FSK message indication service for analog telephone

SIMON NEDOK, BOŠTJAN VLAOVIČ, TATJANA KAPUS, ZMAGO BREZOČNIK
Faculty of Electrical Engineering and Computer Science,
University of Maribor
Smetanova ul. 17, SI-2000 Maribor, SLOVENIA
{simon.nedok,bostjan.vlaovic,kapus,brezocnik}@uni-mb.si

Abstract: - This paper describes an implementation of “message indicating” on an analog telephone using the Frequency Shift Keying (FSK) for telephone exchange system SI2000 V5. With a FSK receiver connected between the telephone and the local exchange, we can provide functions like CLIP (Calling Line Identification Presentation), CLIR (Calling Line Identification Restriction), called party number, and redirect number presentation to analog telephone users. FSK is primarily used for caller identification, but can also be used for storage of missed calls and any other supplementary service based on caller identification.

Key-Words: - telecommunications, analog telephone, switch node, FSK, SDL

1 Introduction

In the last decade, deployment of telecommunications services is on its rise. The progress focused on supplementary services, introduction of Integrated Services Digital Network (ISDN), and convergence of traditional and packet-based networks. Nevertheless, there are still a lot of analog telephone subscribers worldwide. FSK (Frequency Shift Keying) modulation enables telecoms to provide additional services to the analog telephone subscribers. Therefore, we have decided to implement message indication service on the analog telephone for the exchange system SI2000 V5.

The telecommunication system SI2000 V5 is developed by IskraTEL d.o.o., Slovenia. It is a contemporary data-controlled switching system and can be connected to telecommunication networks on a variety of hierarchical levels. It can be used as an access node for analog and ISDN subscribers, as a switch node for small or medium-sized networks, or as a trunk unit. It provides a simple way to plan and build telecommunication networks with different telephone exchange configurations.

Our message indicating service implementation for SI2000 V5 supports three technologies: AON (Avtomaticheskoe Opredelenie Nomera), FSK based on ETSI (European Telecommunications Standards Institute) specifications, and FSK based on Bellcore specifications. AON is a registration signalling system which is used in Russian networks for caller identification when user’s call signalling does not provide call category (voice, data, fax) and caller number. If the user is not authorised for AON, caller number is not available, or indication is not allowed, AON functionality is ignored. The ETSI version of FSK is used in most of Europe. In the USA, the Bellcore version dominates. Both versions support two modes of message transfer — on-hook and off-hook. The only difference between ETSI and Bellcore implementations is the frequency used for modulation.

FSK message transfer is usually associated with ringing. Data transmission can occur prior to ringing or during ringing. If the called party is already engaged in a conversation, off-hook FSK message transfer occurs (FSK message transfer not associated with ringing). Both modes of operation are illustrated with the practical example shown in Fig. 1.

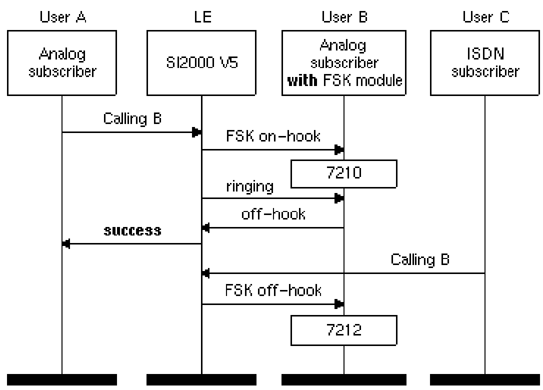

Fig. 1: Example of on-hook and off-hook FSK transmission modes.

Let us suppose that User B is subscribed to Call Waiting. When User A calls User B, transmission of FSK messages is associated with ringing. As shown in Fig. 1, the caller’s number (7210) is presented to User B after he/she receives “FSK on-hook” signal. When User C calls User B while he is talking to User A, off-hook FSK message transfer occurs. As shown in Fig. 1, the caller's number (7212) is presented to User B after he/she receives "FSK off-hook" signal. In this operation mode, messages are transported while User B's handset is off-hook. In this article we will focus only on on-hook FSK message transmission mode.

Section 2 gives an overview of the FSK protocol implementation. In section 3, detailed on-hook FSK specification is presented. Section 4 indicates how and where the service is implemented in SI2000 V5 telephone exchange system software. Final remarks about implementation issues are given in the conclusion.

2 FSK protocol implementation

Our implementation consists of Presentation layer and Data Link layer SDL (Specification and Description Language) program code. SDL is a standard formal language for the specification and description of real-time, concurrent, and heterogeneous systems [3]. Program code for Physical layer has been written in the C programming language.

The Presentation layer code generates Presentation layer messages (Fig. 2). They consist of a number of parameters. Each parameter contains the parameter type, parameter length, and parameter data octet(s). The parameter type is one octet long binary-encoded value. The specification defines the following parameter types:

- Date and Time (8 octets),
- Calling Line Identity Presentation (max. 20 octets),
- Caller Party Number (max. 20 octets),
- Reason for absence of Caller Line Identity (1 octet),
- Calling Party Name (max. 50 octets),
- Reason for absence of Caller Party Name (1 octet),
- Visual Indicator (1 octet),
- Complementary Calling Line Identity (max. 20 octets),
- Call Type (1 octet),
- First Called Line Identity (in case of forwarded call) (max. 20 octets),
- Network Message System Status (1 octet),
- Type of Forwarded Call (1 octet),
- Type of Calling User (1 octet),
- Redirecting Number (1 octet).

The Data Link layer program code generates Data Link layer messages. They consist of the channel seizure signal, mark signal, message type, and message length appended to a Presentation layer message (Fig. 3). They are also responsible for providing

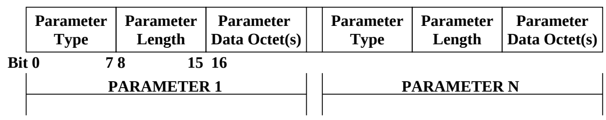

Fig. 2: Presentation layer message.

bit error detection capability. This protocol does not provide message retransmission. Therefore, it has no need for sequence number or acknowledgement generation.

Channel Seizure Signal	Mark Signal	Message Type	Message Lenght	Presentation Layer Message	Checksum

Fig. 3: Data Link layer message.

In more detail, a Data Link layer message includes the following fields:

- Channel seizure signal consists of 300 bits of alternating zeros and ones.
- Mark signal (180 (± 20) mark bits of alternating zeros and ones) is used for synchronisation.
- Message type (one octet) is a binary-encoded value. Call Set-up type is used most often.
- Message length (one octet) is a binary-encoded value. The message type, message length and checksum octets are not included in the message length.
- Checksum contains the two's complement of modulo 256 sum of all octets from the message type field to the end of Presentation layer message.

Physical layer is used for message submission to the telephone line. A start bit (zero) and a stop bit (one) are added to each Data Link message octet. The messages are sent serially. After the last bit of a Data Link message has been sent, the transmission should stop. Simplex asynchronous voiceband data transmission technique is used to transfer data to the telephone. A frequency modulator is required in the local exchange and a demodulator in the terminal equipment.

3 On-hook FSK message transmission

Transmission of FSK messages associated with ringing has two logical parts:

- data transmission prior to ringing,
- data transmission during ringing.

First, Terminal equipment Alert Signal (TAS) has to be sent. TAS alerts the telephone equipment that it will receive FSK data in the near future (45 – 500 ms). After successful FSK data transmission, the normal incoming call procedure is executed.

3.1 Terminal equipment Alert Signal

ETSI defines three types of TAS signals. We have tested only two of them:

- a Dual Tone Alerting Signal (DT-AS),
- a Ringing Pulse Alerting Signal (RP-AS).

Which one will be used is a network operator's choice, but the same method should also be used for data transmission not associated with ringing.

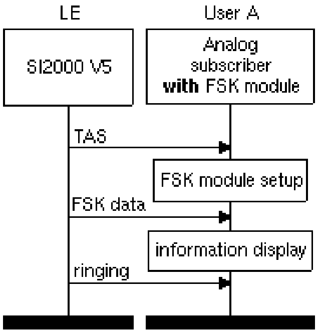

Fig. 4: Data transmission prior to ringing.

3.2 Data transmission prior to ringing

The signal flow for data transmission prior to ringing is shown in Fig. 4. DT-AS uses tones with the frequency of 2130 Hz and 2750 Hz. Signals should be 100 ms (±10 ms) long and the level of each component must not be higher that -15 dBm/tone (±2 dB). Time between the end of DT-AS and start of FSK data transmission should be between 45 and 500 ms. This assures sufficient time delay that the terminal equipment can prepare to receive FSK data. Ringing starts 200 to 500 ms after the FSK data transmission and normal incoming call procedure is conducted. If the telephone subscriber is not authorised for FSK, if the caller number is not present, or the indication is not allowed, the normal call procedure should be executed. If the called party answered the call before FSK data were sent, the connection is established without transmission of the FSK data.

The RP-AS is a short ring pulse on the telephone equipment. The duration of the RP-AS is between 200 and 300 ms. Signal levels are the same as specified for ringing. Time between the end of RP-AS and start of FSK data transmission must be between 500 and 800 ms. Ringing starts 200 to 500 ms after the FSK data transmission and the normal incoming call procedure is conducted.

3.3 Data transmission during ringing

With data transmission during ringing FSK transmission occurs during first silent period between two ring patterns as shown in Fig. 5. The first ring pattern represents TAS which alerts the terminal equipment that the local exchange has prepared FSK data. After a specified time interval that enables terminal equipment to prepare, the FSK data are sent. The normal incoming call procedure occurs after the FSK data reception.

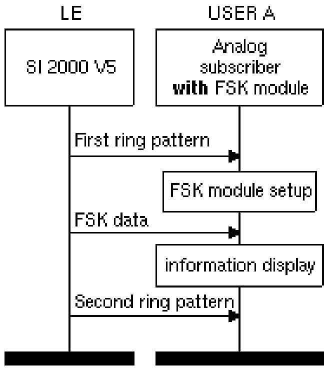

Fig. 5: Data transmission during ringing.

4 Service software implementation

The SI2000 V5 system software is divided into the CVA and CDA part. The hierarchical structure and communication paths between the two are shown in Fig. 6.

SCANDRV is an operator that scans the state of the analog telephone line. It is also used for ringing generation, polarisation turning, and tariff pulse generation. When a change is detected, a signal is sent to CASmux that prepares control signals and generates ASIO processes. Each telephone line has its own ASIO process. The ASIO process performs call control with the help of ASSC. On an ASIO request, the SAM manager generates an ASSC process in which the call control and data preparation, that are used later for message issues, are performed.

The UO processes are generated by the OMUX manager. It receives the data from the ASIO process and submits the data to the UO. UO's job is to control the line signals of CAS and ASS (Analog Subscriber Signalling). Each telephone line has its own UO process.

MFmux is a manager that issues requests for FSK generator seizure to TGCM. Its purpose is also to allocate the free dual tone generators. TGCM generates and controls a FSK_GEN process, which receives the data for submission and sends those with the determined time control to DSPDRV. If needed, FSK_GEN can generate commands for the DT_AS signal generation.

DSPDRV is a digital signalling processor driver that submits messages via the tone generators from the local exchange to the telephone equipment. It is a static process.

All described processes are static except for ASIO, ASSC, UO, and FSK_GEN that are dynamic. Majority of data flow goes through process managers (Fig. 6).

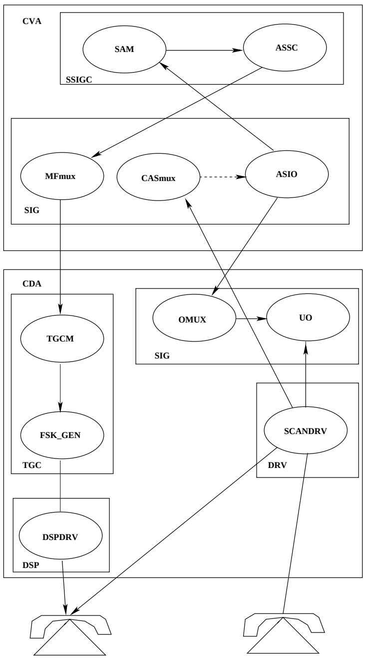

Fig. 6: Communication between CVA and CDA.

The data flow at the connection establishment is shown in Fig. 7. First, SCANDRV scans the telephone line. When it determines that the telephone receiver has been picked up, the data are submitted to CASmux. CASmux then generates ASIO, which requests ASSC generation using the SAM manager. ASSC then reads the subscriber data from the database and submits them through CASmux back to ASIO, which then checks the state of the called party with SCANDRV. If ASSC finds that the called party is authorised for FSK, it requests generation of FSK_GEN from MFmux. It submits this request to TGCM, which generates FSK_GEN. When FSK_GEN is created, an acknowledge signal is sent back to ASSC. Now ASSC sends the FSK data directly to FSK_GEN. FSK_GEN takes care of the time requests and sends the data to DSPDRV. It handles these data and submits them to the channel. Next, ASSC requests the generation of UO from OMUX. UO takes care of further call flow actions.

5 Conclusion

FSK is primarily used for caller identification, but can also be used for storage of missed calls and any other supplementary service based on caller identification. At this time, phones that do not support the FSK protocol present a serious limitation to wider deployment

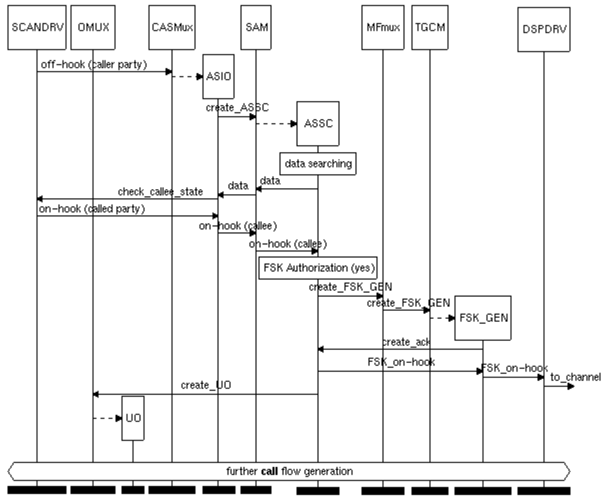

Fig. 7: Signal flow between CVA and CDA.

of this service. This can be solved with small specialised units that would be connected to the users' analog telephones.

When comparing the analog telephony and ISDN service "information display", we can observe the following distinctions:

- In the analog telephony, messages are sent by channel-associated signalling (CAS). With ISDN, they are sent via the D channel, thus using common-channel signalling (CCS).
- ISDN supports the indication of payment and call duration time. Analog telephones do not support that at this moment.

If we want to provide ISDN functionality of "information display" service to analog subscribers, the specification will have to be upgraded.

The FSK is implemented in the SI2000 V5 exchange system in product versions la5071 (access node), ls5071 (switch node), and higher. Our implementation of message indicating service is fully compliant with ETS 300 659-x [1,2] and is available in exchange system SI2000 V5.

References:

[1] ETS 300 659-1, "Subscriber line protocol over the local loop for display (and related) services" Part 1: On-hook data transmission, 1997

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[3] F. Belina, D. Hogrefe and A. Sarma, SDL with aplications from protocol specification, *Prentice Hall International*, 1991