

SCA-COMPLIANT PUBLIC SAFETY P25-FM3TR-VOIP BRIDGE

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ABSTRACT

Public safety protection and disaster relief operations require the coordination of multiple government civilian agencies, as well as military forces. This paper describes the design and implementation of a P25-FM3TR-VoIP bridge with SCA-compliant implementation of both the P25 and the FM3TR waveforms, on the SDR-4000 platform, from Spectrum Signal Processing. Two separate SDR-4000 units run the P25 and the FM3TR systems, respectively, in the bridge. The design of the bridge includes SIP/VoIP client/server architecture that has audio connections to and from both P25 and FM3TR. A push-to-talk control protocol has been designed, incorporating a client/server structure in the bridge to support push-to-talk operation. The lab demonstration setup of the P25-FM3TR-VoIP gateway is also described.

1. INTRODUCTION

Public safety protection and disaster relief operations require the coordination of multiple government civilian agencies, as well as military forces. The organization of these activities is most likely to be geographically distributed, as teams of rescuers spread out in different parts of a hot zone while the command center may be situated in a remote safe zone. Since different participants may use different communication systems, interoperability becomes one of the most critical requirements. A key technology to enable interoperability is a communication bridge that enables communication systems from various agencies to be internetworked into a heterogeneous network delivering real-time communication among multiple parties.

To this end, this paper discusses the design and implementation of an SCA-compatible voice communication bridge, which bridges real-time communications among three different voice systems, including the Project 25 (P25) public safety radio system, the future multiband multiwaveform modular tactical radio (FM3TR) system, and the voice over Internet Protocol (VoIP). The objective of this gateway

is to allow a user on a P25 handset to communicate with another user on an FM3TR system, and vice versa. In addition, both users can also communicate with VoIP clients located anywhere on the Internet. While there are existing products to bridge P25 with other narrowband voice systems, the key feature of the bridge described in this paper is the capability to bridge remote VoIP users, thus extending the interoperability with users on the Internet.

The P25 standard defines a public safety radio waveform system. It is a joint effort of the Association of Public Safety Communications Officials International (APCO), the National Association of State Telecommunications Directors (NASTD), the National Communications System (NCS) and selected federal agencies. It is standardized under the Telecommunications Industry Association (TIA) and has been adopted in North America, Australia and many other places around the world. The P25 standard specifies an open architecture for digital Land Mobile Radio (LMR) servicing multiple public safety organizations. Although the P25 technology is still in the process of being fully adopted and has yet to replace all the legacy analog-FM radio systems, it has been widely used by many federal, state and local civilian agencies [1].

The FM3TR was initially developed as an international cooperative effort between the United States, Germany, France and the United Kingdom to develop a reconfigurable communication system for ground and airborne applications. The Government Furnished Software (GFS) FM3TR source code is an implementation based on this international effort [2]. The FM3TR waveform implements frequency hopping over both VHF and UHF military bands (30 MHz - 400 MHz). The FM3TR waveform defines two operational modes: voice and data [3]. In the voice mode, the FM3TR uses continuously variable slope delta (CVSD) modulation for voice digitizing. The FM3TR waveform is an open standard and used as an instrument to promote international interoperability. It is also a reference waveform chosen by the SDR Forum as a test and demonstration vehicle. Hence, the FM3TR is representative of legacy military radio systems in the bridge design.

VoIP, which carries packetized voices over the IP network, is a widely-used networking application. By incorporating VoIP, the voice bridge presented here enables networking between legacy and modern communication systems. In addition, leveraging VoIP and broadband networking, the bridge not only connects two voice communication systems locally, but also expands the communication capability to remote clients.

To support VoIP, the session initiation protocol (SIP) is deployed in the bridge design. SIP, standardized by IETF [RFC 3261], is a signaling protocol widely used for controlling multimedia communication sessions such as voice and video calls over IP [4]. The protocol can be used for creating, modifying and terminating two-party or multiparty sessions consisting of one or several media streams. SIP is independent of the underlying transport layer. It can run on top of various transportation protocols, such as the transmission control protocol (TCP), user datagram protocol (UDP), or stream control transmission protocol (SCTP).

The key architecture of the voice bridge is built upon the client/server structure. One SIP/VoIP server and two SIP/VoIP clients are implemented in the bridge. One SIP/VoIP client serves the P25 channel, while the other is used for the FM3TR channel. Both P25 and FM3TR systems use push-to-talk (PTT) group communication mode. To support PTT operation with VoIP clients, a PTT control (PTTC) protocol was also designed and is discussed later in the paper. The client/server architecture based on the PTTC protocol is also implemented in the bridge in parallel with the SIP/VoIP client/server structure. Jointly, they enable PTT based communication over VoIP.

The software communication architecture (SCA) has been adopted by the Joint Tactical Radio System (JTRS) to define the common interfaces among waveform components, as well as increase software reuse and interoperability. The SCA-compatibility of the P25-FM3TR-VoIP bridge lies in the SCA-compatible implementation of both P25 and FM3TR waveforms on the SDR-4000, a surrogate JTRS software-defined radio (SDR) platform from Spectrum Signal Processing.

The rest of this paper is organized as follows. In Section 2, the SCA-compliant P25 and FM3TR implementation is described. The design and implementation of the P25-FM3TR-VoIP bridge are elaborated in Section 3. Discussions and conclusions follow in Section 4.

2. SCA-COMPLIANT P25 AND FM3TR SDR IMPLEMENTATIONS

Section 2 describes the SCA-compliant SDR implementation of the P25 and the FM3TR waveforms. Two SDR-4000 platforms are employed to implement these two waveforms. In the following subsection, the SDR-4000 platform and the

RF up/down conversion setup in the lab is introduced first, followed by the SCA architecture for FM3TR and P25 waveform implementations on SDR-4000.

2.1 SDR-4000 and RF up/down conversion

Each SDR-4000 IDS (Integrated Development System) unit includes the following hardware and software components.

The hardware includes a modem unit consisting of two boards interconnected in a mezzanine configuration (200 MB/s full-duplex over ePMC). One board is the PRO-4600 SDR; a heterogeneous processing engine employing a combination of a Xilinx Virtex-4 FPGA, a TMS320C6416T DSP and an MPC8541E GPP. The other board is the XMC-3321; a dual channel transceiver module that supports industry standard 10.7, 21.4 and 70 MHz IF frequencies through the use of dual 14-bit A/D converters sampling at up to 105 MSPS and dual 14-bit D/A converters sampling at up to 300 MSPS. A rear panel transition module (TM2-4900) provides connectivity to the PRO-4600 Gigabit Ethernet module, a high-speed serial, RS232 and JTAG interfaces. The chassis has five cPCI slots with front access for mounting modem board sets (PRO-4600 and XMC-3321) and rear access for mounting transition modules (TM2-4900).

The SDR-4000 uses the Green Hills INTEGRITY as the operation system for its GPP. The SCA core framework is provided by the SCARII++ from CRC and COBRA middleware is realized by ORBexpress from OIS. The quicComm software library, together with the PRO-4600 and XMC-3321 software APIs, all Spectrum Signal Processing products, abstracts the underlying hardware and provides users with basic link level access to communications between the processors in the SDR-4000. The software/firmware design tools include MULTI from Green Hills, Code Composer Studio from TI and Xilinx ISE.

The SDR-4000 outputs or accepts only IF signals. To perform IF-to-RF up/down conversion, National Instruments hardware is used, consisting of an up-converter (NI PXI-5610), down-converter (NI PXI-5600), and an RF preamplifier (NI PXI-5690) with both fixed (+30dB) and variable (-10 to +20dB) gains. A virtual Instrument (VI) program using LabView was developed that controls and configures both the PXI-5610 and PXI-5600 for desired RF operation.

2.2 SCA-compliant SDR P25 implementation

The implementation of one P25 radio on the SDR-4000 is illustrated in Fig.1. The setup includes:

- A headset and speakers connected to the VC-55 audio in/out connectors.

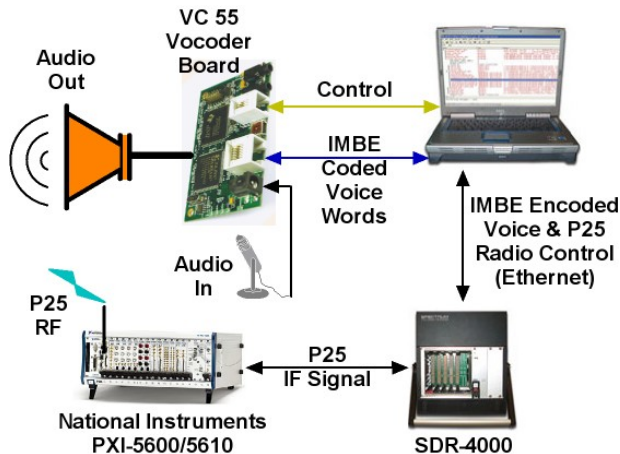


Figure 1: The SDR-4000 P25 implementation setup.

- The VC-55 vocoder from Digital Voice Systems, Inc. encodes the voice samples and send the voice code words over the RS-232 serial interface to the PC. Conversely, received voice code words from the PC are decoded and sent to the analog speaker output.
- The PC routes the encoded packets to/from the serial interface to the Ethernet interface to/from the SDR-4000. UDP/IP sockets are used to send data between the PC and the SDR-4000. The control of the P25 radio is managed from a PC application, which selects the radio frequency, assigns the group identification and/or the network identification, etc.
- The SDR-4000 applies FEC on link control words and encryption sync words, which then are encapsulated into the P25 CAI voice packet format. Moreover, C4FM modulation/demodulation as well as up/down conversion to/from analog IF is done in the SDR-4000.
- A National Instruments PXI-5600/5610 is used for RF up/down conversion.

2.2.1 P25 Packet Processing in the SCA Framework

The GPP of the SDR-4000 is responsible for P25 waveform packet processing. These processing functions are mapped into SCA resources running in the SCA core framework (CF). Fig. 2 gives an overview of P25 SCA resources. The packet control information resource is responsible for receiving/sending the control information, such as network access code and talk group ID, from/to the PTT application in the host PC. The GPP communicates with the PC over a local UDP/IP connection that carries both the control information and the voice code words to be transmitted, or received, by the unit. The OS based UDP/IP functions are encapsulated in an SCA Net Device resource. The FEC encoding and decoding of the different packet types take place in separate SCA resources.

When the “Talk” button on the PC PTT application is pressed, a packet is generated and the following voice code

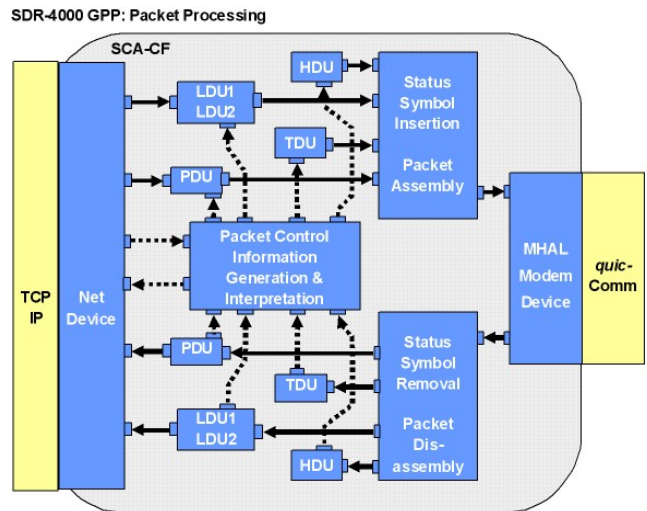


Figure 2: SDR-4000 GPP SCA framework for P25.

words are inserted into a flow of data units. A depressed “Talk” button triggers the end of the PTT session. A packet assembly function puts the stream of data units into a continuous bitstream, forming the P25 voice frame. The transmitted/received bit stream is sent through a modem hardware abstract layer (MHAL) device interfacing the SDR-4000 quicComm communication function set.

A packet disassemble function separates data units from the received bitstream and sends each data unit to its respective decoder function. The received voice code words are sent to PTT applications, over the UDP/IP interface.

The P25 waveform baseband processing is implemented in the DSP and FPGA of the SDR-4000.

2.3 SCA-compliant SDR FM3TR implementation

The FM3TR radio setup on SDR-4000 is similar to that of P25 in Fig. 1. The following SCA resources reside on the SDR-4000 GPP:

- **Net Device (Data/Voice):** the Net Device functions handle the platform specific transport of voice and data packets between the SDR-4000 and the application PC. The interface between the application PC and the SDR-4000 is TCP/IP/Ethernet based.
- **Data Link Control (DLC):** the DLC is an SCA resource that performs segmentation and reassembly of the data messages to and from the ITM application. The ARQ functionality also resides in the DLC resource.
- **Reed-Solomon (R-S) Codec:** the R-S Codec is an SCA resource that encodes outgoing data packets into an R-S block code and decodes received R-S encoded blocks. The system uses an RS (105, 72, 7) inner code and optionally an outer per-hop RS (16, 14, 5) code. The latter allows for an optional per hop ARQ scheme in full-duplex systems. This system will, however, be half duplex only.

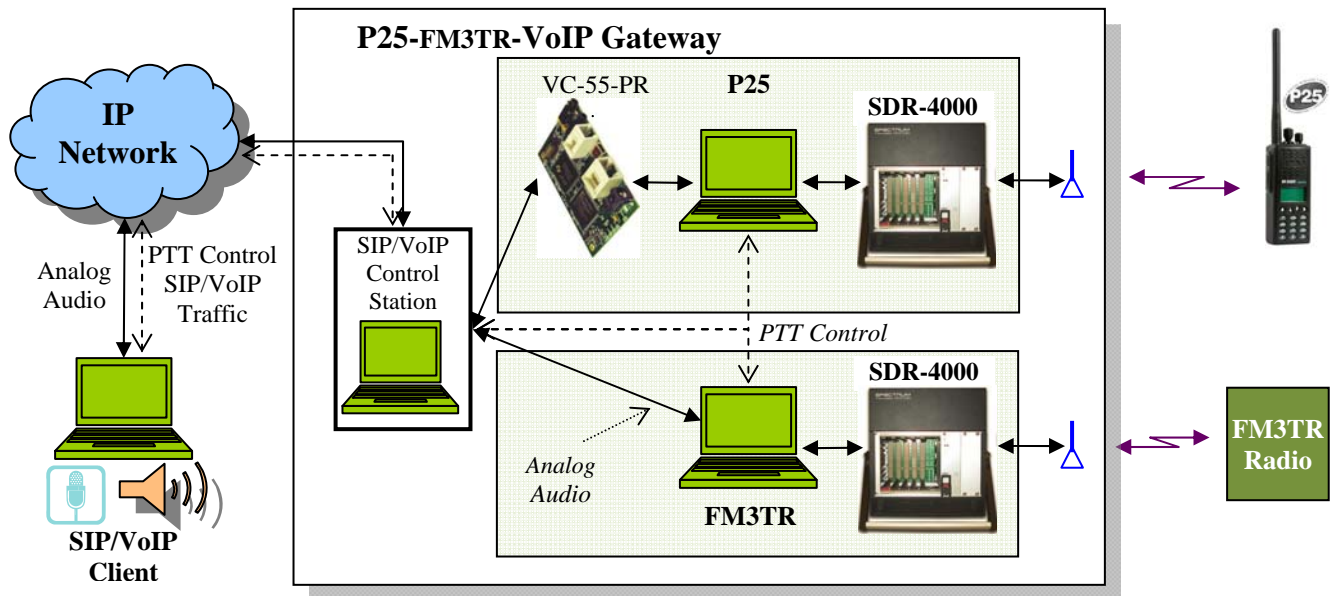


Figure 3: P25-FM3TR-VoIP gateway architecture.

- **Data MAC:** the Data MAC is an SCA resource that converts the format between data frames and data symbols grouped into 21 hops to match the R-S coding format. A total of 6 frames (24 data hops) are generated, including two end of message (EOM) frames that also carries data.
- **Voice MAC:** the Voice MAC is the original SCA resource ported from the base code. The CVSD encoded voice is sent/received on a per frame basis. At the end of a PTT session two EOM frames are sent out.
- **MHAL Modem Device:** an SCA compliant MHAL Modem Device resource encapsulates the voice and data packets (sent from the Voice MAC and Data MAC respectively) into the MHAL frame structure and sends the frames to a corresponding MHAL function in the DSP.
- **CVSD Codec:** the CVSD codec resource is ported directly from the FM3TR base code. The codec converts the 16 bit, 16 kHz PCM voice into 1-bit encoded voice at 16 kbps.

Other parts of the FM3TR waveform are implemented on the DSP and FPGA of SDR-4000, including modulation and demodulation of the FM3TR continuous phase frequency shift keying, correlation of synchronization sequence, frequency hopping, and digital up/down conversion. For more information, please refer to [5].

3. DESIGN OF P25-FM3TR-VOIP GATEWAY

In addition to interworking between a P25 system and an FM3TR system, the design target of the bridge also enables VoIP clients located anywhere on the Internet to communicate with P25 users or FM3TR users. For this purpose, the voice gateway adopts the SIP client/server

architecture. To simplify the implementation and ensure uniform operation, the bridging between the P25 and the FM3TR incorporates the SIP/VoIP architecture. Although the implemented P25-FM3TR-VoIP bridge only supports three clients, one for each system, the SIP/VoIP based bridge architecture allows scalability to bridge other systems, e.g., legacy analog FM land mobile radios (LMR).

Fig. 3 illustrates the overall architecture of the P25-FM3TR-VoIP bridge. The bridge consists of three components. Two separate SDR-4000 units run the P25 and the FM3TR systems, respectively, as the bridge interfaces with those two systems. One Linux based bridge PC hosts the SIP/VoIP server. The software architecture of the bridge PC is illustrated in Fig. 4 and explained in the following.

3.1 SIP/VoIP operation in the P25-FM3TR-VoIP bridge

Individual P25 and FM3TR devices are PTT based narrowband radio terminals for voice communications without IP support. Taking advantages of the fact that only one user can speak during the PTT network, the bridge regards the P25 channel as one VoIP client, regardless of how many P25 terminals. Similarly, the FM3TR channel is another VoIP client with multiple PTT radio terminals. To avoid modification to the SCA-compatible P25 and FM3TR waveform implementation on SDR-4000, the P25 and FM3TR VoIP clients are built into the bridge PC, instead of the SDR-4000, as shown in Fig 4.

Furthermore, since a P25 or FM3TR radio terminal cannot initiate a SIP conference, an operator is necessary at the bridge side. This role can be played by a human operator, or a piece of software. In the design described in

this paper, a human operator is required to load an appropriate SIP/VoIP conference profile in the SIP/VoIP server to activate the right conference mode. Given only three SIP clients in the bridge design of this paper, there are four profiles: 1) P25 and FM3TR, 2) P25 and remote VoIP client, 3) FM3TR and remote VoIP client, and 4) P25, FM3TR and remote VoIP client. While it is possible to have multiple conference profiles active at the same time, the current design only supports one active profile. After a profile is loaded into the SIP/VoIP server, a SIP/VoIP session is activated. The voice from one session participant can be heard by other SIP/VoIP clients of the same session.

The SIP/VoIP server can be configured to only handle the conference setup and maintenance, or force all calls to go through the server. In the former case, the SIP server will give the calling client the IP address and port of the callee so they can communicate directly. In the latter case, all clients get their own port on the server to send their audio data to. The SIP server handles the transferring of audio between the clients. The exception is multi-party conferencing; in which case there has to be a central audio server, where all the audio is mixed and routed. This can happen in the SIP server or the SIP server can allow another machine to handle the conference call mixing and send the mixed audio stream back to the server to be routed to all participants of the conference. Since the most commonly used profile is the P25, FM3TR and the remote VoIP client three-party conference in the design bridge, and both P25 and FM3TR SIP/VoIP clients are co-located in the same PC as the server, each client is forced to send audio data to the server, where the audio is mixed and sent to other clients in turn.

3.2 Support PTT operation for VoIP clients

Since both P25 and FM3TR use PTT voice communication, the receiving (ingress) system needs to invoke the transmitter on the egress system, for instance an incoming voice session on the P25 side of the bridge needs to initiate the transmitter on the FM3TR system and vice versa. To support PTT operation with the SIP/VoIP client/server architecture, a push-to-talk control (PTTC) protocol is designed for the P25-FM3TR-VoIP bridge [6].

The PTTC protocol is a data signaling protocol for managing PTT based voice sessions in a voice sharing environment. The protocol does not define how to set up the voice connection. It assumes that all the participants have been set up to use a common duplex voice channel through VoIP, audio cable, etc. The protocol uses a client/server model. Each PTT participant is represented by a client. A server grants voice channel access to the clients and notifies the clients of the channel state. Thus, a PTTC client/server system is implemented in the bridge PC in parallel with the SIP/VoIP client/server.

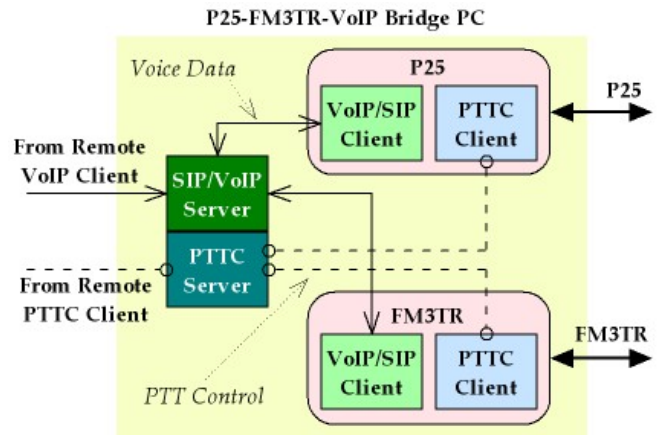


Figure 4: The software architecture of the bridge PC.

When a client joins the talking group, it must register with the PTTC server first. The server replies to the client with the current state of the voice channel. Before the client talks to the group, it must request the channel from the server. If the channel is already occupied, the server replies failure to the client, and the client should remain silent. If the channel is idle, the server notifies all the clients about the new state of the channel. The requesting client can start talking upon receiving this notification.

When a client finishes talking, it must notify the server to release the channel. The server then notifies all the clients about the new state of the channel. When the client leaves the group, it notifies the server. The server sends an acknowledge to the client.

Optionally, the client sends updates periodically to the server to maintain its state. In the server, updates are used in two cases. When the client is talking on the channel, it sends updates to keep its ownership of the channel. Otherwise, the client sends updates to validate its registration with the server. The server may use timeouts to invalidate inactive clients.

3.3 Interfacing SIP/VoIP and PTTC client

Based on above description and Fig. 4, each voice communication system (i.e., P25, FM3TR and remote VoIP) are represented by one SIP/VoIP client and one PTTC client.

The remote VoIP user has both clients installed on its terminal. To start talking, the remote PTTC client sends out a PTT request message to the PTTC server in the bridge PC. If the request is granted, the voice packet is then sent to the SIP/VoIP server.

For P25 and FM3TR, both clients are implemented in the bridge PC. When the corresponding SDR-4000 based radio terminal in the bridge detects a voice transmission in its radio channel, the SDR-4000 sends a message to the corresponding PTTC client in the bridge PC via Ethernet as

a virtual “push the talk button” action. Upon receipt of the message, the receiving PTTC client will send the PTT request message to the PTTC server.

Since the voice codecs used by P25 and FM3TR are different, combined with the fact that SIP/VoIP clients in the bridge PC only accept analog voice as the input, voice is handled differently for P25 and FM3TR. For P25, the VC-55 is used to decode the IMBE voice code. The P25 SDR-4000 outputs the voice data packets to the VC-55 directly. The analog voice output of VC-55 is fed into an audio card and the associated SIP/VoIP client on the bridge PC. For FM3TR, PCM coded voice data is sent from the FM3TR SDR-4000 directly to the control PC, playing out the voice as analog audio to the FM3TR SIP/VoIP PTTC client on the bridge PC.

3.4 Example of P25-FM3TR-VoIP bridge operation

To summarize the design details of this P25-FM3TR-VoIP bridge, an example conference session is described here to illustrate the bridge operation.

In this example, a remote VoIP client starts a conference call with both the P25 and FM3TR networks. The remote VoIP user informs the bridge to load the three-party profile in the bridge PC to setup the voice channel. This is accomplished either automatically or manually. For automatic SIP conference setup, there is one SIP number for each server profile. A server script loads the right conference profile, depending on which SIP number is dialed by the remote VoIP user. In the manual fashion, the bridge operator loads the right profile. After the conference voice channel is setup, each conference participant's PTTC client registers with the PTTC server, claiming the right to occupy the common voice channel. The conference can now start.

All participants follow the listen-then-talk rule during the conference, making sure that no other party is speaking before pushing the “talk button”. If the remote VoIP user wishes to speak, he/she needs to push the talk button on the GUI to claim the channel. This action triggers the PTTC client to send the channel request to the PTTC server. When the request is granted, the PTTC server informs all conference participants' PTTC clients. The P25 and FM3TR PTTC clients, after receiving the notification, inform the corresponding SDR-4000 in the bridge to turn-on the transmitter and send out voice data coming from the remote VoIP user. The P25 and FM3TR users in the field are now able to hear the voice.

The process is slightly different for the P25 or FM3TR user in the field. If a P25 user wishes to speak, he/she just follows the same routine as if there is no bridge – push the “talk button” on the P25 radio terminal. As soon as the button is pushed, the P25 SDR-4000 radio on the bridge side detects the transmission. The radio informs the P25 PTTC

client to request channel and also forward the voice to the P25 SIP/VoIP client. In other words, it is always the P25 or FM3TR radio at the bridge that interacts with the SIP/VoIP and PTTC clients.

4. DESIGN DISCUSSION

In this section, issues related with the P25-FM3TR-VoIP design, such as voice translation among different systems, collision avoidance and arbitration, will be discussed in detail. Possible future directions for research and development are also pointed in the end.

4.1 Voice translation in the P25-FM3TR-VoIP bridge

Digitized voices are often encoded using different types of vocoders in different system. So transcoders are usually a necessary component in a voice bridge, which connects the coded voice from one sub-system to another with a different voice codec. A transcoder convert voice in digital domain, thus only relies on software implementation and save hardware pieces.

The voice codec used in the FM3TR is 16 kbps continuous variable slope delta (CVSD) modulation. The VoIP users deploy G.711 μ -law for voice encoding and decoding. Both above voice codec are open source. However, the voice codec used in P25 is the Improved Multi-Band Excitation (IMBE), a proprietary vocoder developed by the Digital Voice Systems, Inc. (DVSI). Since it is prohibitively expensive for this project to access the internal details of the IMBE codec, it is difficult to develop a voice transcoder for the P25 system. So in this P25-FM3TR-VoIP bridge, voice from one sub-system is first converted into the analog form then digitized by the voice card of other two sub-systems. High quality sound cards are necessary to reduce the noise during this conversion.

On the other hand, it is interesting to note that, in a voice bridge involving n different voice codecs, if using pairwise voice transcoders, there will be $n(n-1)/2$ transcoders. However, if voices from different subsystems are first converted into one common format, only $n-1$ voice transcoders are needed, assuming the common format are also used by at least one subsystem. In the P25-FM3TR-VoIP bridge designed in this paper, the common format is, in fact, the analog voice waveform.

4.2 Collision resolution and arbitration

When multiple users press the talk button simultaneously, a collision occurs in the PTT system. Traditionally, collision avoidance and arbitration within a single group PTT system is solved by users following certain behavior protocols. For example, when a user stops talking, he/she will say “over”. Other users will talk over the channel only after hearing

“over”. PTT based voice communication systems have been in use by many professional groups over the past decades successfully.

However, when multiple PTT systems connected via a bridge, it is possible that users from different PTT subsystems may press the talk button simultaneously and the requests from multiple subsystems arrive in the bridge, causing a talking-request collision at the bridge.

The design of the PTTC protocol described in 3.2 also plays the role of collision resolution and arbitration. When multiple talking requests arriving from different subsystems, the PTTC clients of each subsystem will send out IP packets to the PTTC server to request channel. Although the talking requests arrive at the bridge simultaneously, the PTTC server will nevertheless receive the channel request packet from one subsystem earlier than other subsystems. The subsystem, whose request is handled by the PTTC server first, will get the channel assignment and the right to talk. The requests from other subsystems will be turned down by the PTTC server, and these subsystems will be in the listening mode. Admittedly, the user who has pressed the talking button on its PTT mobile terminal doesn't automatically know its request has been refused by the bridge. However, this is exactly the same as collision in a single PTT network operation. Without changing the existing standards, this type of collision is resolved through user behavior protocols.

4.3 Future research and development

A few R&D directions to further improve the capability of the P25-FM3TR-VoIP bridge are suggested in the following.

Data mode is defined in both P25 and FM3TR systems., However the data mode is not widely deployed. Data mode can be mixed with voice mode, such that the status of the bridge and the channel assignment can be fed back to mobile terminals during voice sessions to simplify the collision resolution and arbitration. Also, the bridge implemented the non-trunking mode of P25 system. How to bridge a P25 trunking system with other voice communication system is another interesting problem with practical significance.

In addition, only one active profile exists in the current SIP server in the P25-FM3TR-VoIP bridge. Multiple active profiles can support multiple user talking groups across several different subsystems. How to leverage SIP server's capability with legacy PTT systems' standard to support multiple simultaneous talking group across the bridge is another interesting direction.

5. SUMMARY

The P25-FM3TR-VoIP bridge described above works as a cross-connect for the P25 and FM3TR systems by setting up a VoIP call using SIP. Two SIP/VoIP clients handle the analog audio connection from the two bridged systems. As an extension, this bridge can connect any VoIP clients on the Internet to the users on the P25 and FM3TR systems.

A PTT control function resides in the designed bridge to handle PTT signals between the sending and receiving sides of the bridge. The PTT control function is communicating over TCP/IP with the two SDR4000 units. A generic VoIP client generates a PTT signal that will invoke the transmitters of both P25 and FM3TR radios of the bridge.

The client/server architecture of both SIP/VoIP and PTTC, adopted by the bridge design, is scalable to bridge communications among many different underlining systems.

6. ACKNOWLEDGMENT

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