♦ The Speech Transcoding Frame 2000 for GSM Cellular Systems

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The Global System for Mobile Communications (GSM) is the leading digital mobile telephone standard in the world. This paper describes the GSM cellular system, its features and services, and its system architecture. It also details the Speech Transcoding Frame (STF) 2000, one component of Lucent Technologies' GSM system. The STF 2000, which provides speech encoding and decoding (known collectively as "transcoding") and data rate adaptation, is cost effective, with a flexible, modular architecture that allows gradual system growth and enhancements with little or no downtime. It has the capability for "half-rate" speech coding, which enables network operators to double their over-the-air capacity without the need for increased radio-frequency (RF) spectrum. The system includes a notebook PC with a graphical user interface that simplifies installation, maintenance, and on-site diagnostics. Fault-recovery software enables the STF 2000 to recover automatically from internally and externally detected faults. Redundancy is built into critical system components so that standby or spare components can be switched into service when there is a fault in the active component. GSM features such as discontinuous transmission increase the system efficiency by decreasing radio interference and saving battery power at mobile stations. The STF 2000 meets European telecommunications standards.

The GSM Cellular System

The Global System for Mobile Communications (GSM),[1] which began as a European standard, is now gaining worldwide acceptance and market share, particularly in Asia and the Pacific region. Before 1982, Europe had more than a dozen incompatible analog cellular systems. Obviously, a common standard was needed, one that would allow international roaming and incorporate advanced services and features.

Development of the standard known as GSM began in 1982, and the first GSM systems were put into service in 1991. GSM includes standards for the 900-MHz band, the 1.8-GHz band—also called the Digital Cellular System (DCS) 1800—and the 1.9-GHz North American GSM band, formerly called Personal Communications Services (PCS) 1900. There are now more than 270 GSM networks in 98 countries. From 13 million in 1995, the number of subscribers is expected to grow to more than 100 million by the year 2000, representing over 25% of the world's cellular subscribers. GSM currently has 15 to 20% of the U.S. PCS market.

Features and Services

The GSM system was designed to be compatible with the integrated services digital network (ISDN) and offers a wide range of telecommunications services. Speech is the primary service, and the GSM system provides good subjective speech quality. Data services are now transmitted at rates as high as 9.6 kb/s, and higher rates are anticipated in the future. The GSM system offers other services, such as facsimile, emergency calls, alternate speech/data (the user can switch

Panel 1. Abbreviations, Acronyms, and Terms

ACELP—algebraic code-excited linear prediction
AUC—authentication center
Abis—interface between the BTS and BSC
BCE—base station control equipment
BSC—base station controller
BSS—base station system
BTS—base transceiver station
CELP—code-excited linear prediction
CEPT—Conference of European Postal and Telecommunications Administrations
codec—coder-decoder
CSPDN—circuit switched packet data network
DCS 1800—digital communication system operating at 1800 MHz
DFU—digital facilities unit
DSP—digital signal processor
EC—European Community
EFR—enhanced full rate
EIR—equipment identity register
ETSI—European Telecommunications Standards Institute
GMSK—Gaussian minimum shift keying
GSM—Global System for Mobile Communications
HLR—home location register
IMT—installation and maintenance terminal
ISDN—integrated services digital network
ITU—International Telecommunications Union
LAPD—link access procedure on the D channel
M—interface between the BCE and the STF 2000
MSC—mobile services switching center
O&M—operations and maintenance
OA&M—operations, administration and maintenance
OMC—operations and maintenance center
PCM—pulse code modulation
PCS—personal communications services
PSPDN—public switched packet data network
PSTN—public switched telephone network
RF—radio frequency
RPELTP—regular pulse excitation long-term prediction
SIM—subscriber identity module
SS7—Signaling System 7
STF—speech transcoding frame
STU—speech transcoding unit
STU H—speech transcoding unit–half rate
STU F—speech transcoding unit–full rate
TCP/IP—transmission control protocol/ Internet protocol
TCU—traffic service group control unit
TDM—time division multiplex
TDMA—time division multiple access
TRAU—transcoder rate adapter unit
TSG—traffic service group
VDE—Verband Deutscher Elektrotechniker (Association of German Electrical Engineers)
VLR—visitor location register
VSELP—vector-sum-excited linear prediction

from speech to data, or vice versa, without interrupting the call), alternate speech/facsimile, call barring and call forwarding, call waiting, call hold, and three-way calling.

Short Message Service, a unique service of GSM, resembles two-way paging. Users can send or receive messages as long as 160 characters. In addition, broadcast messages (such as traffic reports) can be sent to all mobile stations in an area. The message text is displayed on the mobile station's display panel.

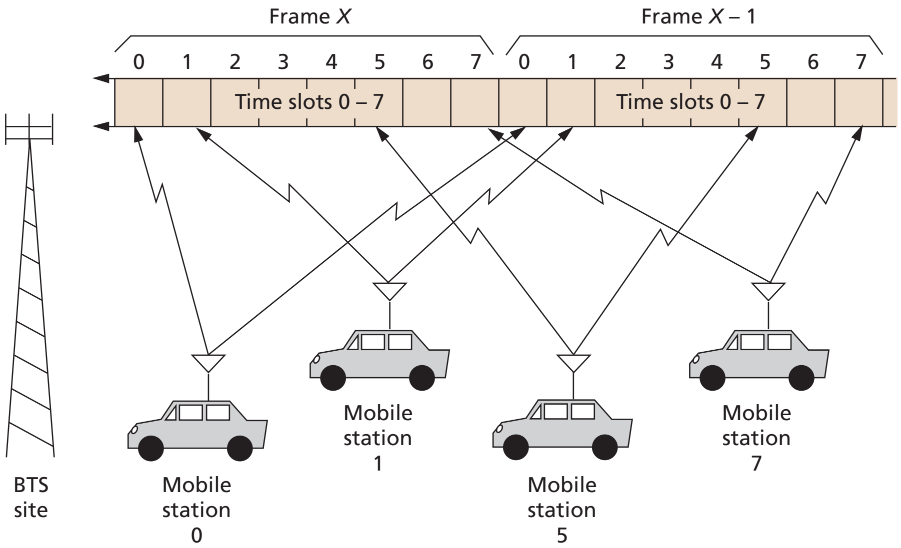
The subscriber identity module (SIM) extends the portability of GSM service. In contrast to most analog systems, which have subscriber information stored in permanent memory in the user's mobile station, the hardware of the GSM mobile station is anonymous, with subscriber-specific data programmed into a credit-card sized "smart card." (Smaller SIMs, called "plug-ins," are also available for hand-held mobile stations.) The SIM technology allows a user's subscription to be independent of a particular phone set. With a SIM card, a rented mobile station can receive calls made to the user's personal phone number, and all charges will show up on the user's bill. The SIM can also store phone numbers and dialing lists, as well as short messages (for example, messages received when the user is not present). Including user information in the SIM allows travelers to extend their mobile service to areas that use different air interfaces.

GSM System Architecture

The GSM air interface is a time division multiple access (TDMA) system, as shown in **Panel 2**. **Figure 1** shows the basic elements of the GSM system. The

Panel 2. The GSM Air Interface

TDMA transmission is in the form of a series of frames, each of which is divided into a number of slots, as shown in the figure below. Each slot position across frames is dedicated to a particular user. In GSM TDMA, the RF channel is partitioned into frames, with each frame in turn partitioned into eight non-overlapping time slots. Therefore, each RF channel supports eight full-rate traffic channels. Individual mobile stations take turns using the *uplink* channel (*from* the mobile subscriber) and may put a burst of data into the assigned time slot. In the *downlink* channel (*to* the mobile subscriber), the cell's BTS is usually transmitting continuously, with the mobile stations listening only during their assigned time slots.



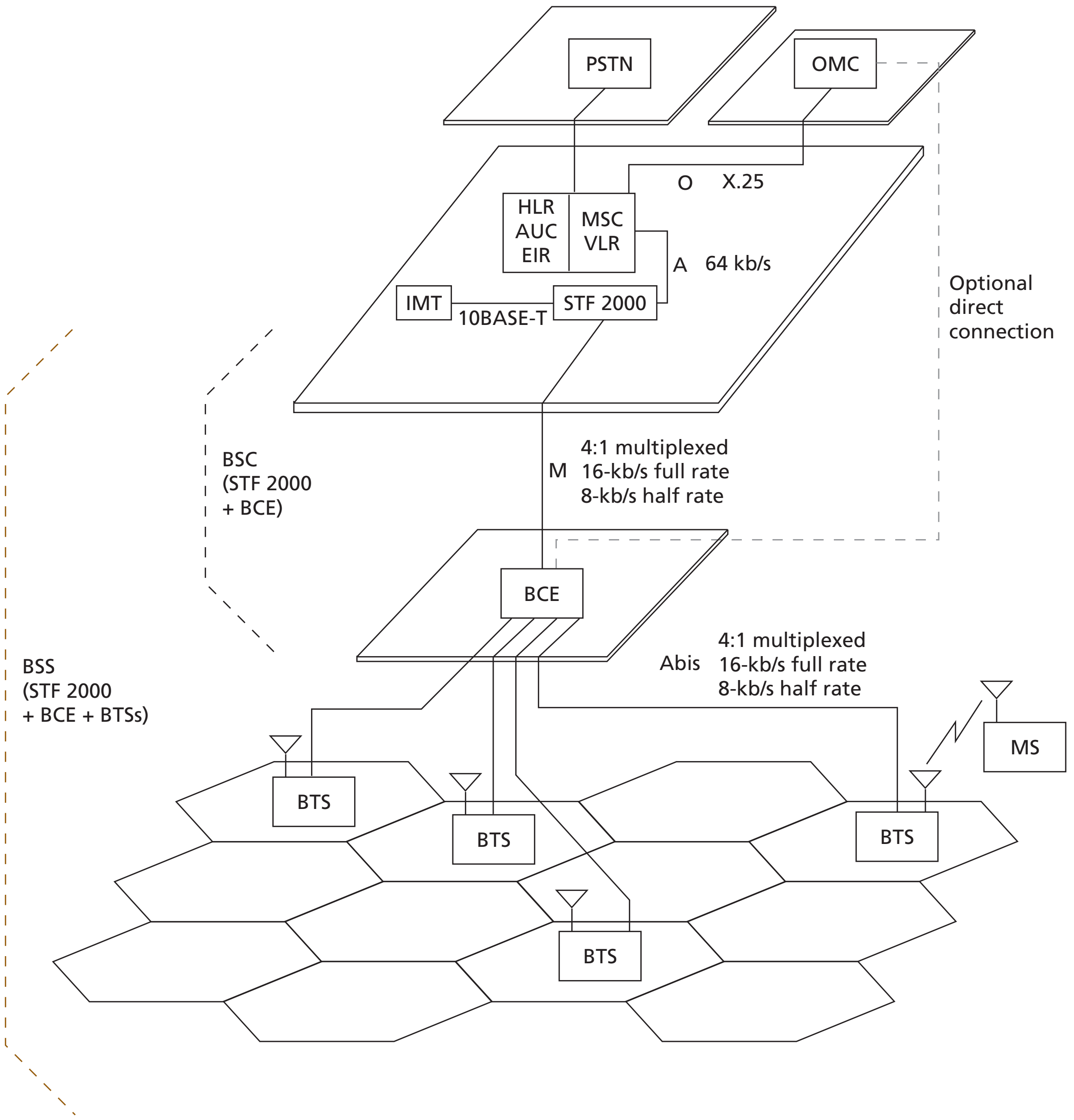
BTS – Base transceiver station

elements required for call processing are the mobile stations, the base station system (BSS), and the mobile services switching center (MSC). The MSC is associated with several subscriber database registers, including the visitor location register (VLR), home location register (HLR), authentication center (AUC), and equipment identity register (EIR). The operations and maintenance center (OMC), a separate centralized control center, monitors and controls system functions from one location.

The basic elements of the GSM system communicate by using the carrier facilities of the Conference of European Postal and Telecommunications Administrations (CEPT), which are leased from the local telephone company. The 2-Mb/s line interface (CEPT-1) is the lowest level of the hierarchy of the European E-carrier digital transmission facility, which multiplexes thirty-two 64-kb/s channels into a serial digital trunk (2,048 kb/s).

The MSC, which in Lucent Technologies is based on the 5ESS®-2000 switch, provides call setup, routing, and administrative functions. Its switching and interface equipment provides all the cellular network switching required to process calls to and from the mobile subscriber. The MSC is also the cellular interface into the public switched telephone network (PSTN). Calls from and to the PSTN and calls from one mobile subscriber to another pass through the MSC. Each MSC can service multiple BSSs.

The OMC, which is responsible for administration, maintenance, system change control, operations and performance management, and security management, can monitor and control several groups of BSSs simultaneously. Typically, the OMC is staffed 24 hours a day, 7 days a week, in contrast to the MSC and the BSSs, which are normally unattended.



A – Interface between the MSC and the STF 2000
Abis – Interface between the BSC and the BTS
AUC – Authentication center
BCE – Base station control equipment
BSC – Base station controller
BSS – Base station system
BTS – Base transceiver station
EIR – Equipment identity register
HLR – Home location register
IMT – Installation and maintenance terminal
M – Interface between the STF 2000 and the BCE
MS – Mobile station
MSC – Mobile services switching center
MUX – Multiplex
O – X.25 interface to the OMC
OMC – Operations and Maintenance Center
PSTN – Public switched telephone network
STF – Speech transcoding frame
VLR – Visitor location register

Figure 1.
Lucent Technologies' GSM system architecture.

The BSS consists of one or more base transceiver stations (BTSs) and a base station controller (BSC). The BTS is the radio element of the BSS, and the BSC is the control element.

The radio equipment that establishes the radio link to the mobile subscriber is within the BTS, which contains one or more transceivers (radios) and serves one cell. The BTS is responsible for communication with the BSC, radio frequency transmission and reception, and routine maintenance testing.

The BSC controls groups of BTSs and links the mobile subscriber, through the MSC, to the PSTN and other mobile subscribers. The BSC consists of the base station control equipment (BCE) and the transcoder rate adapter unit (TRAU). The Speech Transcoding Frame (STF) 2000 is Lucent Technologies' implementation of the TRAU.

The BCE is the central control element in the BSS. Based on over-the-air measurements received from the BTSs, the BCE administers the radio resources and radio frequencies, gathers data for network management, and controls the internal digital switch and the handover process. (As in all cellular systems, handovers are required to maintain calls in progress as mobile stations pass from one BTS coverage area, or cell, to another.) The internal digital switch is the heart of the BSS. The BCE provides a dynamic switching capability for routing individual traffic channels between the BTS and the MSC (via the TRAU). It also performs operations, administration, and maintenance (OA&M) functions for its associated BTSs.

The TRAU provides the speech transcoding, data rate adaptation, and submultiplexing functions. For speech signals, the transcoding function converts the standardized 64-kb/s data rate used on the network side of the system to the 13-kb/s (full-rate) or 5.6-kb/s (half-rate) signal transmitted across the air interface. (**Panel 3** describes the characteristics of half-rate channels.) For data signals, the TRAU adapts the data rate between the connection point to external networks and the intermediate transmission rate of the air interface. Lucent Technologies' STF 2000 also provides the submultiplexing function in which four transcoded or rate-adapted traffic channels on the BCE side are combined onto a single 64-kb/s time slot. Combining these traffic channels realizes a 4:1 savings in required transmission facilities from the TRAU to the BCE and BTSs. Although logically part of the BSS, the STF 2000 is located at the MSC site (a "Type 6" configuration in GSM terminology). Reducing the number of leased facilities from the MSC site to the BCE and BTSs represents a significant cost savings for the network operator.

Panel 3. Characteristics of Half-Rate Channels

A half-rate channel uses half the radio resources of a full-rate channel, which leads to a two-fold increase in spectrum efficiency. Each radio terminal supports the transmission and reception of a group of eight physical channels, which together form one TDMA frame on the air interface. With the introduction of half-rate speech coding, each physical channel can carry either one full-rate channel or two half-rate channels. Half-rate coding is achieved by transmitting/receiving in every other TDMA frame. Just as Lucent Technologies is actively developing half-rate speech coding for the GSM base station, other companies are actively developing half-rate speech coding for the GSM mobile station. The new mobile station, known as the dual-mode mobile, is expected to be available sometime in 1996. It will allow a call to be served on either a half-rate or full-rate channel, which increases the chances that the call will be served if no half-rate channels are available during setup or handover.

During call setup, the BCE performs the necessary switching to establish a traffic channel path between a transcoder at the TRAU and a radio terminal at the BTS. The transcoder and radio terminal use in-band signaling to communicate with one another, resulting in an overall traffic channel rate of 16 kb/s (full rate) or 8 kb/s (half rate).

A mobile station can either be hand held, mounted on a vehicle, or portable. Each mobile station contains a microprocessor that enables it to continually perform operations and communicate with the BTSs, even when it is not being used by the subscriber. Performing these functions keeps the mobile station ready to receive an incoming call or to make a call.

The interfaces between network elements MSC, OMC, and BSS are standard, so that elements from different vendors can be used in the same network. The A interface, which connects the BSS and the MSC, has thirty-one 64-kb/s communication channels that can accommodate a combination of traffic and Common Channel Signaling System 7 (SS7) signaling data. Each traffic or SS7 signaling channel has a per-channel bit rate of 64 kb/s. The Abis interface connects the BTS and the BSC. It has thirty-one 64-kb/s communication channels that can accommodate a combination of traffic and link access procedure on the D channel (LAPD) signaling channels. Each full-rate traffic channel has a per-channel bit rate of 16 kb/s, and each half-rate traffic channel has a per-channel bit rate of 8 kb/s. In Lucent Technologies' implementation, four traffic channels are multiplexed onto one 64-kb/s communication channel. A LAPD signaling channel has a per-channel bit rate of 64 kb/s. The O interface is a standard packet-switched network interface that connects the BCE and OMC, based on the X.25 interface specification of the ITU.

In addition, nonstandardized interfaces exist within network elements, such as Lucent Technologies' M interface between the BCE and the STF 2000 within the BSS. Each M interface has thirty-one 64-kb/s communication channels that can accommodate 120 traffic channels and one SS7 signaling channel. Each full-rate traffic channel has a per-channel bit rate of 16 kb/s, and each half-rate traffic channel has a per-channel bit rate of 8 kb/s. Four traffic channels are multiplexed onto one 64-kb/s communication channel. (Multiplexing four half-rate channels onto one 64-kb/s channel is called "simple" dual-rate coding. In a future enhancement of the STF 2000, eight half-rate channels will be multiplexed onto one 64-kb/s channel. Called "complex" dual-rate coding, this will require modifications to the other network elements, including the MSC.) An SS7 signaling channel has a per-channel bit rate of 64 kb/s and carries call processing messages between the MSC and the BCE through the STF 2000 without any intervention. A channel carrying SS7 signaling is a 64-kb/s clear channel, meaning that no portion of the channel is reserved for carrier framing or control bits. As such, the channel can carry other types of signaling channels (such as X.25) if the network provider so chooses.

The Speech Transcoding Frame 2000

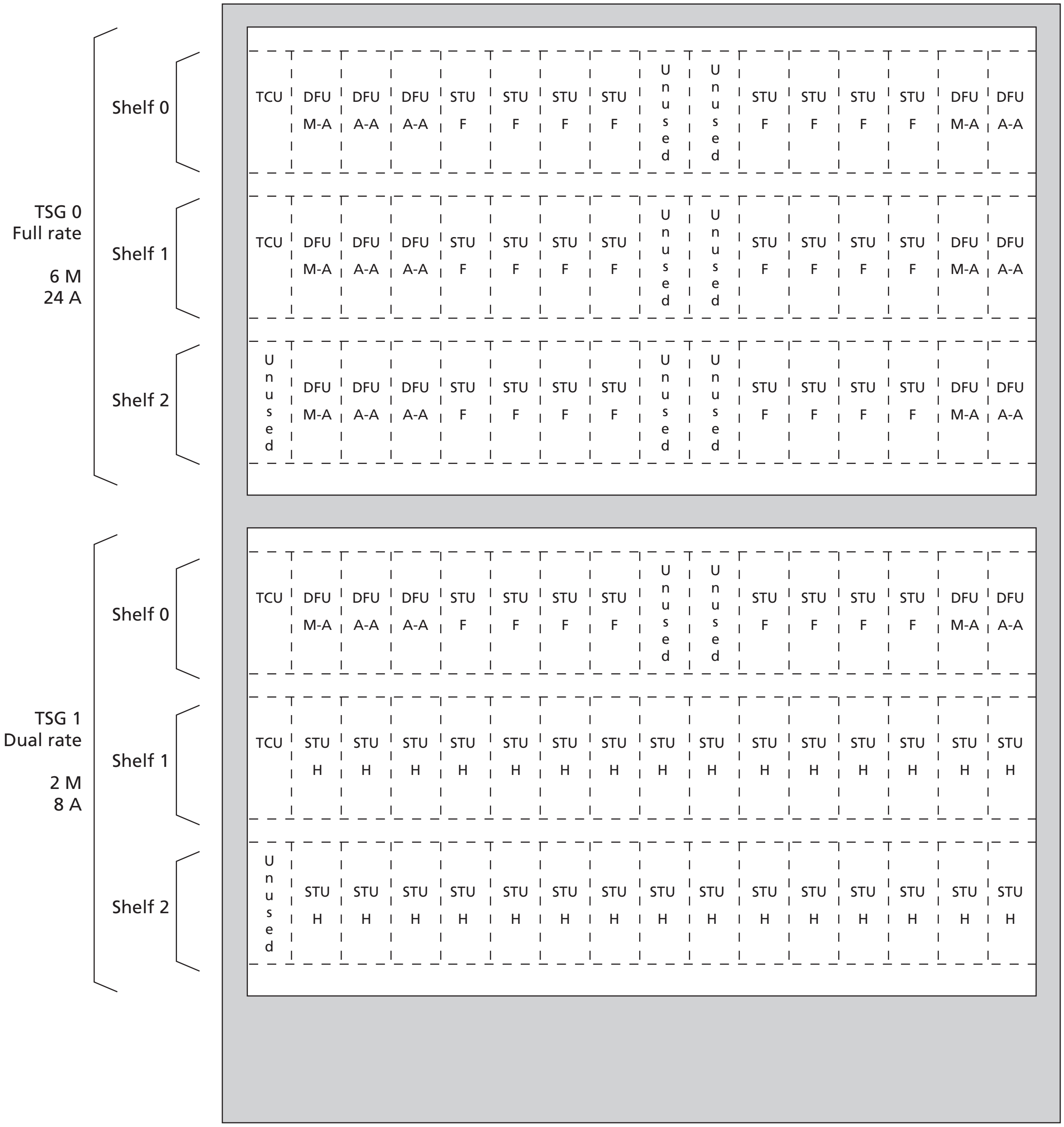
Lucent Technologies' STF 2000 provides state-of-the-art implementation of speech transcoding and data rate adaptation (TRAU) functions.

Architecture

The STF 2000, housed in a cabinet 600 mm wide x 600 mm deep x 2200 mm high (23.6 in x 23.6 in x 86.6 in), weighs approximately 270 kg (605 lb) fully equipped. As **Figure 2** shows, the STF 2000 consists of two traffic service groups (TSGs). Each TSG is an independent entity—there is no communication between them. All communication for a TSG is through the M interface and BCE to the radio devices in the BTSs. A TSG consists of three shelves, each able to support a total of 16 hardware units (plug-in circuit boards). The three types of circuit boards it supports are the digital facilities unit (DFU), the speech transcoding unit (STU), and the TSG control unit (TCU).

The DFU can physically terminate both the A and M interfaces. Each DFU can support two terminations—either one A interface and one M interface, or two A interfaces. The STU performs speech transcoding, with each STU capable of supporting as many as 32 full-rate traffic channels or, in a half-rate pool, 8 half-rate traffic channels. The TCU controls all functions, including supervision of the DFUs and STUs. TCUs operate in pairs to achieve redundancy, with one TCU active on-line, and the other in the standby mode.

A full-rate-only TSG contains 2 TCUs, as many as 15 DFUs (depending on the number of traffic channels required), and up to 24 STUs. A dual-rate (full-rate and half-rate) TSG contains 2 TCUs, up to 5 DFUs, and as many as 38 STUs. Figure 2 shows two types of DFUs (DFU M-A and DFU A-A) and two types of STUs—STU full rate (STU F) and STU half rate (STU H). Each type of board—whether type STU or DFU—is the same, but the function of each type is determined by its location in the shelf. No physical difference exists between a DFU M-A and a DFU A-A, or between an STU F and an STU H. A time division multiplexed (TDM) bus embedded in the backplane allows the



TSG 0 configured for full-rate operation only; TSG 1 configured for dual-rate operation.
Frame dimensions: 2200 mm (h) x 600 mm (w) x 600 mm (d)

DFU – Digital facilities unit (M-A = 1 M & 1 A interface connection, A-A = 2 A interface connections)
STU – Speech transcoding unit (F = full rate, H = half rate)
TCU – Traffic service group control unit
TSG – Traffic service group

Figure 2.
The STF 2000 frame.

boards to communicate with each other. The primary function of the TDM bus is to route user traffic between the CEPT facility lines and the coders. However, the TDM bus also passes control information between the TCU and the DFUs and STUs.

Capacity

The STF 2000 has four times more capacity than the full-rate-only TRAU equipment previously used in the Lucent Technologies' GSM system. An STF 2000 equipped for full-rate operation has a capacity of 1,440 traffic channels—180 radio-frequency (RF) channels. Because the half-rate coding algorithm is more complex than the full-rate algorithm, fewer channels can be processed on each STU, lowering the capacity of the dual-rate STF 2000 to 480 traffic channels (60 RF channels). The two TSGs in an STF 2000 frame are completely independent, allowing a frame to be configured with both full-rate and dual-rate TSGs. For example, one TSG could serve a part of the network where only full rate was required—this TSG could have as many as 6 M interfaces and 24 A interfaces. The other TSG in the same frame could serve part of the network where the need for higher over-the-air capacity made dual rate desirable. This TSG could have as many as 2 M interfaces and 8 A interfaces. An STF 2000 can serve one or more BCEs and therefore one or more BSSs.

Full-Rate Speech Coding

The GSM system uses a special digital speech-coding algorithm selected for its low bit rate (13 kb/s) and its resistance to high error-rate conditions. As with any cellular system, the prime concern of the speech transmission design was spectrum efficiency. To optimize the use of the radio spectrum, the goal was to produce a speech quality similar to that of the fixed telephone network, but with a much lower data rate.

Two very important elements in the design of any digital communication system are source encoding and channel encoding. Source encoding compresses the amount of information to be transmitted over a given channel. A reduction in data rate to the channel encoder allows the communication system to introduce more powerful encoding techniques to counter propagation and interference effects. The compressed data from the source encoder is applied to a channel encoder, which encodes and adds redundancy to the signal, in a form specifically suited to overcoming the detrimental effects of the transmission channel. Channel encoding consists of all the processes involved in conditioning the output of the source encoder prior to its transmission over the channel. Those processes include coding for forward error detection and correction, bit interleaving, and modulation. At the receiver site, the process is reversed—channel decoding followed by source decoding—to yield the original data.

For the Lucent Technologies' GSM system, the source encoding equipment is at the STF 2000, and the channel coding equipment is at the BTS. The two types of equipment communicate through traffic channel connections established over the M and Abis interfaces. More precisely, the communication is between the transcoders of the STUs and the channel coder-decoder (codec) units of the radio devices.

In the GSM system, the information source for speech is 64-kb/s A-law pulse code modulation (PCM). The source encoding in the downlink direction for full-rate speech consists of two steps:

1. The transcoder transforms the 8-bit A-law PCM into 13-bit linear PCM (104 kb/s).
2. The transcoder transforms the 13-bit linear PCM into a 13-kb/s net bit stream that is transmitted in 20-ms bursts, each containing 260 compressed speech bits.

The speech transcoder adds framing and control bits to form a 320-bit speech TRAU frame. In the uplink direction, the order is reversed. The speech codec mode described in Step 2 above is called regular pulse excitation long-term prediction (RPELTP).

In the GSM system, each channel (traffic, control) has its own coding and interleaving scheme. RF paths tend to give rise to both randomly distributed errors and bursts of data errors; variable redundancy coding and interleaved coding reduce the received error rate substantially. Full-rate traffic channels (speech or data) and most of the control channels use a common structure, which is a block of 456 coded bits, all interleaved and mapped onto bursts in a similar way. Transmitting 456 bits every 20 ms results in a transfer rate of 22.8 kb/s across the air interface. The modulation scheme is Gaussian minimum shift keying (GMSK), with a modulation rate of 270.833 kb/s.

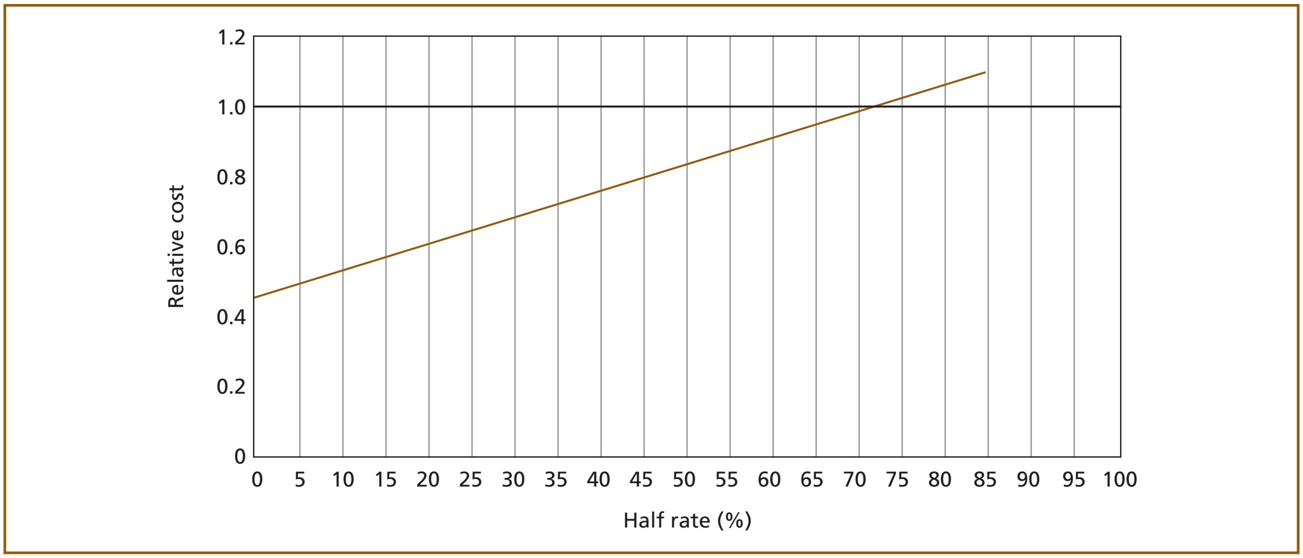


Figure 3.
Relative cost of STF 2000 with pooled half-rate transcoders versus all dual-rate transcoders.

Half-Rate Speech Coding

At the time that the original GSM speech algorithm was defined, it was foreseen that, as digital signal processor (DSP) technology evolved, it would be possible to accomplish the same end with half as many bits. Therefore, the GSM system was defined in two steps, starting with a less efficient 13-kb/s coding scheme, but allowing for a more efficient speech algorithm to be introduced at some time in the future. These half-rate specifications were finalized in 1995. Half-rate speech coding doubles the capacity of the air interface.

In a dual-rate STF 2000, each traffic channel may be either a 16-kb/s full-rate traffic channel or an 8-kb/s half-rate traffic channel. In the Lucent Technologies' GSM system, the half-rate traffic channel only transports digital speech; the half-rate user data feature has not been implemented as of this writing. The 240 half-rate transcoders—30 STU Hs, each of which can process 8 traffic channels—are available for connection in parallel with the 240 full-rate transcoders provided by the 8 STU Fs, each of which can process as many as 32 traffic channels.

Because the TCU manages and maintains the half-rate transcoder pool and assigns a half-rate transcoder to a call as needed, neither the BCE nor the MSC has to know which transcoders are configured as full rate and which as half rate. To the BCE and the MSC, all channels can be either full rate or half rate, and the STF 2000 appears as dual rate. Initially, any new call is handled by a full-rate transcoder. If the call is full rate, the full-rate transcoder processes that call. If the call is half rate, or was full rate but has changed to half rate, the TCU finds and connects the next available idle half-rate transcoder in parallel with the full-rate transcoder via time slot assignments on the internal TDM buses. Then, the half-rate transcoder—not the full-rate one—processes the call. If the call changes back to full rate, the TCU disconnects the half-rate transcoder and commands the full-rate transcoder to process the call once again.

This patented implementation of dual rate uses a pool of half-rate coders to gain a cost advantage over using dual-rate coders. When a network first introduces half rate, relatively few subscribers will have mobile stations that are capable of dual rate; therefore, few half-rate STUs are needed and the initial cost is low. For example, without a pool of half-rate coders, if 20% of the mobile stations are dual mode, 100% of the channels purchased by the operator have to be dual rate. With the pool, the operator can purchase 100% of channels capable of full rate and 30% of channels capable of half rate (extra available channels are required to achieve acceptably low call-blocking rates). **Figure 3** shows the relative cost of an STF 2000 with a pool of half-rate coders versus a 100% dual-rate STF. In the example described above, a 20% half-rate capability results in a 40% cost savings. Only at higher than 74% application of half rate does the cost of the half-rate pool exceed the 100% dual-rate case. In practical operation, the probability that all or even most channels in a system will be operated as half rate at the same time is extremely low.

To support this implementation of dual rate, the STF 2000 sends alarm messages when half-rate use reaches a programmable threshold. This allows operators to monitor their systems and add half-rate coders as needed. Half-rate STUs are added to the frame simply by plugging them into the appropriate slots. The TCU automatically detects their presence, downloads the software, configures them, and adds them to the half-rate pool.

For half-rate speech, the source encoding in the downlink direction consists of two steps:

1. The speech transcoder transforms the 8-bit A-law PCM into 13-bit linear PCM (104 kb/s).
2. The speech transcoder transforms the 13-bit linear PCM into a 5.6-kb/s net bit stream that is transmitted in 20-ms bursts, each containing 112 compressed speech bits.

The speech transcoder adds 48 framing and control bits to form a 160-bit speech TRAU frame. In the uplink direction, the order is reversed. The speech codec mode described in Step 2 above is called vector-sum-excited linear prediction (VSELP). The VSELP algorithm is an analysis-by-synthesis coding technique that belongs to the class of speech-coding algorithms known as code-excited linear prediction (CELP). The half-rate traffic channel for speech uses a 228 coding and interleaving scheme. Transmitting 228 bits every 20 ms results in a transfer rate of 11.4 kb/s across the air interface.

For half-rate coding to be successful in the marketplace, the speech quality has to be perceived as good by the end customer. Surveys of subjective speech quality have shown that, under typical conditions, the speech quality of half rate is comparable to that of full rate. When background noise is high, or when the call is processed through the half-rate codec twice (for example, a mobile-to-mobile call, both using half rate), users perceive a degradation in quality for the half-rate speech.

Enhanced Full-Rate Speech Coding

As advances in DSP technology made half-rate speech coding practical, they also enabled the development of an improved 13-kb/s speech-coding algorithm called enhanced full rate (EFR). The speech quality of EFR coders is significantly better than that of full rate coders, which is a competitive advantage for network operators. The coding scheme used is algebraic code-excited linear prediction (ACELP). EFR speech coding has been adopted for North American GSM, as well as for GSM 900/DCS 1800. In the near future, the STF 2000 will support the EFR codec with a pooled coder implementation similar to that used for half-rate coding.

Data Rate Adaptation

The STF 2000 performs data rate adaptation in accordance with the various speed classes of the data services, whose bit rates range between 0.6 and 9.6 kb/s. For data signals, a rate adaptation function at the STF 2000 adapts the standardized 64-kb/s transmission rate of the network (A interface) to the 12-, 6- or 3.6-kb/s transmission rate of the air interface. Essentially, the speech transcoding function at the transcoder is replaced with a data rate adaptation function. For data signals, the STF 2000 provides the appropriate data rate adaptation between the connection point to external networks—such as the circuit switched packet data network (CSPDN), public switched packet data network (PSPDN), or ISDN, respectively—and the air interface to the mobile station.

Software Architecture

The STF 2000 application software is implemented over the pSOS* System, an operating system licensed by Integrated Systems, Inc. Its software, which resides in the TCU board, consists of tasks that communicate with each other via pSOS+ queues and events. **Figure 4** shows the building blocks of the STF 2000 system software. The system tasks in the TCU application software include initialization, resource manager, download manager, connection manager, transparent manager, Installation and Maintenance Terminal (IMT) manager, alarm collect-

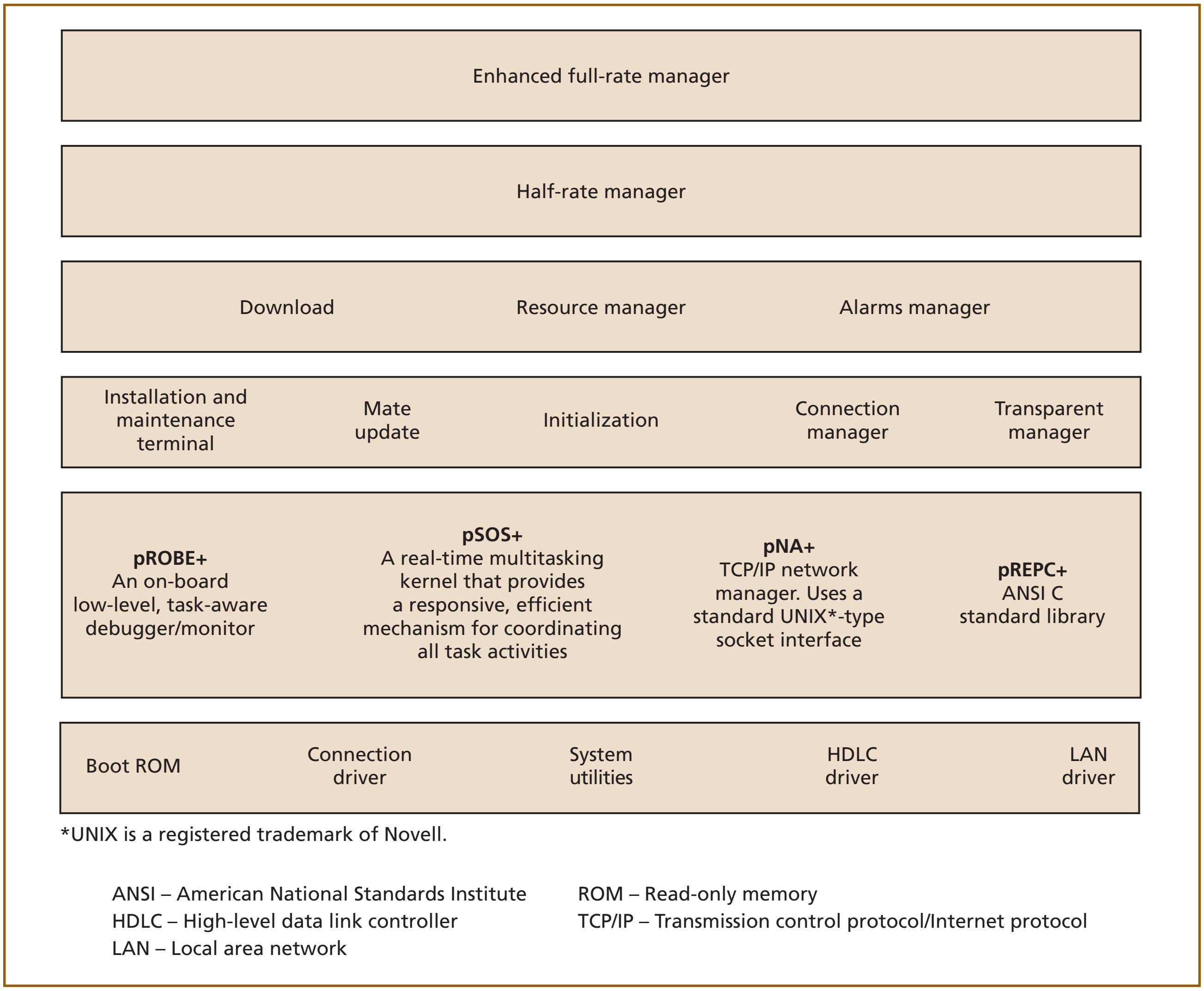


Figure 4.
STF 2000 software building blocks.

ing/reporting task, mate update task, half-rate manager, and enhanced full-rate manager.

Operations and Maintenance Signaling

Operations and maintenance (O&M) signaling between the STF 2000 and the BTS is performed by means of in-band signaling, which substitutes O&M TRAU frames for the speech or data TRAU frames of an established call. Except for the transcoder test messages that originate from the OMC, the O&M messages are alarms that indicate communication line conditions or failure of a hardware component. When the active TCU detects an error condition, it sends an alarm message to the BTS. The BTS sends the received information to the BCE via a LAPD signaling channel on the Abis interface. The BCE interprets the error and determines if it is necessary to send the information to the OMC. The technician at the OMC is notified of an error via an output report. After sending an alarm message, the TCU will also send a "Ceased" message when the error condition either clears itself or is cleared by STF 2000 maintenance personnel.

IMT

Used to manage the STF 2000, the IMT is a Windows*-based tool that runs on a notebook PC. Designed and engineered as a production product to be sold along with the STF 2000 equipment, the IMT enables the network operator to set and download configuration options, download operating software, initi-

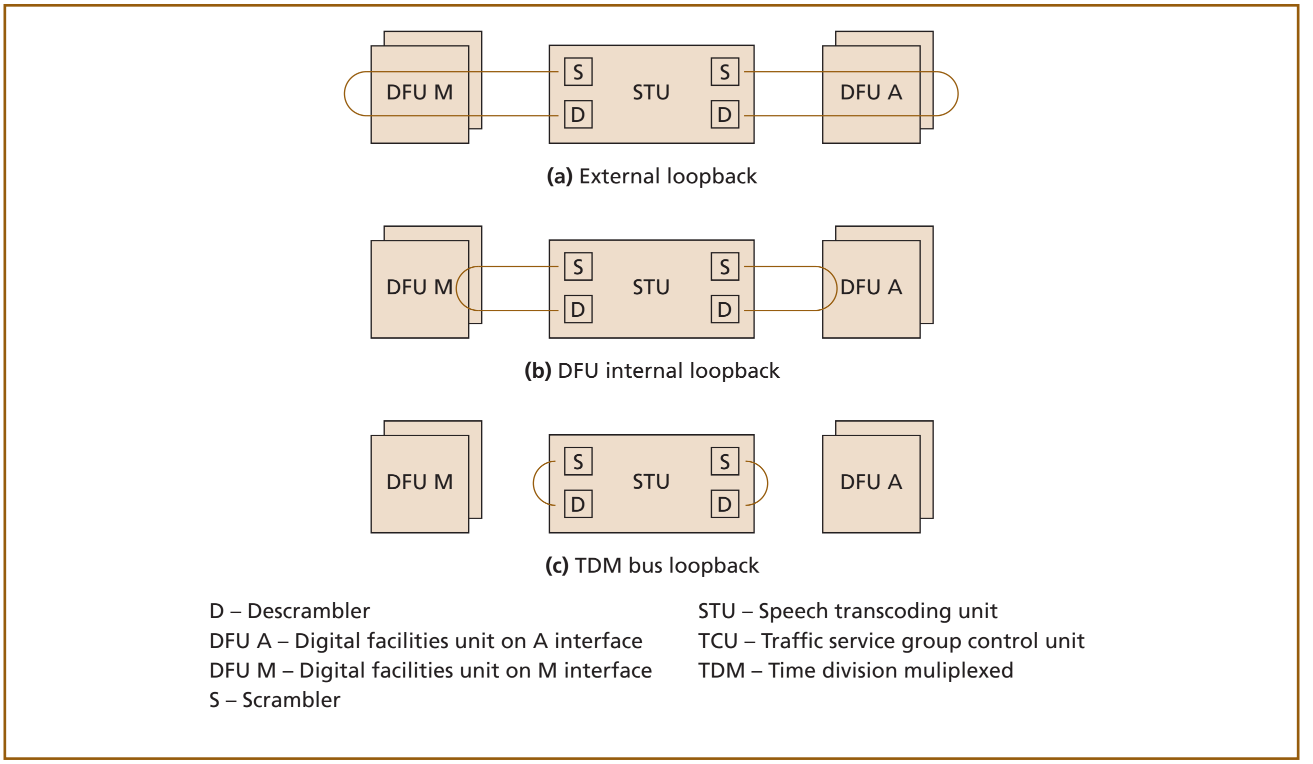


Figure 5.
Loopback test paths (a) external to the DFU, (b) internal to the DFU, and (c) internal to the STU.

ate loopback tests, and monitor the health and status of the STF 2000. The IMT uses the transmission control protocol/Internet protocol (TCP/IP) stack distributed with Windows 95.* The IMT connects to the TCU circuit board and can also be configured to connect to multiple TCUs simultaneously through a network concentrator or hub.

The IMT and the TCU have a client-server relationship. When an action is requested through the IMT application, the application sends the request to the IMT interpreter at the TCU. The interpreter translates the request and passes it along to the TCU main processor. It, in turn, performs the request and sends a reply message back through the interpreter to the IMT, where the reply message is displayed. During normal operation, the active TCU will send messages to both the IMT and the OMC. The IMT, in turn, will display and log all messages received from the TCU. The user can select the level of detail available in the IMT log.

The STF 2000 frame and all circuit boards can be tested by using the IMT to perform resident diagnostics, which consist of loopback tests that check all traffic time slot connections on the STU. These paths are the same ones used to carry traffic during normal operation. The STU sends a test pattern (scrambled data) on all M and A interface traffic paths. **Figure 5** shows the three points at which the test pattern can be looped back—at a point external to the DFU (known as external loopback), at a point within the DFU (known as DFU internal loopback), and at a point internal to the STU (known as TDM bus loopback). For the loopback test to pass, the STU must receive the same test pattern that it transmitted.

Reliability

The STF 2000 has fault-recovery software that allows it to recover automatically from internally and externally detected faults. No single failure of any active device causes the complete loss of frame functionality. This is achieved, in part, by building redundancy into critical system components. Standby or spare components are switched into service when there is a fault in the active component. Once the faulty component is out of service, it can be removed from the frame, repaired, and placed back into service when needed.

Standards Compliance

The STF 2000 complies with GSM technical specifications. It meets European standards for safety, electromagnetic compatibility, and electromagnetic interference and has been awarded European Community (EC) and Association of German Electrical Engineers (VDE) markings. It meets ETSI specifications for physical and environmental design, including the front access requirement. All external connections are made from the front so that the frame can be permanently installed against a wall or back to back with other frames, thereby saving valuable floor space.

Growth/Evolution Path

The STF 2000 is scaleable and therefore cost effective for small systems. Because circuit packs can be inserted and configured without service interruption, the system can be expanded easily. When network growth necessitates additional over-the-air capacity, full-rate frames can be converted with little difficulty to dual-rate operation without the need for additional radio terminals or allocated frequency spectrum.

The STF 2000 was designed to be converted to next-generation features with minimal or no hardware modifications. When features such as complex dual rate (8:1 multiplexing of half-rate channels in one 64-kb/s time slot), out-of-band LAPD control channels for alarming and OA&M activities, software/system configuration download from the OMC, and new speech-coding algorithms (such as EFR) are available in the other network elements, the STF 2000 can be reconfigured to support them via software upgrades.

Summary

The STF 2000 transcoding equipment provides a high-capacity equipment rack with transcoding capabilities for full- and half-rate traffic channels. Its flexible, modular architecture allows dual-rate operations transparent to the MSC and a fast, cost-effective transition to half-rate capability through a software upgrade. The STF 2000 supports a Type 6 BSS architecture and can be co-located with the MSC to achieve significant transmission facility savings. Installation, configuration, software download, and testing are performed with the IMT. The state-of-the-art technology of the STF 2000 provides a stable and robust design that is highly reliable.

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