with Fred's telephone number, the following output would be heard when Fred phones. "You have a call from Fred Flinstone". The output "You have a call from" is spoken words in a male voice. The output "Fred Flinstone" is spoken words in the voice of the user.

When no associated recordings are found in memory, a male voice announces the following "You have a call from <number>". Frequent caller names can be recorded by browsing the received call list. If the desired number is highlighted, the user initiates the record process when the "Record" button is pressed. The following prompts are heard from the unit "Record this callers name after the tone, press any button when finished.....<BEEP>". Recordings are terminated by pressing any button during the recording process. User recordings can be re-recorded, reviewed or erased in the same manner.

#### V. CONCLUSION

Because the unit incorporates audio and visual output, it is now possible for the visually impaired to navigate the CLI unit settings and access the information offered by the device without the assistance of others.

It is sincerely hoped that the device developed and described herein will improve the state of this technology and will be of great benefit to the visually impaired and to ordinary users of telephones.

#### ACKNOWLEDGMENT

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#### L<sup>A</sup>T<sub>E</sub>X

This document was prepared using L<sup>A</sup>T<sub>E</sub>X [6] and the IEEEtran class [5].

#### REFERENCES

- [1] Telecommunications Standardization Sector of ITU, *Data Communication Over the Telephone Network - 600/1200-Baud Modem Standardized for the General Switched Telephone Network*, ITU Recommendation V.23.
- [2] European Telecommunications Standards Institute, *Public Switched Telephone Network (PSTN); Subscriber line protocol over the local loop for display (and related) services; Part 1: On hook data transmission - ETS 300 659-1*, February 1997.
- [3] Independent Communications Authority of South Africa, *Functional requirements for analogue calling line identification (CLI) customer premises equipment - ICASA TE-010*, May 2002.
- [4] Independent Communications Authority of South Africa, *Standard Specification for TLTE for Connection to the Public Switched Telephone Network - ICASA TE-001*, January 1999.
- [5] M. Shell (2002), *How to Use the IEEEtran L<sup>A</sup>T<sub>E</sub>X Class*, IEEEtran homepage on CTAN, [Online]. Available: <http://www.ctan.org/tex-archive/macros/latex/contrib/supported/IEEEtran/>
- [6] H. Kopka and P. W. Daly, *A Guide to L<sup>A</sup>T<sub>E</sub>X*, 3rd ed. Harlow, England: Addison-Wesley, 1999.



**Marius A. Beukes** (M'1995) received a B.Eng degree in electrical engineering from the Rand Afrikaans University, Johannesburg in 2002, and is currently working towards an M.Eng. He is investigating the design and construction of a Talking Call Line Identity Unit. His interests include embedded design, home automation, FPGAs and embedded software algorithms for implementation in the telecommunications industry.