SYSTEM 930
TELEPHONY SIMULATOR

System 930 T1 Version Reference Document

Covers topics including Robbed Bit, ISDN PRI, and GR-303

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The System 930 Table of Contents

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The document describes the System 930 menu system for T1 (1.544MHz) framing, version 1.48, dated 03-Feb-2003.

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Introduction to the System 930

The System 930 is a central office T1/E1 telephony simulator. The system provides two digital interfaces and up to 32 analog ports. The T1 version can be licensed for robbed bit signaling (RBS) both sides, ISDN network side, ISDN user side, GR-303 both sides, and NFAS (non-facility associated signaling). An E1 version is available as a separate product. Each interface is configured independent of the other, permitting both to be identical or different. Splitting interfaces between network side and user side allows calls to be sent from one interface to the other. Use the reversing data cable provided with the system to connect interface 1 to interface 2. This allows the functionality of the system to be explored using features such as the call generator and data analyzer. Splitting interfaces between ISDN and RBS creates an ISDN to RBS converter. RJ48 pins are wired for connection as the network side. Bantum jacks are also provided, wired in parallel with the RJ48 pins.

The System 930A is similar to the 930. Differences are interface wiring is configurable from screen 1, bantum jacks are eliminated, and the first eight analog ports are accessed through modular jacks mounted on the left side panel.

Communication to user screens is by RS-232: 9600 BPS, 8-N-1, XON/XOFF. VT-100 terminal emulation is used for display and keyboard operation. Configuration and status screens are selected from the main menu. Online help is available from all screens by pressing <F1>. Screens are quickly configured in real time and shown in real time. Once a screen has been selected, pressing <ESC> <F4> or <ESC> <F3> skips to the next or previous screen without going to the main menu.

The factory default condition can be set by typing FACTORY then <ENTER> from this help screen. Note that all saved system configuration profiles will be destroyed and restored to their factory default condition.

The System 930 may be licensed for NFAS (non-facility associated signaling) on ISDN and GR-303. This software version does not support D-channel backup on the secondary interface. It does however support D-channel service messages on the primary interface.

The System 930 Help Menu

Software Version History

This section provides an overview of the changes made for each software version. Please note that some listed feature sets are software license dependent and are not automatically included in software version upgrades.

Version 1.48 highlights: Screen 25: Fixed Called Mode to correctly display selections when spacebar is pressed.

Version 1.47 highlights: Screen 17: Fixed network side of ETSI E1 ISDN to send connect ack after receiving a quick connect without a channel ID.

Version 1.46 highlights: Screen 17: Added selection allow FXO/SAO signaling.

Version 1.45 highlights: Screen 23: Added variable audio threshold to S and V commands.

Version 1.44 highlights: Screen 6: Allow bracketed inbound match ranges to contain multiple digits.
Screen 8: Allow bracketed outbound prefix numbers to contain multiple digits.
Screen 25: Added forced disconnect time to 939 card general parameters.

Version 1.43 highlights: Added call waiting. Added Caller ID to GR-303.

Version 1.42 highlights: Added analog caller ID which is supported by the 936 card.
Screen 4: Added external flashhook (RBS & GR-303 only).

Version 1.40 highlights: Screen 18: Show CRV and CID in decimal instead of hex. Fixed GR-303 CRV to show the call reference divided by eight.

Version 1.39 highlights: Added support for Revision G processor card.

Version 1.38 highlights: Changed GR-303 Channel ID to start from interface 1.
Screen 21: Fixed bug that prevented call generator from working when maximum number of calls is blank on the Call Generator Configuration – Analog Ports screen.

Version 1.37 highlights: Changed Euro-ISDN overlap dialing to send the called number IE instead of keypad IE and concatenate digits on the receive side.
Screen 3: Added calling name to NI-2, 4ESS, and EURO-ISDN.
Screen 21: Added maximum number of calls to the call generator.

Version 1.36 highlights: Screen 1: Added GR-303 hybrid signaling.
Screen 3: Renamed caller to calling. Added calling name to 5ESS and DMS-100.
Screen 21: Added random called number groups to the call generator.

Version 1.35 highlights: Screen 25: Added new screen to configure the 939 Analog Card.

Version 1.34 highlights: Screen 3: Added random and sequential caller numbers.
Screen 17: Added send channel ID in first message only. Added increment after each call to cause code override.
Screen 24: Added new screen for time and date configuration.

Version 1.33 highlights: Screen 18: Increased data capture to 10 pages.
Screen 21: Fixed call generator to start correctly the first time a new software version is installed. Fixed internal call ID so that connected calls are not disconnected after 16384 additional calls.

Version 1.32 highlights: Screen 1: Added GR-303 CSC switch emulation to T1 version. Added QSIG switch emulation to E1 version. Added NFAS signaling to E1 and T1 versions. Added loop disconnect to T1 loop start FXO and SAO.
Screen 7: Allow en-bloc or overlap dialing when switch emulation is T1 RBS network or user side.
Screen 8: Added Z to allow match for no outbound prefix.
Screen 11: Added any digital interface or analog port to audio monitor source selection.
Screen 18: Added ISDN message types to data capture.
Screen 20: Added restart system to save/restore action.

Software Version History (continued)

Version 1.32 highlights, continued: Screen 22: Added connect action counters.
Screen 23: Added new screen for connect action configuration. Added ability to route incoming calls to connect action.

Version 1.31 highlights: Screen 6: Added Z to allow match for no inbound called number.

Version 1.30 highlights: Screen 2: Added new selection: Send D-channel status to T1 ISDN.
Screen 4: Added new selections: Receive loss and reference tone. This coincides with a change to 938 card firmware that inserts a –6dB loss in the audio level sent to analog ports required by V.90 (56k) analog modems.
Screen 7: Changed access digits to be flexible instead of hard coded to 8 and 9.

Version 1.29 highlights: Screen 7: Allow only en-bloc dialing when switch emulation is T1 RBS network side.
Screen 9: Added redirect ACD overflow action.
Screen 18: Enhanced call detail records for analog ports.
Former screen 19: Split call generator into screen 21 and 22.
Screen 21: Added running – until no connect to call generator mode.
Screen 22: Extended call counters to 32 bits.

Version 1.28 highlights: Fixed DTMF digit transfer from analog ports. Fixed bearer information element when redirecting from RBS to ISDN.
Screen 4: Added signaling method to individual analog ports.
Screen 18: Added call detail records for analog ports.

Version 1.27 highlights: Increased minimum time between analog port calls.

Version 1.26 highlights: Increased analog port on-hook debounce to 200ms.
Screen 21: Added call routing to analog port call generator.

Version 1.25 highlights: Added support for 932 processor card revision F. This card contains a battery powered clock and selectable A-law/mu-law audio monitoring and recorder/announcer. Software is backwards compatible with 932 processor card revision C.
Screen 2: Added ability to change ISDN channel service on the fly. Fixed ISDN channel disabled to place channel out of service upon layer 2 startup.
Screen 9: Changed least busy ACD hunt method to clockwise circular.
Screen 17: Added selectable exclusive / preferred channel ID.
Screen 18: Added current year, month, and time zone.

Version 1.24 highlights: Fixed intermittent response to ISDN messages. Fixed system reset due to unacknowledged service messages. Fixed ACD queuing problem due to version 1.17 changes. Improved call generator burst mode.
Screen 4: Added configurable alerting time. Allow analog extensions to begin with 7.
Screen 7: Changed explicit channel access digit from 7 to *.
Screen 18: Added text to show ISDN transfer capability after bearer capability information elements.
Added text to show ISDN channel number after channel ID information elements.

Version 1.23 highlights: Screen 11: Improved audio monitor.

Version 1.22 highlights: Fixed multirate to connect channels when bearer capability information element length is 2.
Screen 3: Allow call generator to overlay caller number.
Screen 10: Improved redirect by copying display, transit network selection, low layer compatibility, high layer compatibility, and user-user ISDN information elements from incoming to outgoing call.
Screen 11: Improved fixed channel or port on audio monitor.
Screen 12: Added configurable password reminder.
Screen 17: Added configurable multirate slot assignment.
Screen 18: Add milliseconds to ISDN data capture timestamp. Show call state name in RBS data capture.

Software Version History (continued)

Version 1.22 highlights, continued: Screen 21: Allow sequential DTMF numbers in the call generator connect action.
Screen 22: Save call generator historical data during power loss.

Version 1.20 highlights: Improved analog disconnect synchronization with 938 card.

Version 1.19 highlights: Screen 2: Fixed RBS exclusive channel operation and added it to T1 ISDN.

Version 1.18 highlights: Fixed T1 RBS loop start signaling. Fixed user side 5ESS custom to accept unknown type of number. Added sending complete information element to Euro ISDN setup messages during en-bloc dialing. Fixed numerous problems in redirect.
Screen 2: Added selection to send T1 ISDN service messages upon establishment of layer two.
Screen 6: Added selection to route incoming calls to voice message play continuously.
Screen 7: Added selection for whether analog ports use en-bloc or overlap dialing.
Screen 12: Added selection for automatic startup of ISDN layer two. Added selection for ISDN channel ID format. This allows the system to work with products that accept number format only.
Screen 18: Decode IA5 characters such as called number instead of showing them in hex on data capture screen. Fixed display of single byte information elements such as sending complete.
Screen 22: Added incoming message count to call counter screen.

Version 1.17 highlights: Changed redirect to echo incoming bearer capability information element rather than use screen 17 settings.
Screen 17: Added setup bearer rate for testing multirate calls. Accidentally changed channel ID format in ISDN proceeding messages to send slot map format. Added selection to use last 3 digits of called number for testing cause codes.

Version 1.15 highlights: Screen 17: Added selection do not send channel ID. Fixed network side Euro ISDN to accept messages without channel ID. Added selection do not send ISDN proceeding message and do not send ISDN alerting message to test ISDN quick connect.

Version 1.12 highlights: Combined T1 ISDN and T1 RBS into a single product.

Version 1.10 highlights: First release of Euro ISDN. Updated tone ROM.

Abbreviations and Terminology

ACD: Automatic Call Distribution. Routes incoming calls to analog ports as required. The system provides 32 ACD groups, one for each analog port.

Bearer Channel: Contains digital information for delivery to the end user.

Blue Alarm: Pattern sent by T1 carrier to indicate an idle line. The pattern is all ones, including the framing bit. Also known as Alarm Indication Signal (AIS).

Bonding: Pre-ISDN feature that combines multiple calls to increase data transfer rate. Receiver must adjust for multipath delays. Also see multirate.

CAS: Channel Associated Signaling. See E1 CAS.

CPE: Customer Premise Equipment. Equipment connected to network.

Call by call: Calls are routed by their called number, not the channel they come in on.

DCN: Digital Channel Number.

DNIS: Dialed Number Inbound Service. Inbound calls are routed according to the called number.

DTMF: Dual Tone Multiple Frequency. Tones used for dialing numbers. Also known as touch tones.

Data Channel: Contains signaling information.

E1: Trunk operating at 2.048 MBPS. E1 contains 30 bearer channels, one data channel, and one framing channel.

Abbreviations and Terminology (continued)

E1 CAS: E1 Channel Associated Signaling. The data channel contains a set of signaling bits associated with specific bearer channels. DNIS is accomplishing using in-band MF (multi frequency) tones. E1 CAS is not supported by this system.

E1 ISDN: E1 Integrated Services Digital Network. The data channel contains ISDN signaling.

Euro ISDN: European ISDN defined by European Telecommunications Standards Institute (ETSI). Also known as NET5 and V5. Besides allowing 30 bearer channels, there are some subtle differences between Euro ISDN and its North American counterpart. Euro ISDN does not require the user to send a channel ID information element, giving network side full control of channel selection and eliminating glare. Channels are always enabled. Overlap dialing is allowed in the user to network direction, eliminating outbound number configuration on the user side. Provides a sending complete information element that indicates the called number is complete or there is no called number. Does not allow NFAS (non-facility associated signaling) and hence eliminates the need for backup D-channel signaling. Euro ISDN sends calling name using the Display IE where as North American NI-2 did not support calling name and NI-3 uses the Facility IE.

En-bloc: The called number is sent all at once as opposed to overlap dialing which sends the digits one at a time. Dial tone is supplied locally.

FSK: Frequency Shift Keying. A modulation technique used for sending in-band Caller ID and Message Waiting.

Glare: User and network sides simultaneously seize the same channel, creating a conflict.

Layer 2: Provides error-free communication between the network and user. Also known as the data link layer. Layer 2 down indicates that communication has not be established.

Multirate: ISDN feature that uses multiple channels during a single call to increase data transfer rate. This system supports multirate calls through the redirect facility. Also see bonding.

NFAS: Non-facility associated signaling. ISDN feature that allows one D-channel to provide signaling for multiple DS1s.

Overlap: The called number is sent one digit at a time as opposed to en-bloc dialing which sends the digits all at once. Dial tone is supplied by the near end.

PRI: Primary Rate Interface. ISDN over T1 or E1.

Pulse Dialing: Dialing accomplished by breaking of DC current, or the analog or digital equivalent. Pulse dialing is not supported by this system.

QSIG: Switch emulation for linking PBXs on a private network. Similar to Euro ISDN.

RBS: Robbed Bit Signaling. See T1 RBS.

Switch Hook Flash: A method for changing the call state by depressing the receiver hook for one second. Switch hook flash is supported by this system.

T1: Trunk operating at 1.544 MBPS. Contains 24 channels.

T1 ISDN: T1 Integrated Services Digital Network. T1 trunk containing 23 bearer channels and one ISDN signaling channel.

T1 RBS: T1 Robbed Bit Signaling. T1 trunk containing 24 bearer channels and no data channels. Signaling is accomplished by stealing one bit from each bearer channel.

Yellow Alarm: Pattern sent by T1 carrier to indicate loss of frame synchronization.

The System 930 Help Menu

Standards

The signaling standards used by this system are: Layer-1: ITU-T G.703, G.704, G.706. 4ESS Custom: AT&T TR 41459 ISDN PRI Interface Specification.

5ESS Custom: Lucent 235-900-332 and Lucent 235-900-342 ISDN PRI Specification. DMS100 Custom: Nortel NIS A211-1 ISDN PRI Access User-Network Interface Specification. GR-303: Telcordia GR-303-CORE Integrated Digital Loop Carrier System and Interface. NI-2: ITU-T Q.921 for layer 2 and Q.931 for layer 3. RBS: EIA 464B PBX Switching Equipment for Voiceband Application.

VT-100 Display Standards: The VT-100 display codes used by this system are ^[[2J for clear screen, ^[[0m for normal video, ^[[7m for reverse video, and ^[[row;columnf for row and column screen position. ^[represents ASCII ESC.

VT-100 Keyboard Standards: The VT-100 keyboard codes used by this system are ^[OP for F1, ^[OQ for F2, ^[OR for F3, ^[OS for F4, ^[[A for up, ^[[B for down, ^[[C for right, and ^[[D for left. ^[represents ASCII ESC.

Robbed Bit Signaling

There are three types of RBS protocols: E&M, ground start, and loop start. E&M is the preferred method for PBXs. Loop start is what a plain old telephone (POTS) uses. The following tutorial describes the differences between each.

E&M Signaling

E&M stands for ear and mouth, as one side's ear is the other's mouth. Network and user side are symmetrical. DID (direct inward dial), DOD (direct outward dial), and Tie Trunk utilize forms of E&M signaling. The A and B bits are identical and represent status: on and off.

E&M Signaling: User Initiated Call

	User to Network (User Output)		Network to User (Network Output)	
	A	B	A	B
Idle	0	0	0	0
User off hook	1	1-SETUP	0	0
If wink start is in effect:				
Wink setup (1)	1	1	1	1-RESERVED
Wink on (2)	1	1	1	1-WINK
End of wink	1	1	0	0-SETUP ACK
Dialing follows				
Await answer	1	1	0	0
Network answer (3)	1	1	1	1-CONNECT
Network disconnect (4)	1	1	0	0-DISC
User hangup (4)	0	0-DISC	0	0

Note: (1) Wink setup time is normally 100ms.
Note: (2) Wink duration is normally 200ms.
Note: (3) Start billing.
Note: (4) User hangup may come before or after network disconnect.

E&M Signaling: Network Initiated Call

	User to Network (User Output)		Network to User (Network Output)	
	A	B	A	B
Idle	0	0	0	0
Network seizure	0	0	1	1-SETUP
If wink start is in effect:				
Wink setup (1)	1	1-RESERVED	1	1
Wink on (2)	1	1-WINK	1	1
End of wink	0	0-SETUP ACK	1	1
Dialing follows				
Await answer	0	0	1	1
User answer (3)	1	1-CONNECT	1	1
User hangup (4)	0	0-DISC	1	1
Network disconnect (4)	0	0	0	0-DISC

Note: (1) Wink setup time is normally 100ms.
Note: (2) Wink duration is normally 200ms.
Note: (3) Start billing.
Note: (4) User hangup may come before or after network disconnect.

Robbed Bit Signaling (continued)

Ground Start Signaling

Ground start is very complex. Network and user side are not symmetrical. Status information is carried in both directions. Ring request information is carried in one direction. User A bit represents loop current (LC) used to detect off-hook. User B bit represents ring (outer) conductor. Network A bit represents tip (inner) conductor. Network B bit represents alerting current (AC) used to ring a bell. The ring conductor is the outer conductor of a bantum plug. It has nothing to do with alerting (ringing) a phone and is a source of confusion.

There are four types of ground start signaling:
FXS: Foreign Exchange Subscriber. What the user side sends to FXO.
FXO: Foreign Exchange Office. What the network side sends to FXS.
SAS: Special Access Subscriber. What the user side sends to SAO.
SAO: Special Access Office. What the network side sends to SAS.

A foreign exchange is a remotely located exchange that the subscriber is connected to. This normally will be a central office. Special access is a dedicated line through the central office connected directly to a long distance carrier. Note that FXO/SAO and FXS/SAS are similar except that some bits are inverted.

Ground Start Signaling: User Initiated Call

	User to Network (User Output) FXS/SAS		Network to User (Network Output) FXO/SAO	
	A (LC)	B (ring)	A (tip)	B (AC)
Idle	0	1/0	1/0	1
User grounds ring	0	0/1-SETUP	1/0	1
Network grounds tip (1)	0	0/1	0/1	1-SETUP ACK
User closes loop	1	1/0-CONNECT	0/1	1
Dialing follows				
Connected state	1	1/0	0/1	1
User hangup (2)	0	1/0-DISC	0/1	1
Network disconnect (2)	0	1/0	1/0	1-DISC

Note: (1) User should close loop within 50msec of network grounding tip.
Note: (2) User hangup may come before or after network disconnect.

Ground Start Signaling: Network Initiated Call

	User to Network (User Output) FXS/SAS		Network to User (Network Output) FXO/SAO	
	A (LC)	B (ring)	A (tip)	B (AC)
Idle	0	1/0	1/0	1
Network grounds tip	0	1/0	0/1	1-RESERVED
User alerted (1)	0	1/0	0/1	0-SETUP
Call present (1)	0	1/0	0/1	1
User answer (2)	1	1/0-CONNECT	0/1	1
Network disconnect (3)	1	1/0	1/0	1-DISC
User hangup (3)	0	1/0-DISC	1/0	1

Note: (1) Alternates between user alerted and call presentation.
Note: (2) Start billing.
Note: (3) User hangup may come before or after network disconnect.

The System 930 Help Menu

Robbed Bit Signaling (continued)

Loop Start Signaling

Loop start was developed for telephone handset signaling. Network and user side are not symmetrical. On-hook/off-hook information is carried in one direction, and ring request information is carried in the other. User A bit represents loop current (LC) used to detect off-hook. User B bit is 1 for FXS/FXO, 0 for SAS/SAO. Network A bit represents loop current (LC) used for loop disconnect. Network B bit represents alerting current (AC) used to ring a bell.

There are four types of loop start signaling:
FXS: Foreign Exchange Subscriber. What the user side sends to FXO.
FXO: Foreign Exchange Office. What the network side sends to FXS.
SAS: Special Access Subscriber. What the user side sends to SAO.
SAO: Special Access Office. What the network side sends to SAS.

A foreign exchange is a remotely located exchange that the subscriber is connected to. This normally will be a central office. Special access is a dedicated line through the central office connected directly to a long distance carrier. Note that FXO/SAO and FXS/SAS are similar except that some bits are inverted.

Loop Start Signaling: User Initiated Call

	User to Network (User Output) FXS/SAS		Network to User (Network Output) FXO/SAO	
	A (LC)	B	A (LC)	B (AC)
Idle	0	1/0	0/1	1
User closes loop	1	1/0-CONNECT	0/1	1
Dialing follows				
Await answer	1	1/0	0/1	1
Network disconnect (1)	1	1/0	1/0	1-DISC
User hangup (1)	0	1/0-DISC	0/1	1

Note: (1) User hangup may come before or after network disconnect.

Loop Start Signaling: Network Initiated Call

	User to Network (User Output) FXS/SAS		Network to User (Network Output) FXO/SAO	
	A (LC)	B	A (LC)	B (AC)
Idle	0	1/0	0/1	1
User alerted (1)	0	1/0	0/1	0-SETUP
Await answer (1)	0	1/0	0/1	1
User answer (2)	1	1/0-CONNECT	0/1	1
Network disconnect (3)	1	1/0	1/0	1-DISC
User hangup (3)	0	1/0-DISC	0/1	1

Note: (1) Alternates between user alerted and await answer. If too much time passes between user alerted states, revert back to idle state.
Note: (2) Start billing.
Note: (3) User hangup may come before or after network disconnect.

The System 930 Help Menu

GR-303

GR-303 (formerly TR-303) is a Telcordia (formerly Bellcore) specification that supports remote deployment of the resources provided by a central office or other digital telephone system. This allows the physical interface point to be closer to customer locations. In GR-303, the central office is referred to as the IDT (integrated digital terminal). The remote location is called the RDT (remote digital terminal). GR-303 is a contemporary implementation of loop carrier systems, such as the venerable SLC-96 (TR-08). GR-303 uses standard DS1 circuits to connect an IDT (central office) with a RDT (remote location). DS1 circuits associated with GR-303 use ESF framing and B8ZS line coding.

GR-303 seemingly incorporates every telephony signaling protocol, including loop start, ground start, DID, coin phone, ISDN-BRI, and non-switched services. In addition to supporting standard metallic services, GR-303 also provides multi-rate (128kb/s and higher) data channels. Provision has also been provided for remote monitoring of the RDT. The GR-303 standard calls for the IDT to support a minimum number of services, while the RDT needs only to support the services appropriate to its function.

GR-303 supports two methods for call processing: hybrid signaling and CSC (common signaling channel) signaling. Hybrid signaling uses ABCD codes for on/off hook status and pulse dialing information. When sent over metallic DS1 circuits, ABCD codes use “robbed bits” from the 64kb/s bearer channels. Hybrid signaling uses ISDN-style messages for time-slot assignment and call processing. ISDN time-slot messages are sent on the TMC (time-slot management channel), which is equivalent to an ISDN D-channel. Time-slots are assigned on a call by call basis. Glare is eliminated by assigning time-slots from the IDT only. Calls are identified by their terminal number using the ISDN call reference value. Terminal numbers may range from 1 through 2048. This differs from normal ISDN usage in which the call reference value increments for each new call. Another difference is that within the call reference the GR-303 call reference value is offset by 3 bits.

GR-303 CSC (common signaling channel) signaling eliminates ABCD coding by combining their functionality into ISDN-style messages. Messages are sent over the CSC (common signaling channel) which is the same as the TMC (time-slot management channel).

Both Hybrid and CSC signaling send TMC/CSC information on time-slot 24. Dialing address digits are sent from the RDT to the IDT using DTMF tones on bearer channels. This method saves the expense of installing DTMF decoder resources in the RDT. Note that in the System 930, the IDT resources include a limitation of eight DTMF receivers. These receivers are shared on a first available basis. GR-303 provides handshaking to indicate when a DTMF receiver is available. Note that CSC signaling handles pulse dialing in a unique manner. The RDT receives pulse dialing digits, decodes them, then sends the information to the IDT using ISDN information messages.

GR-303 CSC supports calling number by sending a notify message to the IDT during the silent interval after the first long alert. This indicates that it is time for the IDT to send the calling number using an FSK modem. While the System 930 sends the notify message, the timing is guaranteed only when using the 936 analog card. Note that at present the System 930 does not support enhanced 911 features. If call transfer is set to outside on the Analog Port Configuration screen and GR-303 CSC signaling is selected, during a connected call the RDT sends switchhook flash using an information message containing hexadecimal zero. Hexadecimal zero appears as an @ on the data capture screen. The System 930 IDT uses switchhook flash to perform two way call transfer.

For network management GR-303 monitors the RDT using an EOC (embedded operations channel). The EOC uses time slot 12 of a DS1 to carry CMIP (common management information protocol) messages.

A GR-303 implementation utilizes from 1 to 28 DS1 circuits per GR-303 group. The standard calls for NFAS (non-facility associated signaling) to control all bearer channels within a group. That is, all bearer channels are controlled by one TMC/CSC signaling channel. This is more efficient than using a separate TMC/CSC signaling channel for each DS1 circuit. The standard also calls for the first DS1 circuit to reserve time-slots 12 for EOC monitoring and time-slot 24 for TMC/CSC signaling. The remaining 22 channels are available for bearer traffic. The 24 channels of the other DS1 circuits are available for bearer traffic. The standard allows a second DS1 circuit to reserve time-slots 12 for backup EOC monitoring and channel 24

GR-303 (continued)

for backup TMC/CSC signaling should the first DS1 fail. Facility protection switching enables traffic on a failed DS1 circuit to be moved to other DS1 circuits. Note that the System 930 does not currently support the backup option nor facility protection switching. When the System 930 is licensed and configured for NFAS, channel 24 of the first DS1 controls all bearer channels of both DS1 interfaces. These form a group of 46 bearer channels, 22 from interface one and 24 from interface two.

The System 930 supports hybrid signaling and CSC (common signaling channel) signaling. Future versions are expected to support the EOC. Contact the factory for details. The System 930 does not support pulse dialing. The System 930 does not support hybrid signaling with non-switched (nailed up) circuits. As of late 1999, most central offices support only GR-303 hybrid signaling.

The System 930 provides two DS1 interfaces. When not configured for NFAS, each interface can be independently configured for IDT or RDT support. By setting one DS1 interface as IDT and the other as RDT, as well as connecting the DS1 interfaces together using a reversing data cable, GR-303 calls can be sent from one interface to the other.

Some of the selections on the System 930's digital interface test function screen have no effect on GR-303. This is because GR-303 always sends setup bearer capability information as unrestricted digital information, and channel ID with interface ID present, exclusive channel ID, and numbered channel ID format. Progress messages are not available and progress tones are always sent in-band.

Some of the selections on the System 930's Channel Configuration screen have no effect on GR-303. GR-303 does not send service messages to enable or disable channels across the network.

The following is a list of events that take place during GR-303 Hybrid call setup and teardown using loop start circuit emulation. IDT Initiated Hybrid Signaling Call Setup: IDT sends loop current feed (0101) using ABCD bits. IDT sends setup message containing channel ID. RDT sends loop open (0101) using ABCD bits. RDT sends connect message. IDT sends ringing (0000) and loop current feed (0101) using ABCD bits, alternating between the two. When the called party answers, the RDT sends loop closed (1111) using ABCD bits. Note that the System 930 call counters and status display count the connect message as a setup ack.

RDT Initiated Hybrid Signaling Call Setup: RDT sends setup message. IDT sends loop current feed (0101) using ABCD bits. IDT sends connect message containing channel ID. RDT sends loop open (0101) using ABCD bits. RDT sends connect ack message. RDT sends loop closed (1111) using ABCD bits. IDT sends in-band dial tone. RDT sends in-band DTMF digits. IDT sends in-band audible ring. Note that the System 930 call counters and status display count the connect message as a setup ack.

IDT Initiated Hybrid Signaling Call Teardown: IDT waits for RDT initiated call teardown, per Section 5.4 of GR-505-CORE. IDT sends loop current feed open (1111) using ABCD bits for 800 ms, per Section 5.5.1 of GR-505-CORE. IDT sends in-band ROH (receiver off-hook) tone. IDT sends disconnect message with cause code 27 (destination out of service). RDT sends release message. IDT sends release complete message. The RDT is now in the permanent signal state. When the RDT goes on-hook, RDT sends an information message with switchhook on-hook, per Section 12.5.5.14 of GR-303-CORE. Special thanks to Rob Bond at Telcordia for clarifying this.

RDT Initiated Hybrid Signaling Call Teardown: RDT sends loop open (0101) using ABCD bits. IDT sends disconnect message. RDT sends release message. IDT sends release complete message.

The following is a list of events that take place during GR-303 CSC call setup and teardown using loop start or ground start circuit emulation.

IDT Initiated CSC Signaling Call Setup: IDT sends setup message containing channel ID. RDT sends alerting message. RDT sends notify message with silent interval. IDT sends in-band analog calling number (if applicable). IDT sends connect message.

GR-303 (continued)

RDT Initiated CSC Signaling Call Setup: RDT sends setup message. IDT sends setup ack message containing channel ID. IDT sends in-band dial tone. RDT sends in-band DTMF or information message containing pulse digits. IDT sends in-band audible ring. IDT sends connect message. RDT sends connect ack message.

IDT Initiated CSC Signaling Call Teardown: IDT sends disconnect message. RDT sends release message. IDT sends release complete message.

RDT Initiated CSC Signaling Call Teardown: RDT sends disconnect message. IDT sends notify message with flash disable. IDT sends release message. RDT sends release complete message.

Coding differences between GR-303 and Q.931:
Protocol Discriminator is 0x4F for GR-303, 0x08 for Q.931.
Channel ID octet 3 is 0x69 for GR-303, 0xE9 for Q.931.
The first interface ID is 1 for GR-303, 0 for Q.931.

GR-303 Nomenclature:
CSC: Common Signaling Channel.
IDT: Integerated Digital Terminal (network side).
LDS: Local Digital Switch (network side).
RDT: Remote Digital Terminal (user side).
TMC: Time Management Channel. Also known as hybrid signaling.

Calling Name

There are two methods for sending calling name: Display IE (information element) and Facility IE.

In North America, Display IE is sent in codeset 6 (network specific) because it is a proprietary method. This works for names originating from the network or user side. In Europe, ETSI recommends sending Display IE in codeset 0 (normal). The Display IE is sent from the network to user side only. In either case, calling name is sent as part of setup messages. Coding of the display IE is very simple. Here is an example:

Coding of Display IE for Calling Name “ABC”:
0x28 Display IE
0x03 IE Length
0x41 Display Information: A
0x42 Display Information: B
0x43 Display Information: C

Use of the Facility IE follows the Telcordia GR-1367-CORE recommendation. This service is part of the 1996 NI-3 feature offering and is available on National and 5ESS Custom PRIs. Calling Name consists of three separate features: Calling Name Delivery, Privacy of Calling Name, and Electronic Directory Service. Calling Name Delivery provides name information to a subscribed terminating CPE. Name characters are obtained from a name database corresponding to the calling party number. Privacy of Calling Name allows an originating CPE to signal on a per-call basis whether the calling name should be displayed. Electronic Directory Service allows an originating CPE to send original calling name to intrabusiness group calls.

The System 930 supports the first of these features: Calling Name Delivery. This feature is implemented using the ROSE (Remote Operation Service Element) invoke service component of the facility IE.

Calling Name Delivery is a two-step process. The first step depends on whether the calling number is available and whether the privacy indicator is turned on or off. If the calling number is missing or the privacy indicator is turned on, a facility IE is sent in the setup message indicating that the name is not available or is private. If the calling number is present and the privacy indicator is turned off, a facility IE is sent in the setup message indicating information following. This gives the central office time to look up the calling name. The central office then sends a facility message containing the facility IE with the calling name.

Calling Name (continued)

Coding of the facility IE is very complex. It uses ASN.1 (Abstract Syntax Notation One), which is a database language defined in ITU-T X.209. Here are four examples:

Coding of Facility IE for Name Not Available:
0x1c Facility IE
0x0e IE Length
0x9f Service Discriminator: Network Extension
0x8b Component Type: Interpretation
0x01 Component Length
0x00 Discard Unrecognized Invoke Components
0xa1 Component Type: Invoke
0x08 Component Length
0x02 Invoke ID Tag: Universal Integer
0x01 Invoke ID Length
0x00 Invoke ID Value
0x02 Operation Tag: Universal Integer
0x01 Operation Length
0x00 Operation Value: Calling Name
0x84 Argument Tag: Context-Specific NameNotAvailable
0x00 Argument Length

Coding of Facility IE for Name Presentation Restricted:
0x1c Facility IE
0x0e IE Length
0x9f Service Discriminator: Network Extension
0x8b Component Type: Interpretation
0x01 Component Length
0x00 Discard Unrecognized Invoke Components
0xa1 Component Type: Invoke
0x08 Component Length
0x02 Invoke ID Tag: Universal Integer
0x01 Invoke ID Length
0x00 Invoke ID Value
0x02 Operation Tag: Universal Integer
0x01 Operation Length
0x00 Operation Value: Calling Name
0x87 Argument Tag: Context-Specific NamePresentationRestrictedNull
0x00 Argument Length

Coding of Facility IE for Information Following:
0x1c Facility IE
0x15 IE Length
0x9f Service Discriminator: Network Extension
0x8b Component Type: Interpretation
0x01 Component Length
0x00 Discard Unrecognized Invoke Components
0xa1 Component Type: Invoke
0x0f Component Length
0x02 Invoke ID Tag: Universal Integer
0x01 Invoke ID Length
0x00 Invoke ID Value
0x06 Operation Tag: Universal Object ID
0x07 Operation Length

Calling Name (continued)

0x2a Operation Value: informationFollowing 42 (Note 1)
0x86 informationFollowing 840
0x48
0xce informationFollowing 10005
0x15
0x00 informationFollowing 0
0x04 informationFollowing 4
0x0a Argument Tag: Universal Enumerated
0x01 Argument Length
0x00 Argument Value: includesNameInformation

Note 1: The object ID for informationFollowing is 1.2.840.10005.0.4. The first two digits are encoded as (40 * 1) + 2 = 42. This represents: ISO (1), member-body (2), USA country code (840), ANSI T1 (10005) operations (0), informationFollowing (4).

Coding of Facility IE for Calling Name “ABC”:
0x1c Facility IE
0x11 IE Length
0x9f Service Discriminator: Network Extension
0x8b Component Type: Interpretation
0x01 Component Length
0x00 Discard Unrecognized Invoke Components
0xa1 Component Type: Invoke
0x0b Component Length
0x02 Invoke ID Tag: Universal Integer
0x01 Invoke ID Length
0x01 Invoke ID Value (must be different from other Invoke IDs)
0x02 Operation Tag: Universal Integer
0x01 Operation Length
0x00 Operation Value: Calling Name
0x80 Argument Tag: Context-Specific NamePresentationAllowedSimple
0x03 Argument Length (1 to 50 characters)
0x41 Argument: A
0x42 Argument: B
0x43 Argument: C

Enhanced Explicit Call Transfer

EECT (Enhanced Explicit Call Transfer) allows two outside calls to be connected by the central office instead of the CPE. This frees up B-channels. Billing to the transferring party continues for as long as the outside call remains connected.

ISDN implements EECT using the facility IE (information element) as recommended by Telcordia GR-2865-CORE and ANSI T1S1.1/96-346. On 5ESS PRIs it is know as TBCT (Two B-Channel Transfer) and was first available with software release 5E13. Initially two calls are setup and placed in the connect state. The transfer is initiated by sending a facility message with a facility IE containing an invoke services component.

Coding of Facility IE to invoke TBCT:
0x1c Facility IE
0x15 IE Length
0x91 Service Discriminator: Supplementary Services
0x8b Component Type: Interpretation
0x01 Component Length

Enhanced Explicit Call Transfer (continued)

0x00 Discard Unrecognized Invoke Components
0xa1 Component Type: Invoke
0x0f Component Length
0x02 Invoke ID Tag: Universal Integer
0x01 Invoke ID Length
0x00 Invoke ID Value
0x06 Operation Tag: Universal Object ID
0x07 Operation Length
0x2a Operation Value: EnhancedExplicitECTExecute 42 (Note 1)
0x86 EnhancedExplicitECTExecute 840
0x48
0xce EnhancedExplicitECTExecute 10005
0x15
0x00 EnhancedExplicitECTExecute 0
0x08 EnhancedExplicitECTExecute 8
0x30 Argument Tag: Universal Sequence
0x01 Argument Length
0x00 Argument Value: Call Reference with flag of call to be transferred.

Note 1: The object ID for EnhancedExplicitECTExecute is 1.2.840.10005.0.8. The first two digits are encoded as (40 * 1) + 2 = 42. This represents: ISO (1), member-body (2), USA country code (840), ANSI T1 (10005) operations (0), EnhancedExplicitECTExecute (8). If the transfer can be carried out, the 5ESS switch sends the controller a disconnect message for each of the two B-channels. The disconnect message for the B-channel associated with the TBCT request contains a facility IE with a return result component. The disconnect message for the other B-channel does not contain a facility IE.

Coding of Facility IE with return result component:
0x1c Facility IE
0x14 IE Length
0x91 Service Discriminator: Supplementary Services
0x8b Component Type: Interpretation
0x01 Component Length
0x00 Discard Unrecognized Invoke Components
0xa2 Component Type: Return Result
0x0f Component Length
0x02 Return ID Tag: Universal Integer
0x01 Return ID Length
0x02 Return ID Value (must match invoke ID value)
0x06 Operation Tag: Universal Object ID
0x05 Operation Length
0x2b Operation Value: transfers 43 (Note 2)
0x11 transfers 17
0x66 transfers 102
0x03 transfers 3
0x02 transfers 2
0x30 Argument Tag: Universal Sequence
0x02 Argument Length
0x00 Argument Value: Active Transfers
0x00 Argument Value: Available Transfers

Note 2: The object ID for transfers is 1.3.17.102.3.2. The first two digits are encoded as (40 * 1) + 3 = 43. This represents: ISO (1), member-body (3), Bellcore (17), ISDN-supplementary-services (102) tbct (3), transfers (2).

Enhanced Explicit Call Transfer (continued)

RBS implements EECT using switch hook flash as recommended by Telcordia TR-TSY-000506.

On 5ESS RBS it is known as 2-Way DID with Call Transfer Disconnect. E&M signaling is used. Initially one call is setup and placed in the connect state. A switch hook flash (wink) of between 300 to 1100 ms causes the central office to place the call on hold and return a second dial tone. The user DTMF dials a third party and waits for the third party to connect. After a brief conversation the user goes on-hook and is disconnected, freeing the B-channel. The two remaining parties are connected together by the central office.

The System 930 supports EECT signaling for RBS and GR-303 but not ISDN. To keep things simple, the calls are disconnected across the network and are connected internally on the network side. EECT is enabled by flashhook on the Analog Port Configuration screen.

The System 930 Main Menu

Configuration and status screens are selected from the main menu. Online help is available from all screens by pressing <F1>. Screens are quickly configured in real time and shown in real time. Once a screen has been selected, pressing <ESC> <F4> or <ESC> <F3> skips to the next or previous screen without going to the main menu.

```
Gordon Kapes, Inc.    System 930 Telephony Simulator              Main 1

                                MAIN MENU

                 1. DIGITAL INTERFACE CONFIGURATION
                 2. CHANNEL CONFIGURATION
                 3.  CALLING NUMBER CONFIGURATION
                 4.  ANALOG PORT CONFIGURATION
                 5.  RECORDER/ANNOUNCER
                 6.  INBOUND MATCH CONFIGURATION
                 7.  ANALOG PORT DIALING CONFIGURATION
                 8.  ANALOG PORT OUTBOUND CALL CONFIGURATION
                 9.  ACD CONFIGURATION
                10.  REDIRECT CONFIGURATION
                11.  AUDIO MONITOR CONFIGURATION
                12.  RESERVED FOR FUTURE USE
                13.  DIGITAL INTERFACE CALL STATUS
                14.  TRANSMISSION STATUS


                 Enter Selection:

                 Enter number or press Up/Down Arrow then <ENTER>
                       Press "x" exit, <F1> help, <F4> next
```

```
Gordon Kapes, Inc.    System 930 Telephony Simulator              Main 2

                                MAIN MENU

                  15.  DATA MONITOR
                  16.  TONE CONNECT TEST FUNCTIONS
                  17.  DIGITAL INTERFACE TEST FUNCTIONS
                  18.  DATA CAPTURE
                  19.  QUICK SYSTEM STATUS
                  20.  SAVE/RESTORE SYSTEM CONFIGURATION
                  21.  CALL GENERATOR CONFIGURATION
                  22.  CALL COUNTERS
                  23.  CONNECT ACTION CONFIGURATION
                  24.  TIME & DATE CONFIGURATION
                  25.  939 ANALOG CARD CONFIGURATION
                  26.  SECURITY CONFIGURATION



                 Enter Selection:

                 Enter number or press Up/Down Arrow then <ENTER>
                    Press "x" exit, <F1> help, <F3> previous
```

Digital Interface Configuration—Master

This screen configures the methods used for sending information across both digital interfaces. Press <ENTER> then Y to update both interfaces. Use detailed screens to configure individual interfaces.

```
Gordon Kapes, Inc.     System 930 Telephony Simulator       Screen 1-1

               DIGITAL INTERFACE CONFIGURATION - MASTER

Interface 1 Status:  SYNCHRONIZED
Interface 2 Status:  SYNCHRONIZED

Framing:            ESF (EXTENDED SUPERFRAME)* ALL DETAILED SCREENS MATCH
Line Coding:        B8ZS (ZERO SUPPRESSION)*   ALL DETAILED SCREENS MATCH
Line Build Out:     OdB*                       ALL DETAILED SCREENS MATCH
Sync Source:        INTERNAL*
Switch Emulation:   NATIONAL ISDN-2*           ALL DETAILED SCREENS MATCH
Location:           MIXED*                     ALL DETAILED SCREENS MATCH
Channel Search:     MIXED*                     ALL DETAILED SCREENS MATCH
Interface Enabled: YES*                        ALL DETAILED SCREENS MATCH
Interface Wiring:   NETWORK (RJ48)*            ALL DETAILED SCREENS MATCH




* Indicates factory default.

               Press Space Bar or Backspace then <ENTER>
           Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

Help for Digital Interface Configuration—Master

Interface Status: Shows CARD NOT PRESENT, DISABLED, RECEIVING BLUE ALARM, RECEIVING YELLOW ALARM, SYNCHRONIZED, NOT SYNCHRONIZED, or LAYER 2 DOWN. Indicates framing status of digital interface. Layer 2 down is applicable to ISDN or GR-303 digital interfaces only.

Framing: Select ESF (EXTENDED SUPERFRAME)* (default) or D4 (SUPERFRAME). ESF is required for ISDN.

Line Coding: Select B8ZS* (zero suppression) (default), AMI (no zero suppression), or ZCS (jammed bit). B8ZS stands for bipolar 8 zero suppression. B8ZS is the preferred method as it transparently maintains the one's density necessary for accurate clock recovery. AMI stands for alternate mark inversion with no zero suppression performed. ZCS stands for zero code suppression with destructive jammed bit insertion performed. Line Build Out: Select OdB* (default), –7.5dB, –15dB, or –22dB. Indicates strength of transmit signal. OdB provides the strongest signal and –22dB provides the weakest.

Sync Source: Select INTERNAL* (default), DIGITAL INTERFACE 1, or DIGITAL INTERFACE 2. Indicates whether system timing synchronizes to the internal timing source or to the incoming frame. Sync source is common to both digital interfaces. Affects the outgoing frame only. The incoming frame is always self-synchronizing. This feature is not selectable from detailed screens.

Switch Emulation: Select NATIONAL ISDN-2* (default), NATIONAL ISDN-2 WITH NFAS, 4ESS CUSTOM, 4ESS CUSTOM WITH NFAS, 5ESS CUSTOM, 5ESS CUSTOM WITH NFAS, DMS100 CUSTOM, DMS100 CUSTOM WITH NFAS, GR-303 HYBRID INDEPENDENT DS1, GR-303 HYBRID DUAL DS1, GR-303 CSC INDEPENDENT DS1, GR-303 CSC DUAL DS1 or T1 ROBBED BIT SIGNALING. As of late 1997, all public switches support NATIONAL ISDN-2. Use National ISDN-2 for GTD5 and DMS250 switch emulation. 5ESS, DMS100, GR-303, and GTD switches. 4ESS and DMS250 switches are tandem (interoffice) switches.

Interface Configuration—Master

Help for Digital Interface Configuration—Master (continued)

NFAS stands for non-facility associated signaling and indicates that signaling is done on the D-channel of interface 1. Unless NFAS or dual DS1 is used, assume that each interface carries its own signaling. Changing this selection to one of the GR-303 types automatically changes the outbound access mode on the Analog Port Dialing Configuration screen to immediate access. Some selections may not be available depending on how the system is licensed. <ENTER> must be pressed to change this feature. Changing this field causes all calls in progress to disconnect, the system to reboot, and the screen to be cleared.

Location: Select MIXED* (default), NETWORK SIDE, PRIVATE NETWORK SIDE (ISDN only), or USER SIDE. Mixed indicates interface 1 emulates a central office and interface 2 emulates CPE. Network side indicates both interfaces emulate a central office. Private network side indicates both interfaces emulate the network side of a private system. User side indicates both interfaces emulate CPE. ISDN sends this information in the progress indicator information element and cause information elements of ISDN messages. Changing this field causes all calls in progress to be disconnected. When changed to user side, outbound channel search is set to descending sequential and synchronization source is set to digital interface 1. When changed to network side, outbound channel search is set to ascending sequential and synchronization source is set to internal. Some selections may not be available depending on how the system is licensed.

Channel Search: Select MIXED* (default), ASCENDING SEQUENTIAL, DESCENDING SEQUENTIAL, CLOCKWISE CIRCULAR, or COUNTER CLOCKWISE CIRCULAR. Used for finding the next outbound idle bearer channel. Mixed sets digital interface 1 to ascending sequential and digital interface 2 to descending sequential. Ascending sequential starts searching from channel one. Descending sequential starts searching from the highest channel. Clockwise circular starts one channel higher than the previously utilized channel, wrapping around from the highest channel to channel one. Counter clockwise circular starts one channel lower than the previously utilized channel, wrapping around from channel one to the highest channel. To reduce glare, ascending sequential is appropriate for the network side, while descending sequential is appropriate for the user side.

Interface Enabled: Select YES* (default) or NO. No shuts off the transmitter. If yes is selected and the received signal is not synchronized, the associated front panel lamp blinks. If yes is selected and the received signal is synchronized, the lamp lights steadily. If no is selected, the lamp is extinguished.

Interface Wiring: Select FOLLOWS LOCATION* (default), NETWORK (RJ48), or USER (CI). Follows location indicates the connection pin-out follows the configured location. Network (RJ48) indicates the connection is wired for the network interface as described in specification RJ48. User (CI) indicates the connection is wired for the customer interface. The above applies to System 930A hardware. System 930 hardware is preset to network (RJ48) and cannot be changed.

After most selections the screen displays ALL DETAILED SCREENS MATCH, SOME DETAILED SCREENS MATCH, or NO DETAILED SCREENS MATCH. This indicates whether the selection matches the same field on the detailed screens.

Digital Interface Configuration—Detailed

This screen configures the methods used for sending information across the digital interface. There are two configuration screens, one for each digital interface. Use the master screen to simultaneously configure both interfaces.

```
Gordon Kapes, Inc.     System 930 Telephony Simulator       Screen 1-2

             DIGITAL INTERFACE 1 CONFIGURATION - DETAILED

Interface Status:   SYNCHRONIZED

Framing:            ESF (EXTENDED SUPERFRAME)*
Line Coding:        B8ZS (ZERO SUPPRESSION)*
Line Build Out:     0dB*
Switch Emulation:   NATIONAL ISDN-2*
Location:           NETWORK SIDE*
Channel Search:     ASCENDING SEQUENTIAL
Interface Enabled:  YES*
Interface Wiring:   NETWORK (RJ48)*







* Indicates factory default.

                Press Space Bar or Backspace to select
     Press Arrow Keys, <F1> help, <F2> exit, <F3> previous, <F4> next
```

Help for Digital Interface Configuration—Detailed

Interface Status: Shows CARD NOT PRESENT, DISABLED, RECEIVING BLUE ALARM, RECEIVING YELLOW ALARM, SYNCHRONIZED, NOT SYNCHRONIZED, or LAYER 2 DOWN. Indicates framing status of digital interface. Layer 2 down is applicable to ISDN or GR-303 digital interfaces only.

Framing: Select ESF (EXTENDED SUPERFRAME)* (default) or D4 (SUPERFRAME). ESF is required for ISDN.

Line Coding: Select B8ZS* (zero suppression) (default), AMI (no zero suppression), or ZCS (jammed bit). B8ZS stands for bipolar 8 zero suppression. B8ZS is the preferred method as it transparently maintains the one's density necessary for accurate clock recovery. AMI stands for alternate mark inversion with no zero suppression performed. ZCS stands for zero code suppression with destructive jammed bit insertion performed.

Line Build Out: Select 0dB* (default), –7.5dB, –15dB, or –22dB. Indicates strength of transmit signal. 0dB provides the strongest signal and –22dB provides the weakest.

Switch Emulation: Select NATIONAL ISDN-2* (default), 4ESS CUSTOM, 5ESS CUSTOM, DMS100 CUSTOM, GR-303 HYBRID, GR-303 CSC, NFAS (USES INTERFACE 1), or T1 ROBBED BIT SIGNALING. As of late 1997, all public switches support NATIONAL ISDN-2. Use National ISDN-2 for GTD5 and DMS250 switch emulation. 5ESS, DMS100, GR-303, and GTD5 switches are end-node (central office) switches. 4ESS and DMS250 switches are tandem (interoffice) switches. NFAS (USES INTERFACE 1) is available only on interface 2. NFAS stands for non-facility associated signaling and indicates that signaling is done on the D-channel of interface 1. Unless NFAS is used, assume that each interface carries its own signaling. Some selections may not be available depending on how the system is licensed. <ENTER> must be pressed to change this feature. Changing this field causes all calls in progress to disconnect, the system to reboot, and the screen to be cleared.

Digital Interface Configuration—Detailed

Help for Digital Interface Configuration—Detailed (continued)

Location: Select NETWORK SIDE* (default), PRIVATE NETWORK SIDE (ISDN only), or USER SIDE. Indicates whether the interface emulates a central office (network side), a private network (private network side), or CPE (user side). <ENTER> must be pressed to change this feature. Changing this field causes all calls in progress to be disconnected. ISDN sends this information in the progress indicator information element and cause information elements of ISDN messages. Some selections may not be available depending on how the system is licensed. NFAS signaling on digital interface 2 does not used this field.

Channel Search: Select ASCENDING SEQUENTIAL, DESCENDING SEQUENTIAL, CLOCKWISE CIRCULAR, or COUNTER CLOCKWISE CIRCULAR. Used for finding the next outbound idle bearer channel. Ascending sequential starts searching from channel one. Descending sequential starts searching from the highest channel. Clockwise circular starts one channel higher than the previously utilized channel, wrapping around from the highest channel to channel one. Counter clockwise circular starts one channel lower than the previously utilized channel, wrapping around from channel one to the highest channel. To reduce glare, ascending sequential is appropriate for the network side, while descending sequential is appropriate for the user side. Factory default varies, depending on how location is set. GR-303 shows NOT DETERMINED BY RDT when location is RDT.

Interface Enabled: Select YES* (default) or NO. No shuts off the transmitter. If yes is selected and the received signal is not synchronized, the associated front panel lamp blinks. If yes is selected and the received signal is synchronized, the lamp lights steadily. If no is selected, the lamp is extinguished.

Interface Wiring: Select FOLLOWS LOCATION* (default), NETWORK (RJ48), or USER (CI). Follows location indicates the connection pin-out follows the configured location. Network (RJ48) indicates the connection is wired for the network interface as described in specification RJ48. User (CI) indicates the connection is wired for the customer interface. The above applies to System 930A hardware. System 930 hardware is preset to network (RJ48) and cannot be changed.

Channel Configuration—Master

This screen allows all channels on both digital interfaces to be blocked from processing calls. Press <ENTER> then Y to update all channels. Use detailed screens to configure individual channels.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 2-1

                    CHANNEL CONFIGURATION - MASTER

Interface 1 Status:  SYNCHRONIZED
Interface 2 Status:  SYNCHRONIZED

Operation:         ENABLED*                    ALL DETAILED SCREENS MATCH
Service Update:    NONE*                       ALL DETAILED SCREENS MATCH
D-Channel Status:  DO NOT SEND*










* Indicates factory default.

           Press Space Bar or Backspace to select then <ENTER>
          Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

ISDN Channel Configuration–Master Screen

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 2-1

                    CHANNEL CONFIGURATION - MASTER

Interface 1 Status:  SYNCHRONIZED
Interface 2 Status:  SYNCHRONIZED

Operation:         ENABLED*                    ALL DETAILED SCREENS MATCH
Direction:         TWO WAY*                    ALL DETAILED SCREENS MATCH
Signaling:         E&M WINK*                   ALL DETAILED SCREENS MATCH
Inbound Routing:   INBOUND MATCH*              ALL DETAILED SCREENS MATCH









* Indicates factory default.

          Press Space Bar or Backspace to select then <ENTER>
           Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

RBS Channel Configuration–Master Screen

Channel Configuration—Master

Help for Channel Configuration—Master

Interface Status: Shows CARD NOT PRESENT, DISABLED, RECEIVING BLUE ALARM, RECEIVING YELLOW ALARM, SYNCHRONIZED, NOT SYNCHRONIZED, or LAYER 2 DOWN. Indicates framing status of digital interface.

Operation: Select ENABLED* (default), DISABLED, MAINTENANCE, or EXCLUSIVE. Enabled makes all digital interface channels available for inbound or outbound calls. Disabled and maintenance prevent the digital interface channel from processing inbound or outbound call. Exclusive overrides the channel search method and chooses the outgoing channel based on the calling resource. Calls can originate from analog ports, explicit channel dialing (see Analog Port Dialing Configuration), redirected channel of the opposite interface, or the call generator. Use exclusive to emulate outgoing channel bank operation. Exclusive enables channels to receive calls. Operation implementation depends on whether switch emulation is ISDN, RBS, or GR-303.

ISDN implements operation by sending an ISDN service message containing change status and channel ID information elements. The ISDN service message is not sent immediately, but rather uses the selected update method configured on detailed screens. Each ISDN service message must be acknowledged by the near end or it will be sent again. If an ISDN channel is blocked from receiving inbound calls, the system reassigns the call to another channel.

RBS uses operation to put channels in disabled and maintenance mode by transmitting the busy state, keeping the receiving channel in the transition state when the digital interface goes from not synchronized to synchronized. Exclusive mode is implemented by using the appropriate channel.

GR-303 uses operation to locally put channels in service, out of service, or in maintenance. Maintenance mode behaves the same as in service. Exclusive mode behaves the same as enabled. GR-303 does not support service messages to remotely enable or disable channels.

Service Update: Select NONE* (default), NETWORK SIDE LAYER 2 STARTUP, USER SIDE LAYER 2 STARTUP, or UPON LAYER 2 STARTUP. This selection determines when ISDN service messages containing the B-channel status are sent across the network. Each interface sends a total of 23 service messages, one for each bearer channel. A service message is also sent independent of this selection whenever the status of an individual channel is changed on the channel configuration detailed screen. None indicates that service messages are not sent across the network. Network side layer 2 startup indicates that service messages are sent upon layer 2 startup provided this interface is configured for network side. User side layer 2 startup indicates that service messages are sent upon layer 2 startup provided this interface is configured for user side. Upon layer 2 startup indicates that service messages are sent upon layer 2 startup. B-channel status may be sent by 4ESS custom, 5ESS custom, and DMS-100 custom. B-channel status is not sent by NI-2 although we have seen Nortel central offices running NI-2 that send B-channel status. Not applicable to GR-303 which never sends B-channel status. Shown only when at least one interface is ISDN.

D-Channel Status: Select DO NOT SEND* (default), or NETWORK SIDE LAYER 2 STARTUP. This selection determines whether or not the system sends ISDN service messages containing the D-channel status. If sent, D-channel status is sent according to the selected update method and is sent prior to B-channel service messages. Do not send indicates that the system does not send service messages containing the D-channel status. D-channel service messages are not sent on a network containing one D-channel per DS1, also known as facility associated signaling (FAS). Network side layer 2 startup indicates that the network side sends service messages containing the D-channel status. D-channel service messages are sent on a network containing one D-channel per multiple DS1s, also known as non facility associated signaling (NFAS). Use this selection to emulate the primary PRI of a network using NFAS without D-channel backup. Note that this system does not currently support D-channel backup. This feature is not selectable from detailed screens. Not applicable to GR-303 which never sends D-channel status. Shown only when at least one interface is ISDN.

Channel Configuration—Master

Help for Channel Configuration—Master (continued)

Direction: Select TWO WAY* (default), OUTBOUND, or INBOUND. Two way channels can originate or receive calls. Outbound channels can only originate calls. Inbound channels can only receive calls. Shown only when at least one interface is RBS.

Signaling: Select E&M WINK* (default), E&M IMMEDIATE, GROUND FXS/FXO, GROUND SAS/SAO, LOOP FXS/FXO, or LOOP SAS/SAO. Signaling definitions appear later in this text. Shown only when at least one interface is RBS.

Inbound Routing: Select INBOUND MATCH* (default), ACD 1 through ACD 4, CONSECUTIVE ACD, CONSECUTIVE PORTS, REDIRECT 1 through REDIRECT 4, MSG 1 PLAY TWICE, MSG 2 PLAY TWICE, CONNECT ACTION 1 through CONNECT ACTION 3, BUSY, or REORDER. Indicates routing of inbound calls. Inbound match routes an incoming call by matching the DNIS-DTMF digits with the match number in the Inbound Match Configuration screen. ACD 1 through ACD 4 route incoming calls to an analog port via an ACD group. The ACD group is configured using the ACD Configuration screen. Incoming DNIS-DTMF digits are ignored. There are more than four ACDs, but for ease of use only the first four are selectable from the master screen. Consecutive ACD means that channel 1 is routed to ACD 1, channel 2 is routed to ACD 2, etc. The ACD group is configured using the ACD Configuration screen. Incoming DNIS-DTMF digits are ignored. Consecutive ports means that channel 1 is routed to port 1, channel 2 is routed to port 2, etc. Incoming DNIS-DTMF digits are ignored. Use consecutive port to emulate inbound channel bank operation. Redirect 1 through redirect 4 route incoming calls to an outgoing digital interface channel via a redirect group. The redirect group is configured using the Redirect Configuration screen. Incoming DNIS-DTMF digits are ignored. Msg 1 play twice and msg 2 play twice route incoming calls to the selected voice message. The message is played twice, followed by a forced disconnect. Incoming DNIS-DTMF digits are ignored. Connect action routes the call to the connect action on the connect action screen. Incoming DNIS-DTMF digits are ignored. Busy routes an incoming call to the busy progress tone. Incoming DNIS-DTMF digits are ignored. Reorder routes an incoming call to the reorder progress tone. Incoming DNIS-DTMF digits are ignored. Inbound-routing shows hyphens (---) when direction is outbound. Shown only when at least one interface is RBS.

After each selection the screen displays ALL DETAILED SCREENS MATCH, SOME DETAILED SCREENS MATCH, or NO DETAILED SCREENS MATCH. This indicates whether the selection matches the same field on the detailed screens.

There are three types of RBS protocols: E&M, ground start, and loop start. E&M is the preferred method for PBXs. E&M signaling includes E&M WINK and E&M IMMEDIATE. With wink, the receiving side must send an acknowledgment (wink) before dialing can begin. With immediate, dialing can start immediately. Tie trunks use E&M signaling. History buffs note that E&M stands for ear and mouth.

Ground start signaling includes GROUND FXS, GROUND FXO, GROUND SAS, and GROUND SAO. FXS stands for foreign exchange subscriber, and is what the user side sends to FXO. FXO stands for foreign exchange office, and is what the network side sends to FXS. SAS stands for special access subscriber, and is what the user side sends to SAO. SAO stands for special access office, and is what the network side sends to SAS.

Loop start signaling includes LOOP FXS, LOOP FXO, LOOP SAS, and LOOP SAO. Loop start is what a plain old telephone (POTS) uses. The definitions are essentially the same as ground start. Loop start is the least preferred method for PBXs, because older loop start FXO and SAO facilities cannot signal the user to disconnect.

Channel Configuration—Detailed

This screen allows individual channels to be blocked from making or receiving calls. There are four detailed screens, two for each digital interface. The screens may differ, depending on whether the interface is configured for ISDN or RBS.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator         Screen 2-2

        CHANNEL CONFIGURATION - DETAILED - DIGITAL INTERFACE 1

Interface Status:  SYNCHRONIZED

DCN   Operation       Status
1/1   ENABLED*        IN SERVICE
1/2   ENABLED*        IN SERVICE
1/3   ENABLED*        IN SERVICE
1/4   ENABLED*        IN SERVICE
1/5   ENABLED*        IN SERVICE
1/6   ENABLED*        IN SERVICE
1/7   ENABLED*        IN SERVICE
1/8   ENABLED*        IN SERVICE
1/9   ENABLED*        IN SERVICE
1/10  ENABLED*        IN SERVICE
1/11  ENABLED*        IN SERVICE
1/12  ENABLED*        IN SERVICE


* Indicates factory default.
       Press Space Bar or Backspace to select. Press A for All.
         Press <F1> help, <F2> exit, <F3> previous, <F4> next
```

ISDN Channel Configuration–Detailed Screen

```
Gordon Kapes, Inc.   System 930 Telephony Simulator         Screen 2-2

         CHANNEL CONFIGURATION - DETAILED - DIGITAL INTERFACE 1

Interface Status:  SYNCHRONIZED

DCN   Operation     Direction    Signaling      Inbound Routing
1/1   ENABLED       TWO WAY      E&M WINK*      INBOUND MATCH*
1/2   ENABLED       TWO WAY      E&M WINK*      INBOUND MATCH*
1/3   ENABLED       TWO WAY      E&M WINK*      INBOUND MATCH*
1/4   ENABLED       TWO WAY      E&M WINK*      INBOUND MATCH*
1/5   ENABLED       TWO WAY      E&M WINK*      INBOUND MATCH*
1/6   ENABLED       TWO WAY      E&M WINK*      INBOUND MATCH*
1/7   ENABLED       TWO WAY      E&M WINK*      INBOUND MATCH*
1/8   ENABLED       TWO WAY      E&M WINK*      INBOUND MATCH*
1/9   ENABLED       TWO WAY      E&M WINK*      INBOUND MATCH*
1/10  ENABLED       TWO WAY      E&M WINK*      INBOUND MATCH*
1/11  ENABLED       TWO WAY      E&M WINK*      INBOUND MATCH*
1/12  ENABLED       TWO WAY      E&M WINK*      INBOUND MATCH*


* Indicates factory default.
 Press Space Bar or Backspace to select then <ENTER>. Press A for All.
 Press Tab, Arrow Keys, <F1> help, <F2> exit, <F3> previous, <F4> next
```

RBS Channel Configuration–Detailed Screen

Channel Configuration—Detailed

Help for Channel Configuration—Detailed

Interface Status: Shows CARD NOT PRESENT, DISABLED, RECEIVING BLUE ALARM, RECEIVING YELLOW ALARM, SYNCHRONIZED, NOT SYNCHRONIZED, or LAYER 2 DOWN. Indicates framing status of digital interface.

DCN: Shows digital interface and channel number.

Operation: Select ENABLED* (default), DISABLED, MAINTENANCE, or EXCLUSIVE. Press A to change all channels starting from the current channel on down. Enabled makes the digital interface channel available for inbound or outbound call. Disabled and maintenance prevents the digital interface channel from processing inbound or outbound call. Exclusive overrides the channel search method and chooses the outgoing channel based on the calling resource. Calls can originate from analog ports, explicit channel dialing (see Analog Port Dialing Configuration), redirected channel of the opposite interface, or the call generator. Use exclusive to emulate outgoing channel bank operation. Exclusive enables channels to receive calls.

Operation implementation depends on whether switch emulation is ISDN, RBS, or GR-303.

ISDN implements operation by sending an ISDN service message containing change status and channel ID information elements. The ISDN service message is sent on the fly and also sent according to the selected update method. Each ISDN service message must be acknowledged by the near end or it will be sent again.

RBS uses operation to put channels in disabled and maintenance mode by transmitting the busy state, keeping the receiving channel in the transition state when the digital interface goes from not synchronized to synchronized. Exclusive mode is implemented by using the appropriate channel. GR-303 uses operation to locally put channels in service, out of service, or in maintenance. Maintenance mode behaves the same as in service. Exclusive mode behaves the same as enabled. GR-303 does not support service messages to remotely enable or disable channels.

Status: Shows IN SERVICE, OUT OF SERVICE, MAINTENANCE, or LOCAL OUT OF SERVICE. In service indicates channel is available for inbound and outbound calls. Out of service and maintenance indicates channel is disabled. Local out of service indicates channel is disabled and can not be enabled remotely using a service message. However, it can be enabled locally. Shown for ISDN digital interfaces only.

Service Update: Select NONE* (default), NETWORK SIDE LAYER 2 STARTUP, USER SIDE LAYER 2 STARTUP, or UPON LAYER 2 STARTUP. This selection determines when ISDN service messages containing the B-channel status are sent across the network. Each interface sends a total of 23 service messages, one for each bearer channel. A service message is also sent independent of this selection whenever the status of an individual channel is changed on the channel configuration detailed screen. None indicates that service messages are not sent across the network. Network side layer 2 startup indicates that service messages are sent upon layer 2 startup provided this interface is configured for network side. User side layer 2 startup indicates that service messages are sent upon layer 2 startup provided this interface is configured for user side. Upon layer 2 startup indicates that service messages are sent upon layer 2 startup. B-channel status may be sent by 4ESS custom, 5ESS custom, and DMS-100 custom. B-channel status is not sent by NI-2 although we have seen Nortel central offices running NI-2 that send B-channel status. Not applicable to GR-303 which never sends B-channel status. Shown for ISDN digital interfaces only.

Direction: Select TWO WAY, OUTBOUND, or INBOUND. Press A for all to set all channels on a single digital interface to the current selection. Two way channels can originate or receive calls. Outbound channels can only originate calls. Inbound channels can only receive calls. Shown for RBS digital interfaces only.

Signaling: Select E&M WINK* (default), E&M IMMEDIATE, GROUND FXO, GROUND FXS, GROUND SAO, GROUND SAS, LOOP FXO, LOOP FXS, LOOP SAO, or LOOP SAS. Press A for all to set all channels on a single digital interface to the current selection. Choices depend on whether interface is configured for network or user. Shown for RBS digital interfaces only. Signaling definitions appear later in this text.

Channel Configuration—Detailed

Help for Channel Configuration—Detailed (continued)

Inbound Routing: Select INBOUND MATCH* (default), ACD 1 through ACD 32, PORT 1 through PORT 32, REDIRECT 1 through REDIRECT 4, MSG 1 PLAY TWICE, MSG 2 PLAY TWICE, CONNECT ACTION 1 through CONNECT ACTION 3, BUSY, or REORDER. Press A for all to set all channels on a single digital interface to the current selection. This selection indicates inbound call routing.

Inbound match routes an incoming call by matching the DNIS-DTMF digits with the match number in the Inbound Match Configuration screen.

ACD 1 through ACD 32 routes an incoming call to an analog port via an ACD group. The ACD group is configured using the ACD Configuration screen. Incoming DNIS-DTMF digits are ignored.

PORT 1 through PORT 32 routes an incoming call to an analog port. Incoming DNIS-DTMF digits are ignored. Use PORT 1 through PORT 24 to emulate incoming channel bank operation.

Redirect 1 through redirect 4 routes an incoming call to an outbound digital interface channel via a redirect group. The redirect group is configured using the Redirect Configuration screen. Incoming DNIS-DTMF digits are ignored.

Msg 1 play twice and msg 2 play twice route incoming calls to the selected voice message. The message is played twice, followed by a forced disconnect. Incoming DNIS-DTMF digits are ignored.

Connect action 1 through connect action 3 routes the call to the connect action on the connect action screen. Incoming DNIS-DTMF digits are ignored. Busy routes an incoming call to the busy progress tone. Incoming DNIS-DTMF digits are ignored.

Reorder routes an incoming call to the reorder progress tone. Incoming DNIS-DTMF digits are ignored.

Inbound-routing shows hyphens (---) when direction is outbound. Shown for RBS digital interfaces only.

There are three types of RBS protocols: E&M, ground start, and loop start. E&M is the preferred method for PBXs. E&M signaling includes E&M WINK and E&M IMMEDIATE. With wink, the receiving side must send an acknowledgment (wink) before dialing can begin. With immediate, dialing can start immediately. Tie trunks use E&M signaling. History buffs note that E&M stands for ear and mouth.

Ground start signaling includes GROUND FXS, GROUND FXO, GROUND SAS, and GROUND SAO. FXS stands for foreign exchange subscriber, and is what the user side sends to FXO. FXO stands for foreign exchange office, and is what the network side sends to FXS. SAS stands for special access subscriber, and is what the user side sends to SAO. SAO stands for special access office, and is what the network side sends to SAS.

Loop start signaling includes LOOP FXS, LOOP FXO, LOOP SAS, and LOOP SAO. Loop start is what a plain old telephone (POTS) uses. The definitions are essentially the same as ground start. Loop start is the least preferred method for PBXs, because older loop start FXO and SAO facilities cannot signal the user to disconnect.

Calling Number Configuration

This screen configures calling party number related parameters. It also configures the called number plan for ISDN digital interfaces and DNIS-DTMF timing parameters for GR-303 and RBS. There are two screens, one for each digital interface. The screens may differ, depending on whether the interface is configured for ISDN, RBS network side, RBS user side, GR-303 IDT, or GR-303 RDT. Digital interface 2 is not used when NFAS signaling is enabled.

```
Gordon Kapes, Inc.    System 930 Telephony Simulator          Screen 3-1

          CALLING NUMBER CONFIGURATION - DIGITAL INTERFACE 1

Calling Number:             --
Calling Number Mode:        NOT SENT*
Calling Presentation:       ALLOWED*
Calling Type of Number:     NATIONAL NUMBER*
Calling Number Plan:        ISDN NUMBERING PLAN*
Called Number Plan:         ISDN NUMBERING PLAN*
Calling Name Method:        DISPLAY IE*
Calling Name Mode:          NOT SENT*
Calling Name:               --
Setup Progress Indicator:   SEND WHEN CALLING NUMBER IS NOT SENT*







* Indicates factory default.

  Enter fixed, sequential, or random number (0-35 digits). Backspace to edit.
           Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

ISDN Calling Number Configuration Screen

```
Gordon Kapes, Inc.    System 930 Telephony Simulator          Screen 3-1

           CALLING NUMBER CONFIGURATION - DIGITAL INTERFACE 1

Calling Number:                     --
Calling Number Mode:                NOT SENT*
Outbound DTMF Timing:               100/100 MS*
Outbound DNIS Calling Prefix:       *
Outbound DNIS Calling Suffix:       *
Outbound DNIS Called Prefix:        --
Outbound DNIS Called Suffix:        --
Inbound DTMF First Digit Timeout:   10* SECONDS
Inbound DTMF Interdigit Timeout:    10* SECONDS








* Indicates factory default.

  Enter fixed, sequential, or random number (0-35 digits). Backspace to edit.
           Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

RBS Network Side Calling Number Configuration Screen

Calling Number Configuration

```
Gordon Kapes, Inc.   System 930 Telephony Simulator         Screen 3-2

          CALLING NUMBER CONFIGURATION - DIGITAL INTERFACE 2

Outbound DTMF Timing:                100/100 MS*
Inbound DTMF First Digit Timeout:    10* SECONDS
Inbound DTMF Interdigit Timeout:     10* SECONDS
Inbound DNIS Calling Number Format:  BETWEEN FIRST AND SECOND ASTERISKS*

* Indicates factory default.

                      Press Space Bar or Backspace
         Press Arrow Keys, <F1> help, <F2> exit, <F3> previous
```

RBS User Side Calling Number Configuration Screen

```
Gordon Kapes, Inc.   System 930 Telephony Simulator         Screen 3-2

          CALLING NUMBER CONFIGURATION - DIGITAL INTERFACE 2

Calling Number:                    --
Calling Number Mode:               NOT SENT*
Calling Presentation:              ALLOWED*
Calling Name Mode:                 NOT SENT*
Calling Name:                      --
Inbound DTMF First Digit Timeout:  10* SECONDS
Inbound DTMF Interdigit Timeout:   10* SECONDS

* Indicates factory default.

  Enter fixed, sequential, or random number (0-35 digits). Backspace to edit.
           Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

GR-303 IDT Calling Number Configuration Screen

Calling Number Configuration

```
Gordon Kapes, Inc.   System 930 Telephony Simulator         Screen 3-2

          CALLING NUMBER CONFIGURATION - DIGITAL INTERFACE 2

Outbound DTMF Timing:                100/100 MS*

GR-303 RDT does not need calling number parameters













* Indicates factory default.

                     Press Space Bar or Backspace
         Press Arrow Keys, <F1> help, <F2> exit, <F3> previous
```

GR-303 RDT Calling Number Configuration Screen

Help for Calling Number Configuration

Calling Number: Enter up to 35 digits and special characters. Digits 0123456789 are allowed. This is the calling party number to be sent when making an outbound call by way of a digital interface. Hyphens (--) indicate no number entered. The factory default is no number entered. The system can send and receive up to 27 digits. Shown for ISDN, RBS network side, and GR-303 IDT only.

Digits [+-] are special characters. Sequential numbers may be created using [min+max]. Random numbers may be created using [min-max]. Min and max represent the minimum and maximum values of the number. A left bracket, plus or minus, and right bracket are required. For example: 55512[00-99] indicates to send a random number between 5551200 and 5551299. The minimum and maximum values are limited to four digits each, may range from 0 through 9999, and do not require the same number of digits. The minimum and maximum values may not contain * or #. Brackets may be repeated, but may not be nested.

Calling Number Mode: Select NOT SENT* (default), SEND CALLING NUMBER, SEND CALLING NUMBER WITH EXT NUMBER OVERLAY, or SEND CALLING NUMBER WITH TWO DIGIT OVERLAY. This selection indicates how the system will send the calling party number when calls are originated from the system. ISDN digital interfaces send the calling number in the ISDN setup message using the calling number IE (information element). The ISDN setup message may send a progress indicator information element indicating that the origination address is non-ISDN. Robbed bit signaling digital interfaces send the calling number using in-band DTMF tones. An asterisk (*) usually separates the called number from the calling number. GR-303 digital interfaces send the calling number using an in-band FSK analog modem on the IDT side. The FSK modem is supplied by the last port on each 938 analog card. This means the last port on each 938 analog card is not available for connection to a telephone handset when either digital interface is set for GR-303 IDT switch emulation. NOT SENT indicates the system does not send the calling party number. SEND CALLING NUMBER indicates the system sends the calling party number. On ISDN digital interfaces the calling party number is sent in the calling party number information element of the ISDN setup message. SEND CALLING NUMBER WITH EXT NUMBER OVERLAY indicates the system sends the calling party

Calling Number Configuration

Help for Calling Number Configuration (continued)

number, replacing the last 3 to 5 digits with the extension number of the analog port that generated the call. If the call is generated by the call generator, the last two digits are replaced by the active call number starting from call 01 through the number of active calls on the Call Generator Configuration screen. Initially active call numbers are assigned in ascending sequential order, but in rolling mode may become randomized as calls disconnect and then become active again. SEND CALLING NUMBER WITH TWO DIGIT OVERLAY indicates the system sends the calling party number, replacing the last two digits with the last two digits of the extension number of the analog port that generated the call. If the call is generated by the call generator, the last two digits are replaced by the active call number starting from call 01 through the number of active calls on the Call Generator Configuration screen. Initially active call numbers are assigned in ascending sequential order, but in rolling mode may become randomized as calls disconnect and then become active again. Shown for ISDN, RBS network side, and GR-303 IDT only.

Calling Presentation: Select ALLOWED* (default) or RESTRICTED. Calling presentation is sent in the calling party number information element of the ISDN setup message. ALLOWED indicates that the called party is allowed to retrieve the calling party number. RESTRICTED indicates that the calling party number is sent, but that only law enforcement may retrieve it. Shown for ISDN and GR-303 IDT only.

Calling Type of Number: Select NATIONAL NUMBER* (default), INTERNATIONAL NUMBER, SUBSCRIBER NUMBER, ABBREVIATED NUMBER, or UNKNOWN NUMBER. Calling type of number is sent in the calling party number information element of the ISDN setup message. NATIONAL NUMBER includes a national area code. Example: 3125551212. INTERNATIONAL NUMBER includes a country code. Example: 443125551212. SUBSCRIBER NUMBER includes a local exchange number. Example: 5551212. ABBREVIATED NUMBER is 3 to 5 digits long. Example: 1212 or 51212. UNKNOWN NUMBER is either not identified or contains prefix digits that are not part of the calling party number. Example: *70 to disable call waiting. Shown for ISDN only.

Calling Number Plan: Select ISDN NUMBERING PLAN* (default), TELEPHONY NUMBERING PLAN, PRIVATE NUMBERING PLAN, or UNKNOWN NUMBERING PLAN. Calling number plan is sent in the calling party number information element of the ISDN setup message. Select ISDN NUMBERING PLAN if the system is emulating a public network. Shown for ISDN only.

Called Number Plan: Select ISDN NUMBERING PLAN* (default), TELEPHONY NUMBERING PLAN, PRIVATE NUMBERING PLAN, or UNKNOWN NUMBERING PLAN. Called number plan is sent in the called party number information element of the ISDN setup message. Select ISDN NUMBERING PLAN if the system is emulating a public network. Shown for ISDN only.

Calling Name Method: DISPLAY IE* (default), or FACILITY IE. This selection indicates how the system will send the calling party name. DISPLAY IE indicates Display IE (information element) is used to send the calling party name. In North America, Display IE is sent in codeset 6 (network specific) because it is a proprietary method. This works for names originating from the network or user side. In Europe, ETSI recommends sending Display IE in codeset 0 (normal). The Display IE is sent from the network to user side only. In either case, calling name is sent in the setup message whether or not the calling number is present or the calling number presentation is restricted. FACILITY IE indicates that Facility IE is used to send the calling party name. This follows the Telcordia GR-1367-CORE recommendation and is available on National and 5ESS Custom PRIs. This is a two-step process. The first step is to send a setup message with a facility IE indicating information following. A facility message with a facility IE containing the calling name is sent at a later time so as not to delay call setup. However, if the calling number is not present a facility IE is sent in the setup message indicating that the name is not available. If the calling number presentation is restricted a facility IE is sent in the setup message indicating that the calling number is restricted. See the calling name tutorial on the main help screen for more details. Shown for ISDN only.

Calling Name Mode: Select NAME* (default), or NAME WITH NUMBER OVERLAY. NOT SENT indicates that the calling party name is not sent. SEND NAME sends the calling name only if the calling number is present and calling number presentation is allowed. SEND NAME WITH CALLING NUMBER APPENDED is similar to SEND NAME except the calling number is appended to the end of the calling name. See the calling name tutorial on the main help screen for more details. Shown for ISDN and GR-303 IDT only.

Calling Number Configuration

Help for Calling Number Configuration (continued)

Calling Name: Enter up to 17 digits. All alphanumeric characters are allowed. This is the calling party name to be sent by way of an ISDN digital interface. Shown for ISDN and GR-303 IDT only.

Setup Progress Indicator: Select SEND WHEN CALLING NUMBER IS NOT SENT* (default) or DO NOT SEND. This selection indicates when the system will send a progress indicator information element in the ISDN setup message indicating that the origination address is non-ISDN. SEND WHEN CALLING NUMBER IS NOT SENT sends the progress indicator when the calling number is not sent. DO NOT SEND indicates the progress indicator is not sent. This selection does not apply to redirected calls which always pass the progress indicator information element from the incoming to the outgoing setup message. Shown for ISDN only.

Outbound DTMF Timing: Select 100/100 MS* (default) or 50/50 MS. Determines on/off time, in milli-seconds, when dialing outbound numbers. Shown for RBS and GR-303 RDT only.

To support different DNIS requirements, the system provides four outbound DNIS fields which may be filled in or blank (--). If blank (--), DNIS will only send the called number. Possible uses for these fields would be to add digits to the called number or send * to separate the calling number from the called number. These fields are sent in the following order:

1. Outbound DNIS Calling Prefix (if calling number is sent)
2. Calling number (if calling number is sent)
3. Outbound DNIS Calling Suffix (if calling number is sent)
4. Outbound DNIS Called Prefix
5. Called number
6. Outbound DNIS Called Suffix

Shown for RBS network side only.

Outbound DNIS Calling Prefix: Enter up to 17 digits. Digits 0123456789*# are allowed. Factory default is *. If calling number is sent, these digits are sent before the calling number. Shown for RBS network side only.

Outbound DNIS Calling Suffix: Enter up to 1 digit. Digits 0123456789*# are allowed. Factory default is *. If calling number is sent, this digit is sent after the calling number. Shown for RBS network side only.

Outbound DNIS Called Prefix: Enter up to 17 digits. Digits 0123456789*# are allowed. These digits are always sent before the called number. Shown for network side RBS only.

Outbound DNIS Called Suffix: Enter up to 17 digits. Digits 0123456789*# are allowed. These digits are always sent after the called number. Shown for RBS network side only.

Inbound DTMF First Digit Timeout: Select 1 through 10* (default) in seconds. If the first DNIS-DTMF digit of the inbound number is not received within configured time, the call is routed to reorder progress tone. Shown for RBS and GR-303 RDT only.

Inbound DTMF Interdigit Timeout: Select 1 through 10* (default) in seconds. Determines maximum time allowed between DNIS-DTMF digits, and how long the system will wait before the inbound number se-quence is considered to be complete. Shown for RBS and GR-303 RDT only.

Inbound Calling Number Method: Select BETWEEN FIRST AND SECOND ASTERISK* (default) or NONE. Between first and second asterisk indicates that DTMF calling number digits are expected to be between the first and second asterisk. The called number follows the second asterisk. None indicates that DTMF calling number digits are not expected and that all incoming DTMF digits are part of the called number. Shown for RBS user side only.

Sidenote: When the digital interface is configured for ISDN and the call is redirected by the system's redirect feature, the original calling party number is sent using the calling party number information element. The system calling party number is sent using the redirecting number information element. This is to assist law enforcement in identifying calls that are redirected through a third party.

Analog Port Configuration—Master

This screen allows configuration of all analog ports. Press <ENTER> then Y to update all analog ports. Use detailed screens to configure individual ports. This screen also sets the audio coding and signaling method used by all analog ports.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator         Screen 4-1

                  ANALOG PORT CONFIGURATION - MASTER

Extension Digit Length:     4*                 ALL DETAILED SCREENS MATCH
Base Extension Number:      1001               ALL DETAILED SCREENS MATCH
ACD Assignment:             ACD 1*             ALL DETAILED SCREENS MATCH
Outbound Access:            ENABLED*           ALL DETAILED SCREENS MATCH
936/938 Signaling Method:   LOOP START*        ALL DETAILED SCREENS MATCH
Call Transfer:              DISABLED*
Audio Compression:          G.711 MU-LAW*
936/938 Ring Fault Action:  CONTINUE RINGING*
Maximum Alert Time:         NONE*
936/938 Receive Loss:       -6dB*
936/938 Reference Tone:     OFF*
936 Analog Caller ID:       SEND TIME/NUMBER/NAME*
936 Private Numbers:        SEND PRIVATE NUMBERS*
936 Alerting Cadence:       1000/3000 MS*
Call Waiting:               DISABLED*

* Indicates factory default.

               Press Space Bar or Backspace then <ENTER>
           Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

Help for Analog Port Configuration—Master

Extension Digit Length: Select 3, 4* (default), or 5. Number of digits assigned to an analog port extension number. Press <ENTER> after selecting the number. Changing the number of digits causes all analog port extensions to be renumbered sequentially starting from the base extension number.

Base Extension Number: Enter a 3, 4, or 5-digit number. Digits 0123456789 are allowed. Starting number used to sequentially number all analog port extensions. The number of digits must agree with the analog port extension digit length. Press <ENTER> after entering the number. The system renumbers all analog port extensions after <ENTER> is pressed. The base extension number is automatically adjusted so as to not conflict with outbound access digits.

If the calling number mode in the Calling Number Configuration screen is set for send calling number with analog port extension overlay, it is recommended that the base extension number be set to match the last few digits of the calling number. If, for example, the assigned calling number in the Calling Number Configuration screen is 3125556600, and the extension digit length is 4, set the base analog port extension number to 6600.

ACD Assignment: Select ACD 1* (default) through ACD 4, CONSECUTIVE ACD, or NONE. Assigns analog ports to an ACD group. The system supports more than four ACD groups, but for ease of use only the first four are assignable from this screen. Consecutive ACD means that analog port 1 is assigned to ACD 1, analog port 2 is assigned to ACD 2, etc., up to the maximum number of ACD groups. Use it when one-to-one mapping between the analog ports and the ACD groups is desired. None restricts all analog ports from ACD assignment.

Analog Port Configuration—Master

Help for Analog Port Configuration—Master (continued)

Outbound Access: Select ENABLED* (default) or DISABLED. Enables or disables analog port access to digital interface channels for making outbound calls. 938 Signaling Method: Select LOOP START* (default) or GROUND START. Supported by 938 analog card only. The 914 OPS card and 939 analog card use loop start only. This assigns the signaling method used by the analog ports. Loop start is used by plain old telephones (POTS). Ground start is used by trunks.

Call Transfer: Select DISABLED* (default), INTERNAL ONLY, or INTERNAL & OUTSIDE. Internal only allows incoming or outgoing calls to be transferred to an analog extension. Call transfer is activated by depressing the receiver hook for one second. Upon releasing the receiver hook, a stutter dial tone is sent, indicating the call is on hold and the system is ready to accept dialed digits of an analog port extension number. The user dials the extension, hangs up, and the call is transferred to that extension. Enabling call transfer increases analog on-hook detection time from 0.4 second to 1.4 seconds. If the station user disconnects while a party is still on hold, the station user is automatically rung back and, upon answer, is connected to the held party. Internal & outside allows incoming or outgoing calls to be transferred to an analog extension or an outside call. When making an outside call, signaling is accomplished by sending a switchhook flash across the digital interface. If the network side is configured to accept outside call transfer, the network returns a dial tone and accept dialed digits. If the network side is not configured to accept outside call transfer, the network will probably disconnect the call. This feature works only with T1 Robbed Bit Signaling and GR-303 and only from the user (RDT) to network (IDT) direction. Internal and outside increases robbed bit signaling disconnect time from 0.3 seconds to 1.4 seconds. Outside call transfer works only when the transferred call is routed to ACD, port, message, busy, and reorder on the Inbound Match Configuration screen. All other choices are routed to reorder. Outside call transfer is also known as DID two way call transfer or Centrex call transfer. Calls initiated by the call generator does not respond to call transfer due to its complexity. This feature is not selectable from detailed screens.

Audio Compression: Select G.711 MU-LAW* (default) or G.711 A-LAW. Supported by 938 and 939 analog cards only. The 914 OPS card uses mu-law compression only. Mu-law is normally used with T1. A-law is normally used with E1. This setting selects the audio compression used by the analog ports. ISDN sends a code representing the compression method in the bearer capability information element. This selection has no effect on voice messages or the audio monitor which use mu-law compression on T1 systems. This feature is not selectable from detailed screens.

938 Ring Fault Action: Select CONTINUE RINGING* (default), SEND REORDER, or REMOVE FROM ACD. Action to be taken when analog port fails to sense ring current during alerting. This condition occurs when no device is connected to the analog port. Supported by 938 analog card only. The 914 OPS card does not sense ring current. The 939 analog card contains FXS circuits and does not use this selection. Continue ringing indicates that an analog port will continue ringing even though a current load is not detected. Send reorder indicates that the caller is sent reorder progress tone when ring current is not detected. Remove from ACD indicates that an analog port is temporarily removed from ACD selection when ring current is not detected. In addition, the caller is sent reorder progress tone. Once an analog port is removed from ACD selection it is automatically reinstated upon power up, reboot, or whenever that analog port goes off-hook. This feature is not selectable from detailed screens.

Maximum Alert Time: Select NONE* (default), or 15 SECONDS through 120 SECONDS in 15 second increments. Amount of time analog port should alert (ring) before returning reorder progress tone. None means there is no maximum time. Applies to calls originated from analog ports as well as from digital interfaces. If a digital interface uses ISDN, returns disconnect message with cause code 19, user alerted no answer. This feature is not selectable from detailed screens.

938 Receive Loss: Select –6dB* (default) or 0dB. –6dB reduces the audio level sent to all 938 card analog ports. 0dB sends the audio level unattenuated. 939 analog cards are selected on a different screen. This feature is preset to 0dB on 914 cards, older 938 card firmware (revision 1.02 and prior), and is not selectable from detailed screens.

Analog Port Configuration—Master

Help for Analog Port Configuration—Master (continued)

938 Reference Tone: Select OFF* (default) or ON. Sends a 1 kHz 1 milliwatt (0dB) reference tone to all 938 card analog ports. 939 analog ports are selected on a different screen. This feature is not available on 914 cards, older 938 card firmware (revision 1.02 and prior), and is not selectable from detailed screens.

938 Analog Caller ID: Select SEND TIME/NUMBER/NAME* (default), SEND SINGLE DATA MESSAGE FORMAT, DO NOT SEND, SEND NUMBER, SEND NUMBER/NAME, SEND TIME/NUMBER, or NOT LICENSED. Supported by 938 analog card only. Most of the selections use MDMF (Multiple Data Message Format) to send time, calling party number, and calling party name. The exception is SDMF (Single Data Message Format) which is an obsolete method for sending time and calling party number. Do not send disables analog Caller ID. Not licensed indicates the system is not licensed for analog Caller ID.

938 Private Numbers: Select SEND PRIVATE NUMBERS* (default) or DO NOT SEND PRIVATE NUMBERS. Send private numbers allows the calling party number to be sent, regardless of the caller's wishes. Used by law enforcement. Do not send private numbers sends a private indicator instead of the calling party number when the sender wishes to remain private. Supported by 938 analog card only.

938 Alerting Cadence: Select 1000/3000 MS* (default) or 400/200/400/3000 MS. 1000/3000 ms causes the analog port to alert for one second on and three seconds off. 400/200/400/3000 ms alerts for 400 ms on, 200 ms off, 400 ms on, and three seconds off. Also known as English ringing. Supported by 938 analog card only. The 938 analog card always alerts for one second on and three seconds off.

Call Waiting: Select DISABLED (default) or INTERNAL ONLY. Disabled sends reorder to incoming calls attempting to connect to an active analog extension. Internal only allows incoming calls from analog extensions to wait for a specific analog extension to become available. Call Waiting is a terminal feature which operates on the terminating portion of a call. The Call Waiting feature gives a station user, engaged in a telephone conversation, an audible 440Hz alert tone for 0.3 seconds indicating that an incoming call is attempting to terminate to their line. The station user that receives the audible alert tone can flash the switchhook for one second to put the first call on hold and answer the incoming call or can go on hook and allow the terminal to ring and answer the incoming call. The alert tone is only audible to the Call Waited station user; the originating station has audible ringing. The Call Waiting feature also provides a hold function that is activated by a switchhook flash. Consecutive flashes allow the station user to alternately talk to the original caller or to new calling party. If the station user disconnects while a party is still on hold, the station user is automatically rung back and, upon answer, is connected to the held party. Enabling this feature increases analog on-hook detection time from 0.4 second to 1.4 seconds. This feature is not selectable from detailed screens.

After some selections the screen displays ALL DETAILED SCREENS MATCH, SOME DETAILED SCREENS MATCH, or NO DETAILED SCREENS MATCH. This indicates whether the selection matches the same field on the detailed screens.

914 OPS Card Description
Older card containing eight FXO loop start circuits.

938 Analog Card Description
Contains eight FXO loop start or ground start circuits.

939 Analog Card Description
Emulates eight FXS (subscriber) loop start circuits.

Analog Port Configuration—Detailed

This screen configures individual analog ports for extension number, ACD group assignment, and outbound access. It also displays the port's current state. There are four detailed screens, one for every eight analog ports.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 4-2

                  ANALOG PORT CONFIGURATION - DETAILED

Card 1 Present:   YES - 938 CARD

Port   Ext     ACD     Outbound  Sig   Connect   Hold    State
 1     1001    ACD 1*  ENABLED*  LS*   --        --      01-ON HOOK
 2     1002    ACD 1*  ENABLED*  LS*   --        --      01-ON HOOK
 3     1003    ACD 1*  ENABLED*  LS*   --        --      01-ON HOOK
 4     1004    ACD 1*  ENABLED*  LS*   --        --      01-ON HOOK
 5     1005    ACD 1*  ENABLED*  LS*   --        --      01-ON HOOK
 6     1006    ACD 1*  ENABLED*  LS*   --        --      01-ON HOOK
 7     1007    ACD 1*  ENABLED*  LS*   --        --      01-ON HOOK
 8     1008    ACD 1*  ENABLED*  LS*   --        --      01-ON HOOK




* Indicates factory default.

     Enter extension (1000-9999) then <ENTER>. Backspace to edit.
Press Tab, Arrow Keys, <F1> help, <F2> exit, <F3> previous, <F4> next
```

Help for Analog Port Configuration—Detailed

Card Present: Shows YES – 938 ANALOG CARD, YES – 939 CARD, YES – 914 CARD, or NO. The 938 analog card drives eight analog loop/ground start FXO circuits. The 939 card emulates eight analog loop start FXS stations. The 914 card drives eight analog loop start line circuits. It has been discontinued but is supported by this software.

Port: Shows 1 through 32. Identifies the internal hardware number of each analog port.

Ext: Shows the extension number associated with the analog port. If a different extension number is desired, enter it here and then press <ENTER>. Numbers that are not unique, or are outside the permissible range, are automatically disallowed. Outbound access digits and analog port extensions must not conflict with each other.

ACD: Select ACD 1* (default) through ACD 32, or NONE. Assigns the analog port to an ACD group. NONE restricts the analog port from being part of an ACD group.

Outbound: Select ENABLED* (default) or DISABLED. Enables or disables analog port access to digital interface channels for making outbound calls.

Sig: Select LS* (default) or GS. This assigns the signaling method used by the analog port. Supported by 938 analog card only. The 914 OPS card uses loop start only. Loop start (LS) is used by plain old telephones (POTS). Ground start (GS) is used by trunks.

Connect: Shows ACT, DCN, GEN, MSG, or PORT. Shows resource connected to analog port. DCN stands for digital channel number. GEN stands for call generator. MSG stands for recorder/announcer message number. PORT stands for analog port number. Hyphens (--) indicate no connection.

Analog Port Configuration—Detailed

Help for Analog Port Configuration—Detailed (continued)

Hold: Shows DCN or PORT. Shows resource placed on hold by the analog port using flashhook. Hyphens (--) indicate no connection.

State: Shows analog port internal state number followed by description: ALERTING, AUDIBLE RING, BUSY TONE, CONNECT, DIAL TONE, DIALING, FSK MODEM, HOWLER TONE, MSG DIAL TONE, MSG DIALING, MSG PROGRAM, MSG PLAYBACK, MSG RECORD, RING FAULT, ON HOLD, ON HOOK, PROGRESS TONE, REORDER TONE, SILENCE, STUTTER TONE, VACANT TONE, or WAIT ACCESS (wait for access to a digital interface channel). Indicates active condition. Hyphens (--) indicate analog card is not present.

Recorder/Announcer

This screen allows the recorded messages to be erased. It also shows the real-time status of the voice messages. Note: A 938 analog card is required to record voice messages.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator           Screen 5

                          RECORDER/ANNOUNCER

Erase voice messages:  NO*

Message 1 Status:  FF - IDLE
Message 2 Status:  FF - IDLE


Use any analog port to play or record voice messages.
Go off hook and DTMF dial *99.
Dial 1 to play message 1.
Dial 2 to play message 2.
Dial 3 to record message 1.
Dial 4 to record message 2.




* Indicates factory default.

                       Press Y or N then <ENTER>
                       Press <F1> help, <F2> exit
```

Help for Recorder/Announcer

Erase voice messages: Select NO* (default) or YES. Select Y, then press <ENTER> to erase all voice messages.

Message Status: Shows hexadecimal representation of digitally encoded voice followed by the current state: PLAY, RECORD, PAUSE, or IDLE. Hexadecimal 7F or FF is the normal state when a voice signal is not present.

Use any analog port to play or record voice messages. Go off hook and DTMF dial *99. A periodic beep, once every second, indicates connection to the recorder/announcer. Four commands are available:
Dial 1 to play message 1.
Dial 2 to play message 2.
Dial 3 to record message 1.
Dial 4 to record message 2.

Hang up to stop recording.

Recorded messages can be as long as 20 seconds. Hang up to disconnect from the recorder/announcer. Messages cannot be recorded while the recorder/announcer is playing a message.

Inbound Match Configuration

This screen configures inbound routing for calls received on both digital interfaces. If preset is configured to shared by both interfaces, there are a total of four screens which are shared by both digital interfaces. If preset is configured to separate for each interface, there are a total of eight screens, four for each digital interface. Each screen contains 12 match numbers.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 6-1

       INBOUND MATCH CONFIGURATION - SHARED BY BOTH INTERFACES

No.  Match Number                             Route if Match
 1.  --                                       ACD 1*
 2.  --                                       ACD 1*
 3.  --                                       ACD 1*
 4.  --                                       ACD 1*
 5.  --                                       ACD 1*
 6.  --                                       ACD 1*
 7.  --                                       ACD 1*
 8.  --                                       ACD 1*
 9.  --                                       ACD 1*
10.  [2-9]NNNNNN                              ACD 1*
11.  [0-1][2-9]NNNNNNNNN                      ACD 1*
12.  --                                       ACD 1*

Preset:  SHARED BY BOTH INTERFACES*

* Indicates factory default.

  Enter match number (0-31 digits) or match range. Backspace to edit.
         Press Tab, Arrow Keys, <F1> help, <F2> exit, <F4> next
```

Help for Inbound Match Configuration

When the digital interface is configured for RBS, the system uses this screen only when routing is set to inbound match on the Channel Configuration screen. The system routes calls by comparing the called number with the list of match numbers.

When the digital interface is configured for ISDN, the system routes incoming ISDN calls by comparing the called number with the list of match numbers. The list is scanned starting from match number 1 until a successful match is found. If the system does not find a match, it connects the call to reorder tone for 30 seconds, followed by a forced disconnect. When the digital interface is configured for RBS network side or GR-303 IDT, the system uses one of eight DTMF receivers for receiving DNIS-DTMF digits. These receivers are shared by the digital interface channels on a first available basis. If a DTMF receiver is available, the caller is sent dial tone. Only E&M wink, ground start FXO, ground start SAO signaling and GR-303 provide handshaking to indicate that a DTMF receiver is available.

Match Number: Enter up to 31 digits and special characters. Digits 0123456789*# are allowed. Indicates the called number to be matched. Digits NXZ[-] are special characters. N matches any number from 0 through 9. Example: 9NN matches any number from 900 through 999. X matches any digit from 0 through #. Example: *XX matches any 2 digits after *. Z creates a match for calls containing no called number. No called number is permitted by switch emulations such as Euro ISDN, DMS100, T1 Robbed Bit Signaling, and GR-303. Otherwise incoming calls containing no called number will be rejected. Match ranges are created using [min-max]. A left bracket, hyphen (-), and a right bracket are required. Example: [234-789] matches any number from 234 through 789. Minimum and maximum values are limited only

Inbound Match Configuration

Help for Inbound Match Configuration (continued)

by the line length but must contain the same number of digits. Minimum and maximum values may not contain * or #. Brackets may be repeated, but not nested. Two hyphens (--) indicate that no match number is specified. GR-303 RDT uses a four digit terminal number as the match number. Terminal numbers range from 0001 to 2048. The terminal number is the call reference value (CRV). Example: To route CRV 0010 to analog port 10, enter 0010 as the match number and port 10 for route if match.

Route if Match: Select ACD 1 through ACD 32, PORT 1 through PORT 32, REDIRECT 1 through REDIRECT 4, MESSAGE 1 PLAY TWICE, MESSAGE 2 PLAY TWICE, MESSAGE 1 PLAY CONTINUOUS, MESSAGE 2 PLAY CONTINUOUS, CONNECT ACTION 1 through CONNECT ACTION 3, BUSY, or REORDER. ACD routes the call to the specified ACD group. Redirect routes the call to the specified redirect group. Port routes the call to the specified analog port. Message 1 play twice and message 2 play twice routes the call to the specified voice message, plays the message twice, and followed by a forced disconnect. Message 1 play continuous and message 2 play continuous routes the call to the specified voice message and plays it until the caller disconnects. Connect action routes the call to the connect action on the connect action screen. Busy routes the call to busy progress tone for 30 seconds, followed by a forced disconnect. Reorder routes the call to reorder (fast busy) tone for 30 seconds, followed by a forced disconnect.

Preset: Select SHARED BY BOTH INTERFACES* (default), SEPARATE FOR EACH INTERFACE, NORTH AMERICAN NUMBERING PLAN, PROGRESSIVE DIGITS, CONSECUTIVE EXTENSION ACD, CONSECUTIVE EXTENSION PORTS, CONSECUTIVE GR-303 USER TERMINAL PORTS, MATCH FOR NO NUMBER, 3 DIGIT DIALING PLAN through 11 DIGIT DIALING PLAN, ANY DIGIT MATCH, or COPY TO OTHER INTERFACE. Use preset to determine whether there is a common or separate inbound match configuration for each digital interface. It is also a means to quickly configure several useful inbound match configurations. Shared by both interfaces indicates that a single configuration is used by both digital interfaces. Separate for each interface indicates that there is a separate configuration for each digital interface. North American Numbering Plan routes seven digit numbers starting with 2 through 9, and eleven digit numbers whose first digit is 0 or 1 and second digit is 2 through 9 to ACD 1. Progressive digits route numbers of various lengths to ACD 1. The first digit indicates the digit length. Consecutive extension ACD route analog port extension numbers to consecutive ACD groups. Consecutive GR-303 terminal ports route 0001 through 0032 to consecutive analog ports. It is useful for GR-303 RDT. Match for no number routes a call lacking a called number to ACD 1. 3 digit dialing plan routes any three digit match to ACD 1. 4 digit dialing plan routes any four digit match to ACD 1, etc. Any digit match creates a list of matches to ACD 1. Copy to other interface copies all inbound match parameters from the current interface to the other interface.

Analog Port Dialing Configuration

This screen configures analog port dialing features.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator          Screen 7

                  ANALOG PORT DIALING CONFIGURATION

Dial # Mode:                           END OF NUMBER*
Outbound First Digit Timeout:          10* SECONDS
Outbound Interdigit Timeout:           10* SECONDS
Outbound Access Mode:                  DIGIT REQUIRED - DESIGNATED INTERFACE *
Interface 1 Access Digit:              9*
Interface 2 Access Digit:              8*
Interface 1 Explicit Channel Access:   *1
Interface 2 Explicit Channel Access:   *2
Recorder/Announcer Access Digits:      *99
Interface 1 Dialing Method:            EN-BLOC*
Interface 2 Dialing Method:            EN-BLOC*





* Indicates factory default.

                Press Space Bar or Backspace to select
                Press Arrow Keys, <F1> help, <F2> exit
```

Help for Analog Port Dialing Configuration

Dial # Mode: Select END OF NUMBER* (default) or PASS THROUGH. Action taken when DTMF # has been dialed on an outbound call. End of number speeds up outbound call processing by immediately processing the dialed number. The # is stripped. The remaining digits are matched with the prefix number on the analog port outbound call configuration screen. Maximum length is ignored. Pass through enables # to be included in the dialed number but does not speed up outbound call processing.

Outbound First Digit Timeout: Select 1 through 10* (default). The maximum time in seconds that the system waits to detect the first digit in the outbound number, after the outbound access digit has been dialed. When this time expires, the caller is sent reorder progress tone.

Outbound Interdigit Timeout: Select 1 through 10* (default). The maximum time in seconds that the system waits to detect interdigit numbers in the outbound number. When this time expires, the system processes the number as dialed.

Outbound Access Mode: Select DIGIT REQUIRED – DESIGNATED INTERFACE* (default), DIGIT REQUIRED – ALTERNATE INTERFACES. IMMEDIATE ACCESS TO INTERFACE 1, IMMEDIATE ACCESS TO INTERFACE 2, IMMEDIATE ACCESS ALTERNATE INTERFACES, or IMMEDIATE ACCESS TO BOTH INTERFACES. DIGIT REQUIRED – DESIGNATED INTERFACE indicates that an access digit must be dialed by a device connected to the analog port to access a digital interface. If an access digit is not dialed, only analog port extension numbers and recorder/announcer can be dialed. DIGIT REQUIRED – ALTERNATE INTERFACES indicates that an access digit must be dialed by a device connected to the analog port to access a digital interface. The selected interface alternates between interface 1 and 2 regardless of the dialed access digit. If an access digit is not dialed, only analog port extension numbers and recorder/announcer can be dialed. IMMEDIATE ACCESS TO INTERFACE indicates that calls are sent to the selected digital interface without dialing an access digit. Analog port extension numbers cannot be dialed, but calls can be routed to ACD groups from Analog Port Outbound Call Configuration screen or Inbound Call Configuration screen. Recorder/Announcer cannot be dialed, but calls can be routed to voice messages from Analog Port Outbound Call Configuration

Analog Port Dialing Configuration

Help for Analog Port Dialing Configuration (continued)

screen or Inbound Call Configuration screen. In all cases, calls sent to a digital interface are routed according to the Analog Port Outbound Call Configuration screen. IMMEDIATE ACCESS ALTERNATE INTERFACES indicates that the interface alternates between interface 1 and 2. IMMEDIATE ACCESS TO BOTH INTERFACES is shown when NFAS signaling is enabled. The factory default for GR-303 is immediate access.

Interface 1 Access Digit: Select 0 through 9 or NOT AVAILABLE. Factory default is DTMF 9. Digit dialed on an analog port to access digital interface 1. Functional only when access digit is required. NOT AVAILABLE indicates this feature is not provided. The system will match the remaining digits dialed with the Analog Port Outbound Call Configuration screen to determine when the dialed number is complete. Shows DISABLED BY IMMEDIATE ACCESS when outbound access mode is immediate access.

Interface 2 Access Digit: Select 0 through 9 or NOT AVAILABLE. Factory default is DTMF 8. Digit dialed on an analog port to access digital interface 2. Functional only when access digit is required. NOT AVAILABLE indicates this feature is not provided. The system will match the remaining digits dialed with the Analog Port Outbound Call Configuration screen to determine when the dialed number is complete. Shows DISABLED BY IMMEDIATE ACCESS when outbound access mode is immediate access. Shows DISABLED BY NFAS SIGNALING when NFAS signaling is enabled.

Interface 1 Explicit Channel Access: Factory preset to DTMF *1. Digit dialed on an analog port to access a specific channel on digital interface 1. Functional only when access digit is required. Two digits must follow *1, representing the channel number. For example, dial *102 to access digital interface 1, channel 2. A leading 0 is required for channel numbers 1 through 9. The system will match the remaining digits dialed with the Analog Port Outbound Call Configuration screen to determine when the dialed number is complete. Shows DISABLED BY IMMEDIATE ACCESS when outbound access mode is immediate access. This feature does not work correctly with GR-303 RDT because channel selection is done on the network side.

Interface 2 Explicit Channel Access: Factory preset to DTMF *2. Similar to interface 1 explicit channel access but for interface 2. For example, dial *203 to access digital interface 2, channel 3.

Recorder/Announcer Access Digits: Factory preset to *99. Digits dialed on an analog port to record or playback voice messages. Functional only when access digit is required. Shows DISABLED BY IMMEDIATE ACCESS when outbound access mode is immediate access.

Interface 1 Dialing Method: Select OVERLAP* (default) or EN-BLOC. Dialing method used by analog port on digital interface 1. Overlap sends the dialed digits one digit at time. Dial tone is supplied by near end. The called number is not processed by the Analog Port Outbound Call Configuration screen. ISDN uses information messages to send digits. GR-303 RDT and Robbed Bit Signaling use DTMF tones.

En-bloc sends the dialed digits all at once. Dial tone is supplied locally. The called number is processed by the Analog Port Outbound Call Configuration screen and sent when the called number is complete. ISDN sends the called number using the setup message. GR-303 RDT and Robbed Bit Signaling use DTMF tones. GR-303 IDT converts the last four digits to a terminal number and sends the terminal number in the ISDN call reference field. No called number is permitted with DMS100 custom. Preset to EN-BLOC* (default) when restricted by switch emulation.

Interface 2 Dialing Method: Select OVERLAP* (default) or EN-BLOC. Similar to interface 1 dialing method but for interface 2.

Analog Port Outbound Call Configuration

This screen configures how the system will interpret outbound number dialing from the analog ports. There are four configuration screens, two for each digital interface. Each screen contains 12 match configurations. The screens may differ, depending on whether the interface is configured for ISDN or RBS.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator         Screen 8-1

    ANALOG PORT OUTBOUND CALL CONFIGURATION - DIGITAL INTERFACE 1

No.  Prefix                        Max  Strip   Type  Route if Match
 1.  --                            --   --       -    --
 2.  --                            --   --       -    --
 3.  311                           3    0        N    SEND OUT*
 4.  911                           3    0        N    SEND OUT*
 5.  [2-9]NNNNNN                   7    0        S    SEND OUT*
 6.  011                           17   3        I    SEND OUT*
 7.  --                            --   --       -    --
 8.  --                            --   --       -    --
 9.  0[2-9]NNNNNN                  11   0        S    SEND OUT*
10.  1[2-9]NNNNNNNNN               11   0        S    SEND OUT*
11.  *                             31   0        U    SEND OUT*
12.  0                             31   0        U    SEND OUT*

Preset:  --

* Indicates factory default.

           Enter number (0-31 digits). Backspace to edit.
        Press Tab, Arrow Keys, <F1> help, <F2> exit, <F4> next
```

Help for Analog Port Outbound Call Configuration

This screen is used when the dialed number is sent en-bloc. This screen is not used when the interface uses overlap dialing. This screen is not used by the call generator. GR-303 IDT sends the last four digits as the terminal number using the ISDN call reference field. Terminal numbers may range from 1 to 2048. This screen is used to determine when the caller has finished dialing, after which the call is sent to the indicated route. After an outbound access digit has been dialed, the system collects the dialed digits until a combination that matches prefix, max length, and strip length is found. The list is scanned starting from line one until a successful match is found. If a match is not found, the call is not sent out and caller is sent vacant tone. Prefix: Enter up to 31 digits and special characters. Digits 0123456789*# are allowed. These are the digits to be compared with the leading digits of the dialed number to determine a match. Digits NXZ[-] are special characters. N matches any number from 0 through 9. Example: 9NN matches any number from 900 through 999. X matches any number from 0 through #. Example: *XX matches any 2 digits after *. Z creates a match for calls containing no called number. No called number is permitted by switch emulations such as DMS100, T1 Robbed Bit Signaling, and GR-303. Otherwise outbound calls containing no called number will be sent out but rejected on the inbound side. Prefix ranges are also be created using [min-max]. Min and max represent the minimum and maximum values of the prefix range. A left bracket, hyphen (-), and a right bracket are required. Example: [234-789] matches any number from 234 through 789. Minimum and maximum values are limited only by the line length but must contain the same number of digits. Minimum and maximum values may not contain * or #. Brackets may be repeated, but not nested. Hyphens (--) indicate no number has been specified.

Analog Port Outbound Call Configuration

Help for Analog Port Outbound Call Configuration (continued)

Max: Select 1 through 31. The maximum number of digits (after the outbound access digit) that the caller must dial before a match is automatically sent to the indicated route. The system ignores this number when an interdigit timeout occurs or when the caller dials #, and # is configured as a immediate outdial on the Analog Port Dialing Configuration screen. Hyphens (--) indicate no prefix entry has been made.

Strip: Select 0 through 31. The number of digits that the system removes from the beginning of the dialed number, after the access digit, to isolate and send out the desired called number. Hyphens (--) indicate no strip entry has been made.

Type: Select N (default), I, S, A, or U. Indicates the called number type used to dial the called party. N indicates national: a number that includes a national area code. Example: 3125551212. I indicates international: a number that includes a country code. Example: 443125551212. S indicates subscriber: a number that includes a local exchange number. Example: 5551212. A indicates abbreviated: a number that is 3 to 5 digits long. Example: 1212 or 51212. U indicates unknown: a number that is either not identified or contains prefix digits that are not part of the called number. Example: *70 to disable call waiting. This information is sent in the called party number information element of the ISDN setup message. Not used by GR-303 which always sends the called number type as unknown. Shown for ISDN digital interfaces only.

Route if Match: Select SEND OUT* (default), REORDER, BUSY, MSG 1 PLAY CONTINUOUS, MSG 2 PLAY CONTINUOUS, CONNECT ACTION 1 through CONNECT ACTION 3, or ACD 1 through ACD 32. The action the system takes if a number match is made. Send out sends the dialed number as the called number to the digital interface. Reorder and busy are provided as a means of blocking specific called numbers. If a match is made, the caller is sent reorder or busy progress tone and the outbound call is not processed. Msg 1 play continuous and msg 2 play continuous routes the call to the specified voice message and plays it until the caller disconnects. Connect action 1 through connect action 3 routes the call to the specified connect action. ACD routes the call to the specified ACD group. Hyphens (--) indicate no prefix entry has been made.

Preset: Select NORTH AMERICAN NUMBERING PLAN* (default), PROGRESSIVE DIGITS, or COPY TO OTHER INTERFACE. Use preset to set up examples of outbound match configurations to filter outgoing digits. Hyphens (--) do nothing. North American Numbering Plan allow the following to be sent: 311, 911, seven digit numbers starting with 2 through 9, eleven digit numbers starting with 1, 17 digit numbers starting with 011, and 31 digit numbers starting with 0 or *. Progressive digits allow numbers whose first digit indicates the number of digits to be sent. Copy to other interface copies all outbound match parameters from the current interface to the other interface.

ACD Configuration—Master

This screen allows configuration of ACD parameters for all 32 ACD groups. Press <ENTER> then Y to update the ACD parameter for all ACD groups. Use detailed screens to configure individual ACD groups.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 9-1

                      ACD CONFIGURATION - MASTER

Queue Depth:      0*                           ALL DETAILED SCREENS MATCH
Queue Action:     AUDIBLE RING*                ALL DETAILED SCREENS MATCH
Overflow Action:  BUSY TONE*                   ALL DETAILED SCREENS MATCH
Hunt Method:      ASCENDING SEQUENTIAL*











* Indicates factory default.

          Press Space Bar or Backspace to select then <ENTER>
    Press Arrow Keys, <F1> help, <F2> exit, <F3> previous, <F4> next
```

Help for ACD Configuration—Master

Incoming calls are routed through ACD groups to analog ports. If an analog port is not available, the incoming call can be placed in queue. Queued calls are connected to the next available analog port assigned to the ACD group. While a call is in queue, queue action provides either audible ring, continuous message playback, or play message twice. If the queue becomes full, queue overflow action causes additional calls to hear busy progress tone, or connect to a voice message. If no analog ports are assigned to a specific ACD group, calls routed to that ACD group are connected to reorder progress tone.

Queue Depth: Select 0* (default) through 32. Indicates the maximum number of calls that can be queued. 0 indicates that no calls can be queued.

Queue Action: Select AUDIBLE RING* (default), MESSAGE 1 CONTINUOUS, MESSAGE 2 CONTINUOUS, MESSAGE 1 PLAY TWICE, or MESSAGE 2 PLAY TWICE. Message 1 continuous and message 2 continuous routes the call to the selected voice message until an analog port becomes available. Message 1 play twice and message 2 play twice routes the call to silence after the voice message has played twice.

Overflow Action: Select BUSY TONE* (default), MESSAGE 1 PLAY TWICE, or MESSAGE 2 PLAY TWICE, or REDIRECT 1-4. Message 1 play twice and message 2 play twice does a forced disconnect after the selected message has played twice. Redirect routes calls to the associated redirect group.

Hunt Method: Select ASCENDING SEQUENTIAL* (default), or CLOCKWISE CIRCULAR. Determines the method used for routing inbound calls to analog ports assigned to ACD groups. Ascending sequential uses the lowest available port number. Clockwise circular assigns port numbers in a clockwise circular pattern. This feature is not selectable from detailed screens.

ACD Configuration—Master

Help for ACD Configuration—Master (continued)

After most selections the screen displays ALL DETAILED SCREENS MATCH, SOME DETAILED SCREENS MATCH, or NO DETAILED SCREENS MATCH. This indicates whether the selection matches the same field on the detailed screens.

ACD Configuration—Detailed

This screen configures individual ACD groups. There are 32 detailed screens, one for each ACD group.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 9-2

                   ACD 1 CONFIGURATION - DETAILED

Queue Depth:       0*
Queue Action:      AUDIBLE RING*
Overflow Action:   BUSY TONE*

Channels Assigned by Interface 1:  23
Channels Assigned by Interface 2:  23
Analog Ports Assigned:             PORT-1,2,3,4,5,6,7,8


Number of Queued Calls:    0
Number of Overflow Calls:  0





* Indicates factory default.

               Press Space Bar or Backspace to select
  Press Arrow Keys, <F1> help, <F2> exit, <F3> previous, <F4> next
```

Help for ACD Configuration—Detailed

Incoming calls are routed through ACD groups to analog ports. If an analog port is not available, the incoming call can be placed in queue. Queued calls are connected to the next available analog port assigned to the ACD group. While a call is in queue, queue action provides either audible ring, continuous message playback, or play message twice. If the queue becomes full, queue overflow action causes additional calls to hear busy progress tone, or connect to a voice message.

Queue Depth: Select 0* (default) through 32. Indicates the maximum number of calls that can be queued. 0 indicates that no calls can be queued.

Queue Action: Select AUDIBLE RING* (default), MESSAGE 1 CONTINUOUS, MESSAGE 2 CONTINUOUS, MESSAGE 1 PLAY TWICE, or MESSAGE 2 PLAY TWICE. Message 1 continuous and message 2 continuous routes the call to the selected voice message until an analog port becomes available. Message 1 play twice and message 2 play twice routes the call to silence after the voice message has played twice.

Overflow Action: Select BUSY TONE* (default), MESSAGE 1 PLAY TWICE, MESSAGE 2 PLAY TWICE, or REDIRECT 1-4. Message 1 play twice and message 2 play twice does a forced disconnect after the selected message has played twice. Redirect routes the call to the specified redirect group.

Channels Assigned by Interface: Shows the total number of channels assigned to this ACD group by each digital interface. When the digital interface is configured for ISDN, inbound matches are assigned on the Inbound Match Configuration screen. When the digital interface is configured for RBS, inbound matches are assigned on the Channel Configuration screen. When the digital interface is configured for RBS, and DNIS routing is assigned to a channel, inbound matches must also be assigned on the Inbound Match Configuration screen.

ACD Configuration—Detailed

Help for ACD Configuration—Detailed (continued)

Analog Ports Assigned: This is a static display that shows PORT followed by the port number of all ports assigned to this ACD group. Hyphens (---) indicate no ports are assigned. Analog ports are assigned on the Analog Port Configuration screen.

Number of Queued Calls: This is a real-time display that shows the number of inbound calls currently waiting in the ACD queue.

Number of Overflow Calls: This is a real-time display that shows the number of inbound calls currently routed to overflow action.

Redirect Configuration—Master

This screen allows configuration of redirect parameters for all four redirect groups. Press <ENTER> then Y to update the redirect parameter for all redirect groups. Use detailed screens to configure individual redirect groups.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 10-1

                   REDIRECT CONFIGURATION - MASTER

Prefix Number:   --                           ALL DETAILED SCREENS MATCH
Inbound Number:  INSERT ENTIRE NUMBER*        ALL DETAILED SCREENS MATCH
Type of Number:  UNKNOWN NUMBER*              ALL DETAILED SCREENS MATCH
Action:          IMMEDIATE*                   ALL DETAILED SCREENS MATCH
Destination:     OPPOSITE DIGITAL INTERFACE*  ALL DETAILED SCREENS MATCH











* Indicates factory default.

 Enter fixed, sequential, or random number (0-31 digits) then <ENTER>.
           Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

Help for Redirect Configuration—Master

Redirect causes an incoming call to initiate an outbound call on a digital interface, dial a specified number, and then connect both calls together. If desired, the incoming call can be connected to a voice message first, after which the outbound call is initiated.

Sidenote: ISDN calls redirected by the system's redirect facility copy the following information elements from incoming to outgoing call: bearer capability, display, calling party (caller), high layer compatibility, low layer compatibility, progress indicator, transit network selection, and user-user. The channel ID is not copied, but must be recalculated. The number of outgoing channels is determined from the bearer capability information transfer rate. The called number may be copied or replaced as configured by prefix number, inbound number, and type of number. The redirecting number information element is sent using the system's calling number. This is to assist law enforcement in identifying calls redirected through a third party.

Prefix Number: Enter up to 31 digits and special characters. Digits 0123456789*# are allowed. Factory default shows hyphens (--) to indicate that a number has not been entered. This number is inserted in front of the inbound number to form the called number. The called number should not contain an access digit (8 or 9) and is not processed by the Analog Port Outbound Call Configuration screen. The called number is sent in the called party number information element of the ISDN setup message. The system can send and receive up to 31 digits.

Digits [+-] are special characters. Sequential numbers may be created using [min+max]. Random numbers may be created using [min-max]. Min and max represent the minimum and maximum values of the number. A left bracket, plus or minus, and right bracket are required. For example: 55512[00-99] indicates to send a random number between 5551200 and 5551299. The minimum and maximum

Help for Redirect Configuration—Master (continued)

values are limited to four digits each, may range from 0 through 9999, and do not require the same number of digits. The minimum and maximum values may not contain * or #. Brackets may be repeated, but may not be nested.

Inbound Number: Select INSERT ENTIRE NUMBER* (default), STRIP FIRST DIGIT, STRIP FIRST 2 DIGITS through STRIP FIRST 30 DIGITS, or EXCLUDE ENTIRE NUMBER. Selects the number of inbound digits to insert after the prefix number when dialing the redirect called number. Numbers are stripped starting from left to right. Example: *1*5551212, strip first 3 digits becomes 5551212.

GR-303 IDT sends the last four digits of the redirect number as the terminal number. Terminal numbers may range from 0001 to 2048. This identifies the RDT (remote digital terminal) to receive the call. No DTMF digits are sent.

Type of Number: Select UNKNOWN NUMBER* (default), INTERNATIONAL NUMBER, NATIONAL NUMBER, SUBSCRIBER NUMBER, or ABBREVIATED NUMBER. Used by ISDN signaling only. Type of number is sent in the called party number information element of the ISDN setup message. Unknown number is either not identified or contains prefix digits that are not part of the called number. Example: *70 to disable call waiting. International number includes a country code. Example: 443125551212. National number includes a national area code. Example: 3125551212. Subscriber number includes a local exchange number. Example: 5551212. Abbreviated number is 3 to 5 digits long. Example: 1212 or 51212.

Action: Select IMMEDIATE* (default), MESSAGE 1 PLAY ONCE, or MESSAGE 2 PLAY ONCE. If message is selected, the message is played once before the call is redirected.

Destination: Select OPPOSITE DIGITAL INTERFACE* (default), SAME CHANNEL OPPOSITE INT, SAME DIGITAL INTERFACE, or LAST 3 INBOUND DIGITS. Opposite digital interface indicates that the redirected call will use the digital interface not used by the inbound call. For example, if a call comes in on digital interface 1, the redirected call will go out on digital interface 2. The channel number is determined by the channel search configuration.

Same channel opposite interface indicates that the redirected call will use the same channel on the opposite digital interface.

Same digital interface indicates that the redirected call will use the same digital interface that is used by the inbound call. For example, if a call comes in on digital interface 1, the redirected call will go out on digital interface 1.

Last 3 inbound digits indicates that the last three digits of the inbound number determines the destination digital interface and channel number. The digital interface is indicated by the third to last digit. The channel number is indicated by the last two digits. For example, if the last three digits are 102, the redirected call goes out on digital interface 1 channel 2.

After each selection the screen displays ALL DETAILED SCREENS MATCH, SOME DETAILED SCREENS MATCH, or NO DETAILED SCREENS MATCH. This indicates whether the selection matches the same field on the detailed screens.

Redirect Configuration—Detailed

This screen configures individual redirect groups. There are four detailed screens, one for each redirect group.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator         Screen 10-2

                  REDIRECT 1 CONFIGURATION - DETAILED

Prefix Number:     --
Inbound Number:    INSERT ENTIRE NUMBER*
Type of Number:    UNKNOWN NUMBER*
Action:            IMMEDIATE*
Destination:       OPPOSITE DIGITAL INTERFACE*

Channels Assigned by Interface 1:  0
Channels Assigned by Interface 2:  0
Inbound Matches Assigned:          --


ACD Overflows Assigned:            --

Number of Redirected Calls:        0


* Indicates factory default.

   Enter fixed, sequential, or random number (0-31 digits). Backspace to edit.
    Press Arrow Keys, <F1> help, <F2> exit, <F3> previous, <F4> next
```

Help for Redirect Configuration—Detailed

Redirect causes an incoming call to initiate an outbound call on a digital interface, dial a specified number, and then connect both calls together. If desired, the incoming call can be connected to a voice message first, after which the outbound call is initiated.

Sidenote: ISDN calls redirected by the system's redirect facility copy the following information elements from incoming to outgoing call: bearer capability, display, calling party (caller), high layer compatibility, low layer compatibility, progress indicator, transit network selection, and user-user. The channel ID is not copied, but must be recalculated. The number of outgoing channels is determined from the bearer capability information transfer rate. The called number may be copied or replaced as configured by prefix number, inbound number, and type of number. The redirecting number information element is sent using the system's calling number. This is to assist law enforcement in identifying calls redirected through a third party.

Prefix Number: Enter up to 31 digits and special characters. Digits 0123456789*# are allowed. Factory default shows hyphens (--) to indicate that a number has not been entered. This number is inserted in front of the inbound number to form the called number. The called number should not contain an access digit (8 or 9) and is not processed by the Analog Port Outbound Call Configuration screen. The called number is sent in the called party number information element of the ISDN setup message. The system can send and receive up to 31 digits.

Digits [+-] are special characters. Sequential numbers may be created using [min+max]. Random numbers may be created using [min-max]. Min and max represent the minimum and maximum values of the number. A left bracket, plus or minus, and right bracket are required. For example: 55512[00-99] indicates to send a random number between 5551200 and 5551299. The minimum and maximum values are limited to four digits each, may range from 0 through 9999, and do not require the same number of digits. The minimum and maximum values may not contain * or #. Brackets may be repeated, but may not be nested.

Redirect Configuration—Detailed

Help for Redirect Configuration—Detailed (continued)

Inbound Number: Select INSERT ENTIRE NUMBER* (default), STRIP FIRST DIGIT, STRIP FIRST 2 DIGITS through STRIP FIRST 30 DIGITS, or EXCLUDE ENTIRE NUMBER. Selects the number of inbound digits to insert after the prefix number when dialing the redirect called number. Numbers are stripped starting from left to right. Example: *1*5551212, strip first 3 digits becomes 5551212.

GR-303 IDT sends the last four digits of the redirect number as the terminal number. Terminal numbers may range from 0001 to 2048. This identifies the RDT (remote digital terminal) to receive the call. No DTMF digits are sent.

Type of Number: Select UNKNOWN NUMBER* (default), INTERNATIONAL NUMBER, NATIONAL NUMBER, SUBSCRIBER NUMBER, or ABBREVIATED NUMBER. Used by ISDN signaling only. Type of number is sent in the called party number information element of the ISDN setup message. An unknown number is either not identified or contains prefix digits that are not part of the called number. Example: *70 to disable call waiting. An international number includes a country code. Example: 443125551212. A national number includes a national area code. Example: 3125551212. A subscriber number includes a local exchange number. Example: 5551212. An abbreviated number is 3 to 5 digits long. Example: 1212 or 51212.

Action: Select IMMEDIATE* (default), MESSAGE 1 PLAY ONCE, or MESSAGE 2 PLAY ONCE. If message is selected, the message is played once before the call is redirected.

Destination: Select OPPOSITE DIGITAL INTERFACE* (default), SAME CHANNEL OPPOSITE INT, SAME DIGITAL INTERFACE, or LAST 3 INBOUND DIGITS. Opposite digital interface indicates that the redirected call will use the digital interface not used by the inbound call. For example, if a call comes in on digital interface 1, the redirected call will go out on digital interface 2. The channel number is determined by the channel search configuration. Same channel opposite interface indicates that the redirected call will use the same channel on the opposite digital interface.

Same digital interface indicates that the redirected call will use the same digital interface that is used by the inbound call. For example, if a call comes in on digital interface 1, the redirected call will go out on digital interface 1. The channel number is determined by the channel search configuration.

Last 3 inbound digits indicates that the last three digits of the inbound number determines the destination digital interface and channel number. The digital interface is indicated by the third to last digit. The channel number is indicated by the last two digits. For example, if the last three digits are 102, the redirected call goes out on digital interface 1 channel 2.

Channels Assigned by Interface: This is a static display that shows the total number of channels assigned to this redirect group by each digital interface. When the digital interface is configured for ISDN, inbound matches are assigned on the Inbound Match Configuration screen. When the digital interface is configured for RBS, inbound matches are assigned on the Channel Configuration screen. When the digital interface is configured for RBS and inbound match routing is assigned to a channel, inbound matches must also be assigned on the Inbound Match Configuration screen.

Inbound Matches Assigned: This is a static display that shows MATCH followed by the reference number of all matches assigned to this redirect group. Hyphens (--) indicate no matches are assigned. Inbound matches are assigned on the Inbound Match Configuration screen.

ACD Overflows Assigned: This is a static display that shows ACD followed by the ACD number containing the overflow action assigned to this redirect group. Hyphens (--) indicate no ACD overflows are assigned. ACD overflows are assigned on the ACD Configuration screen.

Number of Redirected Calls: This is a real-time display that shows the total number of inbound calls currently being redirected.

Audio Monitor Configuration

This screen configures audio monitor operation.

```
Gordon Kapes, Inc.    System 930 Telephony Simulator          Screen 11

                      AUDIO MONITOR CONFIGURATION

Output:                 ON*
Mode:                   NEXT CALL - MONITOR UNTIL DISCONNECT*
Source:                 ANY DIGITAL INTERFACE OR ANALOG PORT*
DCN/Port:               ---

Audio Monitor Status:   ---









* Indicates factory default.

                      Press Space Bar or Backspace
                 Press Arrow Keys, <F1> help, <F2> exit
```

Help for Audio Monitor Configuration

Output: Select ON* (default) or OFF. Enables or disables the audio output.

Mode: Select NEXT CALL – MONITOR UNTIL DISCONNECT* (default), NEXT CALL, or FIXED CHANNEL OR PORT. Next call – monitor until disconnect: monitors the most recent call until the call disconnects. Next call: monitors the most recent call. Note that next call – monitor until disconnect and next call can monitor calls starting only from the initial call setup. Monitoring cannot be started once a call has been established. Fixed channel or port can monitor calls at any time during the call process.

Source: Select ANY DIGITAL INTERFACE OR ANALOG PORT* (default), DIGITAL INTERFACE 1, DIGITAL INTERFACE 2, ANALOG PORT, or DIGITAL INTERFACE 1 OR 2. Selects the primary source to be monitored.

DCN/Port: Select the DCN or PORT to be monitored when mode is set to fixed channel or port. DCN stands for digital channel number. PORT stands for analog port number. Shows hyphens (---) when mode is not fixed channel or port.

Audio Monitor Status: Shows the specific channel or port being monitored. Digital interfaces show DCN. Analog ports show PORT. Hyphens (---) indicate no channel is currently being monitored. Shows DCN-X/X OUT OF RANGE when the fixed digital channel number exceeds the range of possible digital channels.

Digital Interface Call Status

This screen shows each digital interface's call status. There are four status screens, two for each digital interface. Each screen has two modes: display called number or display calling number and name. Press the space bar to toggle between them.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 13-1

          DIGITAL INTERFACE CALL STATUS - DIGITAL INTERFACE 1

Interface Status:  SYNCHRONIZED

DCN    State          Dir  Connect   Called Number
1/1    00-NOT IN USE  ---  ---       ---
1/2    00-NOT IN USE  ---  ---       ---
1/3    00-NOT IN USE  ---  ---       ---
1/4    00-NOT IN USE  ---  ---       ---
1/5    00-NOT IN USE  ---  ---       ---
1/6    00-NOT IN USE  ---  ---       ---
1/7    00-NOT IN USE  ---  ---       ---
1/8    00-NOT IN USE  ---  ---       ---
1/9    00-NOT IN USE  ---  ---       ---
1/10   00-NOT IN USE  ---  ---       ---
1/11   00-NOT IN USE  ---  ---       ---
1/12   00-NOT IN USE  ---  ---       ---



      Press Space Bar to toggle between called and calling number
                 Press <F1> help, <F2> exit, <F4> next
```

Help for Digital Interface Call Status

Interface Status: Shows CARD NOT PRESENT, DISABLED, RECEIVING BLUE ALARM, RECEIVING YELLOW ALARM, SYNCHRONIZED, NOT SYNCHRONIZED, or LAYER 2 DOWN. Indicates framing status of digital interface.

DCN: Shows digital channel number as interface/number.

State: Shows state number followed by state name: ALERTING, MULTIRATE, CONNECT, DATA CHAN, DISCONNECT, FLASH, NOT IN USE, PROCEEDING, PROGRESS, RESERVED, SETUP, SETUP ACK, TRANSITION, or WINK.

Dir: Shows direction. IN for inbound or OUT for outbound. Hyphens (---) indicate channel not in use.

Connect: Shows ACD, ACT, DCN, GEN, MSG, or PORT. Shows resource connected to digital channel. ACD stands for automatic call distribution. ACT stands for connect action. DCN stands for digital channel number. GEN stands for call generator. MSG stands for recorder/announcer message. PORT stands for analog port number. Hyphens (---) indicate no connection.

Called Number: Shows the called number, up to 31 digits. Hyphens (---) indicate that a call is not present.

Calling Number: Shows the calling party number that originated the call, up to 17 digits. Hyphens (---) indicate that a call is not present or number not available. Shows PRIVATE if ISDN presentation is restricted. Press space bar to view this field.

Calling Name: Shows the calling party name that originated the call, if presentation is allowed, up to 17 digits. Hyphens (---) indicate that a call is not present or name not available. Shown for ISDN and GR-303 IDT digital interfaces only.

Transmission Status

This screen displays digital interface transmission information. There are two status screens, one for each digital interface.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 14-1

              TRANSMISSION STATUS - NETWORK INTERFACE 1

Interface Status:         NOT SYNCHRONIZED
Debounced Synchronized:   NO
Transmit Slips:           00038
Receive Slips:            00000
Bipolar Violations:       00000
ESF CRC Errors:           00000
Synchronization Losses:   00000
Receive Level:            LESS THAN -22dB
Elapsed Sync Time:        00 00:00 00
System Reboots:           0000000026
Reboot Error:             NONE



Reset Status Counters:    NO*

* Indicates factory default.

              Press Space Bar or Backspace then <ENTER>
                Press <F1> help, <F2> exit, <F4> next
```

Help for Transmission Status

Interface Status: Shows CARD NOT PRESENT, DISABLED, RECEIVING BLUE ALARM, RECEIVING YELLOW ALARM, SYNCHRONIZED, NOT SYNCHRONIZED, or LAYER 2 DOWN. Indicates framing status of digital interface. Layer 2 down is applicable to ISDN or GR-303 digital interfaces only.

Debounced Synchronized: Shows YES or NO. Yes indicates digital interface has maintained synchronization for one second or more. No indicates it has lost synchronization for five seconds or more.

Transmit Slips: Increments each time transmit elastic store buffer either repeats or deletes a frame. This number can increment only when debounced synchronized is yes.

Receive Slips: Increments each time receive elastic store buffer either repeats or deletes a frame. This number can increment only when debounced synchronized is yes.

Bipolar Violations: Increments each time the receiver circuitry fails to detect an alternate pulse or excessive zeros. This number can increment only when debounced synchronized is yes.

ESF CRC Errors: Shows number of ESF checksum errors. Hyphens (---) indicate D4 framing. This number is updated only when debounced synchronized is yes.

Synchronization Losses: Shows number of times system has lost synchronization.

Receive Level: Shows +2dB TO –7.5dB, –7.5dB TO –15dB, –15dB TO –22.5dB, or LESS THAN –22.5dB, or hyphens (---). Indicates strength of incoming signal; +2dB to –7.5dB is the strongest, less than –22.5dB is the weakest. Hyphens (---) indicate no card present.

Elapsed Sync Time: Shows time since start of last debounced synchronization. Format is days hours:minutes seconds.

Transmission Status

Help for Transmission Status (continued)

System Reboots: Shows number of times system has rebooted due to power failure, manual reset through keystrokes, or software run-time error.

Reboot Error: Shows NONE, BUS ERROR, ADDRESS ERROR, ILLEGAL INSTRUCTION, DIVIDE BY ZERO, PRIVILEGED INSTRUCTION, UNEXPECTED SINGLE STEP, UNEXPECTED TRAP, and GENERAL ERROR. Used by factory to show cause of unexpected system reboot.

Reset Status Counters: Select NO* (default), THIS SCREEN ONLY, ALL TRANSMISSION STATUS SCREENS, or ALL STATUS SCREENS. This selection allows status counters on this screen, all transmission screens, or all status screens to be reset. All status screens include Transmission Status, Call Counters, and Data Capture Display screens. <ENTER> must be pressed to reset the status counters. Status counters are saved when power is shut off.

Data Monitor—Digital Interface

This screen displays channel data for each digital interface. It also shows the interface status. There are two digital interface status screens, one for each digital interface. The screens may differ, depending on whether the interface is configured for ISDN, RBS, or GR-303.

```
Gordon Kapes, Inc.    System 930 Telephony Simulator       Screen 15-1

                 DATA MONITOR - DIGITAL INTERFACE 1

Interface Status:  SYNCHRONIZED

                              1 1 1 1 1 1 1 1 1 1 2 2 2 2 2
Channel:      1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4

Inbound Data (Hex):
Bits 5-8:     0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
Bits 1-4:     0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0











                Press <F1> help, <F2> exit, <F4> next
```

ISDN Data Monitor–Digital Interface Screen

```
Gordon Kapes, Inc.    System 930 Telephony Simulator       Screen 15-1

                 DATA MONITOR - DIGITAL INTERFACE 1

Interface Status:  SYNCHRONIZED

                              1 1 1 1 1 1 1 1 1 1 2 2 2 2 2
Channel:      1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4

Inbound Data (Hex):
Bits 5-8:     0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
Bits 1-4:     0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
Inbound Signaling Bits (Binary):
Bit A:        0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
Bit B:        0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
Bit C:        0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
Bit D:        0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
Outbound Signaling Bits (Binary):
Bit A:        0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
Bit B:        0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
Bit C:        0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
Bit D:        0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

                Press <F1> help, <F2> exit, <F4> next
```

RBS Data Monitor–Digital Interface Screen

Data Monitor—Digital Interface

```
Gordon Kapes, Inc.     System 930 Telephony Simulator       Screen 15-1

                  DATA MONITOR - DIGITAL INTERFACE 1

Interface Status:  SYNCHRONIZED

                               1 1 1 1 1 1 1 1 1 1 2 2 2 2 2
Channel:     1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4

Inbound Data (Hex):
Bits 5-8:    0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
Bits 1-4:    0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
Inbound Signaling Bits (Binary):
Bit A:       0 0 0 0 0 0 0 0 0 0 0 - 0 0 0 0 0 0 0 0 0 0 0 -
Bit B:       0 0 0 0 0 0 0 0 0 0 0 - 0 0 0 0 0 0 0 0 0 0 0 -
Bit C:       0 0 0 0 0 0 0 0 0 0 0 - 0 0 0 0 0 0 0 0 0 0 0 -
Bit D:       0 0 0 0 0 0 0 0 0 0 0 - 0 0 0 0 0 0 0 0 0 0 0 -
Outbound Signaling Bits (Binary):
Bit A:       1 1 1 1 1 1 1 1 1 1 1 - 1 1 1 1 1 1 1 1 1 1 1 -
Bit B:       1 1 1 1 1 1 1 1 1 1 1 - 1 1 1 1 1 1 1 1 1 1 1 -
Bit C:       1 1 1 1 1 1 1 1 1 1 1 - 1 1 1 1 1 1 1 1 1 1 1 -
Bit D:       1 1 1 1 1 1 1 1 1 1 1 - 1 1 1 1 1 1 1 1 1 1 1 -

                Press <F1> help, <F2> exit, <F4> next
```

GR-303 Data Monitor–Digital Interface Screen

Help for Data Monitor—Digital Interface

Interface Status: Shows CARD NOT PRESENT, DISABLED, RECEIVING BLUE ALARM, RECEIVING YELLOW ALARM, SYNCHRONIZED, NOT SYNCHRONIZED, or LAYER 2 DOWN. Indicates framing status of digital interface.

Channel: Shows channel number for items below.

Inbound Data: Shows data coming into the system. Most of the channels are bearer channels and contain digitally encoded voice or data. 00 indicates that the interface is not synchronized. FF is the idle state. Occasionally FF changes to FE on channels containing robbed bit signaling. Channel 24 of an ISDN digital interface is the data channel and contains HDLC packets. FF indicates HDLC packets are not present. 3F, 7E, 9F, CF, E7, F3, F9, and FC are valid HDLC idle states. The idle state tends to change as new HDLC packets are received.

Inbound Signaling Bits: Shows ABCD signaling in binary coming into the system. SF framing uses only the A and B bits. ESF framing also contains C and D bits. Shown for RBS and GR-303 Hybrid digital interfaces only.

Outbound Signaling Bits: Shows ABCD signaling bits in binary sent from the system. SF framing uses only the A and B bits. ESF framing also contains C and D bits. Shown for RBS and GR-303 Hybrid digital interfaces only.

Data Monitor—Analog Ports

This screen displays port data for the analog ports.

```
Gordon Kapes, Inc.    System 930 Telephony Simulator        Screen 15-3

                         DATA MONITOR - ANALOG PORTS


                                 1 1 1 1 1 1 1 1 1 1 2 2 2 2 2 2 2 2 2 2 3 3 3
Port:          1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2

Inbound Data (Hex):
Bits 5-8: F F F F F F F F F F F F F F F F F F F F F F F F F F F F F F F F
Bits 1-4: F F F F F F F F F F F F F F F F F F F F F F F F F F F F F F F F













                 Press <F1> help, <F2> exit, <F3> previous
```

Help for Data Monitor—Analog Ports

Port: Shows port number for items below.

Inbound Data: Shows data coming into the system. Ports 1 through 32 contain digitally encoded voice or data.

Tone Connect Test Functions

This screen connects a test tone to a digital interface channel or analog port. Note that DTMF tones don't have a cadence, but are steady tones.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator          Screen 16

                      TONE CONNECT TEST FUNCTIONS

Connect Status:  NORMAL OPERATION*
Tone Number:     1 (DTMF 1)*
Destination:     DIGITAL INTERFACE 1*
DCN/Port:        DCN 1*















* Indicates factory default.

                Press Space Bar or Backspace to select
                Press Arrow Keys, <F1> help, <F2> exit
```

Help for Tone Connect Test Functions

Connect Status: Select NORMAL OPERATION* (default) or CONNECT TONE. Normal operation indicates the tone is not connected. Connect tone indicates to connect the tone to the destination device. Upon power up or system reboot, connect status is normal operation.

Tone Number: Select 0 through 9 and 14 through 19. See tone list.

Destination: Select DIGITAL INTERFACE 1* (default), DIGITAL INTERFACE 2, or ANALOG PORTS. This selects the group of ports or channels the tone should be connected to.

DCN/Port: Select the specific DCN or PORT the tone should be connected to. DCN stands for digital channel number. PORT stands for analog port number. The factory default is DCN 1.

Tone List

0 - DTMF 0	4 - DTMF 4	8 - DTMF 8	16 - Dial Tone
1 - DTMF 1	5 - DTMF 5	9 - DTMF 9	17 - Audible Ring
2 - DTMF 2	6 - DTMF 6	14 - DTMF *	18 - Busy/Reorder
3 - DTMF 3	7 - DTMF 7	15 - DTMF #	19 - 440Hz

Digital Interface Test Functions

This screen allows configuration of digital interface test functions. There are four screens, two for each digital interface. The screens may differ, depending on whether the interface is configured for GR-303, ISDN, or RBS. Many of these tests are not related to each other.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 17-1

        DIGITAL INTERFACE TEST FUNCTIONS - DIGITAL INTERFACE 1

Loopback:                    NO LOOPBACK*
Send Alarm:                  AUTO ALARM*
Setup Bearer Capability:     SPEECH*
Information Transfer Rate:   64 KBPS*
Channel ID Format:           SEND CHANNEL ID WITHOUT INTERFACE ID*
Channel ID Mode:             SEND CHANNEL ID IN FIRST MESSAGE ONLY*
ISDN Setup Ack Message:      SEND SETUP ACK MSG*
ISDN Proceeding Message:     SEND PROCEEDING MSG*
ISDN Alerting Operation:     SEND ALERTING MSG*
ISDN Progress Description:   IN-BAND PROGRESS AVAILABLE*
In-band Tones:               SUPPLY IN-BAND TONES*
Cause Code Override:         DO NOT OVERRIDE CAUSE CODE*

ISDN Alerting Action:  SEND ALERTING MSG WITH PROGRESS IE
ISDN Progress Action:  SEND PROGRESS MSG WITH PROGRESS IE

* Indicates factory default.

               Press Space Bar or Backspace then <ENTER>
           Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

ISDN Digital Interface Test Functions Screen

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 17-1

        DIGITAL INTERFACE TEST FUNCTIONS - DIGITAL INTERFACE 1

Loopback:                 NO LOOPBACK*
Send Alarm:               AUTO ALARM*
In-band Tones:            SUPPLY IN-BAND TONES*
Local Ringback Tone:      CUT-THROUGH*
Allow FXO/SAO Signaling:  NOT ALLOWED ON USER SIDE*

* Indicates factory default.

               Press Space Bar or Backspace then <ENTER>
           Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

RBS Digital Interface Test Functions Screen

Digital Interface Test Functions

Help for Digital Interface Test Functions

Loopback: Select NO LOOPBACK* (default), LINE LOOPBACK, PAYLOAD LOOPBACK, or INTERNAL LOOPBACK. This selection affects layer 1. No loopback causes the digital interfaces to operate normally. No loopback is the default condition upon power up or reboot. Line loopback takes the received data and transmits it back to the source. Received data is retransmitted without change in framing format and any bipolar violations are not removed. Received data is also passed unchanged through to the system as incoming data. Payload loopback takes the received data and transmits it back to the source. The framing FDL bits and CRC bits are recalculated before being retransmitted and bipolar violations are removed. Normally this loopback is enabled in ESF framing mode. Received data is also passed unchanged through to the system as incoming data. Internal loopback takes the transmitted data and loops it back internally as the incoming data. Received data from the source is ignored. Transmit data is not sent to the source.

Send Alarm: Select AUTO ALARM* (default), YELLOW ALARM, BLUE ALARM, or NO ALARM. This selection affects layer 1. Auto alarm sends a yellow alarm as part of the transmitted data whenever the received data has been out of sync for more than five seconds. Upon power up or reboot, send alarm is set to auto alarm. When an interface is set to D4 framing, yellow alarm sends zeros in bit position 2 on all 24 digital interface channels. Does not cause loss of synchronization but causes transmitted data to be intentionally corrupted. When an interface is set to ESF framing, yellow alarm sends 8 zeros and 8 ones in the facilities data link time slot of the framing bit. Does not cause loss of synchronization or corruption of transmitted data. Blue alarm sends ones in all bit positions, including the framing bit. Causes loss of synchronization and essentially no transmit data. No alarm prevents transmitting a yellow or blue alarm, even when the incoming data warrants it.

Setup Bearer Capability: Select SPEECH* (default), UNRESTRICTED DIGITAL INFORMATION, RESTRICTED DIGITAL INFORMATION, 3.1 KHZ AUDIO (ANALOG MODEM), 7 KHZ AUDIO, or VIDEO. This information is sent in the bearer capability information element. This selection is not used by redirected calls. GR-303 does not use this setting and always sends unrestricted digital information. Shown for ISDN digital interfaces only.

Information Transfer Rate: Select 64 KBPS* (default) through 23 x 64 KBPS (MULTIRATE) in 64 KBPS increments. This information is sent in the bearer capability information element and channel ID information element. This selection is not used by redirected calls. 5ESS, DMS100, Euro ISDN, and NI-2 accept all transfer rates. 4ESS accepts 64 KBPS and 6 x 64 KBPS (H0) only. All other 4ESS transfer rates are special cases and are not coded according to CCITT standards. Shown for GR-303 and ISDN digital interfaces only.

Channel ID Format: Select SEND CHANNEL ID WITHOUT INTERFACE ID* (default) or SEND CHANNEL ID WITH INTERFACE ID. Send channel ID without interface ID indicates that the channel ID information element is sent in ISDN messages with interface ID not present. This is the normal case for FAS (Facility Associated Signaling) and indicates that each digital interface contains its own ISDN signaling channel. Send channel ID with interface ID indicates that the channel ID information element is sent in ISDN messages with interface ID present. This is the normal case for NFAS (Non-Facility Associated Signaling) and indicates that only one digital interface contains an ISDN signaling channel. This system must be licensed for NFAS to operate properly on more than one interface. GR-303 does not use this setting and always sends the channel ID with interface ID. Shown for ISDN digital interfaces only.

Channel ID Mode: Select SEND CHANNEL ID IN FIRST MESSAGE ONLY* (default), SEND CHANNEL ID IN ALLOWABLE MESSAGES, or DO NOT SEND CHANNEL ID. Send channel ID in first message only indicates that the channel ID information element is sent in the first ISDN message of a new call. This is usually the setup, call proceeding, or setup acknowledge message. Send channel ID in allowable messages indicates that the channel ID information element is sent in all allowable ISDN messages. This includes the setup, setup acknowledge, proceeding, progress, and connect messages. Do not send channel ID indicates that the channel ID information element is not sent in ISDN messages. This is allowed for user side Euro ISDN and causes problems for other switch emulations. GR-303 does not use this setting and sends the channel ID when appropriate. Shown for ISDN digital interfaces only.

Digital Interface Test Functions

Help for Digital Interface Test Functions (continued)

ISDN Setup Ack Message: Select SEND SETUP ACK MSG* (default), or DO NOT SEND SETUP ACK MSG. Send setup ack msg causes the system to send the ISDN setup acknowledge message in response to receiving an ISDN setup message. Do not send setup ack msg prevents the system from sending the ISDN setup acknowledge message. This message is normally sent to acknowledge a setup message not containing a called number. This is allowed by GR-303. Shown for GR-303 and ISDN digital interfaces only.

ISDN Proceeding Message: Select SEND PROCEEDING MSG* (default), or DO NOT SEND PROCEEDING MSG. Send proceeding msg causes the system to send the ISDN call proceeding message in response to receiving an ISDN setup message. Do not send proceeding msg prevents the system from sending the ISDN call proceeding message. This message is normally sent to acknowledge a setup message containing a called number. Shown for GR-303 and ISDN digital interfaces only.

The next four test functions provide a way to test how the receiving side handles progress tones. In-band (circuit switched) progress tones are sent across the digital interface faster than ISDN progress messages (packet switched), but automated equipment might prefer using ISDN progress messages. Network side dial tone (after any access digits) is always supplied locally. User side T1 ISDN uses en-bloc sending, so dial tone (after any access digits) is supplied locally. User side T1 RBS (robbed bit signaling) uses overlap sending, so dial tone (after any access digits) must be supplied by the network side. The ISDN call proceeding message is always sent after the call setup message.

ISDN Alerting Operation: Select SEND ALERTING MSG* (default), SEND PROGRESS MSG, SEND CONNECT MSG, or DO NOT SEND ALERTING MSG. Send alerting msg causes the system to send the ISDN alerting message and supply in-band audible ring progress tones. Send progress msg prevents the ISDN alerting message from being sent. Instead, it sends the ISDN progress message while supplying in-band audible ring progress tones providing ISDN progress description is not do not send progress IE. Send connect msg prevents the ISDN alerting message from being sent. Instead, it sends the ISDN connect message while supplying in-band audible ring progress tones providing ISDN progress description is not do not send progress IE. Do not send alerting msg prevents any ISDN messages from being sent when an alerting message would normally be sent. This creates an abnormal situation. Combined with do not send proceeding msg, the near end should resend the setup message and then release the call. To help understand the full effect from this selection, the message sent is shown at the bottom of this screen. GR-303 does not use this setting and sends alerting messages and audible progress tones when appropriate. Shown for ISDN digital interfaces only.

ISDN Progress Description: Select IN-BAND PROGRESS AVAILABLE* (default), NOT END-TO-END ISDN, RETURN TO END-TO-END ISDN, or DO NOT SEND PROGRESS IE. This is a code sent in the progress indicator information element of progress messages and alerting messages. It also determines which ISDN messages are sent. To help understand the full effect from this selection, the message sent is shown at the bottom of this screen. Use in-band progress available, not end-to-end ISDN, and return to end-to-end ISDN to indicate there are in-band progress tones. On the receiving side, presence of a progress IE indicates to connect the bearer channel to the audio path. Use do not send progress IE to indicate there are no in-band progress tones. Do not send progress IE implies an end-to-end ISDN connection and there are no in-band progress tones. On the receiving side, lack of a progress IE indicates not to connect the bearer channel to the audio path. GR-303 does use this setting because it does not send progress messages. Shown for ISDN digital interfaces only.

In-band Tones: Select SUPPLY IN-BAND TONES* (default) or DO NOT SUPPLY IN-BAND TONES. Supply in-band tones permits the system to send tones across the digital interface. This includes dial, busy, reorder, and audible ring progress tones. These tones are normally sent by the system when it is the terminating side to a call. Do not supply in-band tones prevents the system from supplying in-band tones across the digital interface. This is not the normal case for the public network. Use this selection to determine the origin of progress tones. This selection affects the audio path only, it has no effect on ISDN messages sent by the system.

Digital Interface Test Functions

Help for Digital Interface Test Functions (continued)

Local Ringback Tone: Select CUT-THROUGH* (default) or SUPPLY LOCAL RINGBACK. Cut-through connects the audio path so that the caller can hear tones from the connected system. Supply local ringback is intended for use with channel banks that do not supply ringback tone. Local ringback is supplied on the network side (FXO/SAO) of loop start or ground start systems, and both sides for E&M immediate and E&M wink. Shown for RBS digital interfaces only.

Allow FXO/SAO Signaling: Select NOT ALLOWED ON USER SIDE* (default) or ALLOWED ON USER SIDE. Not allowed on user side does not allow FXO/SAO signaling on the user side and does not allow FXS/SAS on the network side. Allow on user side permits FXO/SAO signaling on the user side and FXS/SAS on the network side. Shown for RBS digital interfaces only.

Cause Code Override: Select DO NOT OVERRIDE CAUSE CODE* (default), CAUSE 0 through CAUSE 127, USE LAST 3 DIGITS OF CALLED NUMBER, or INCREMENT AFTER EACH CALL. This is the code sent in the cause information element of ISDN progress and disconnect messages. Other ISDN messages are not affected and send the cause code normally supplied by the system. This affects cause codes sent by the system, not cause codes received by the system. Do not override cause code indicates to ignore this selection and send the cause code normally supplied by the system. Cause 0 through cause 127 indicates to send the indicated cause code instead of the cause code normally supplied by the system. Use last 3 digits of called number indicates to send the last three digits of the called number as the cause code. If the last three digits exceed 127, send the cause code normally supplied by the system. Increment after each call indicates to increment the cause code after each call. Wraps around from 127 to 0. Shown for GR-303 and ISDN digital interfaces only.

The next two fields are static displays to help understand the full effects from ISDN Alerting Operation and ISDN Progress Description. ISDN Alerting Action: Shows SEND ALERTING MSG, SEND ALERTING MSG WITH PROGRESS IE, NONE, SEND PROGRESS MSG WITH PROGRESS IE, or SEND CONNECT MSG WITH PROGRESS IE. Send alerting message indicates that the ISDN alerting message is sent without the progress indicator information element and the system supplies in-band audible ring progress tones. This indicates that the receiving side should supply local audible ring progress tone even though the remote audible ring progress tone is present. Send alerting message with progress information element indicates that the ISDN alerting message is sent with the progress indicator information element and the system supplies in-band audible ring progress tones. None indicates that no ISDN message is sent and the system supplies in-band audible ring progress tones. Send progress message with progress information element indicates that the ISDN progress message is sent with the progress indicator information element and the system supplies in-band audible ring progress tones. Send connect message with progress information element indicates that the ISDN connect message is sent with the progress indicator information element and the system supplies in-band audible ring progress tones. GR-303 does not show this field and sends alerting messages when appropriate. Shown for ISDN digital interfaces only.

ISDN Progress Action: Shows SEND DISCONNECT MSG, SEND PROGRESS MSG WITH PROGRESS IE or NONE. Send disconnect message indicates that the ISDN disconnect message is sent without supplying an in-band progress tone, indicating that the far end should supply local busy or reorder progress tones based on the cause information element included in the disconnect message. Send progress message with progress information element indicates that the ISDN progress message is sent with the progress indicator information element and the system supplies an in-band busy or reorder progress tone. None indicates that the switch emulation does not send a progress message. GR-303 does not show this field and does not send progress messages. Shown for ISDN digital interfaces only.

Digital Interface Test Functions

This screen allows configuration of digital interface test functions. There are four screens, two for each digital interface. The screens may differ, depending on whether the interface is configured for GR-303, ISDN, or RBS. This screen has no configurable parameters for RBS. Most of these tests are not related to each other.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 17-2

        DIGITAL INTERFACE TEST FUNCTIONS - DIGITAL INTERFACE 1

ISDN Layer 2 Startup:          START USER SIDE IMMEDIATELY*
Setup Exclusive Channel ID:    PREFERRED*
Non-Setup Channel ID Format:   NUMBER/SLOT MAP*
Multirate Slot Assignment:     FIXED SLOTS*
Redirect Elements:             COPY END TO END ELEMENTS*
Network Specific Service:      DO NOT SEND NETWORK SPECIFIC SERVICE*













* Indicates factory default.

               Press Space Bar or Backspace then <ENTER>
   Press Arrow Keys, <F1> help, <F2> exit, <F3> previous, <F4> next
```

Help for Digital Interface Test Functions

ISDN Layer 2 Startup: Select START USER SIDE IMMEDIATELY* (default), START NETWORK SIDE IMMEDIATELY, START EITHER SIDE IMMEDIATELY, or WAIT UNTIL FIRST OUTGOING CALL. This selection selects when the interface should initiate layer 2 handshaking. Start user side immediately indicates to initiate handshaking if this interface is the user side. Start network side immediately indicates to initiate handshaking if this interface is the network side. Start either side immediately indicates to initiate handshaking whether this interface is the network side or user side. Wait until first outgoing call indicates to initiate handshaking upon making the first outgoing call. Shown for GR-303 and ISDN digital interfaces only.

Setup Exclusive Channel ID: Select PREFERRED* (default) or EXCLUSIVE. This information is sent by the ISDN channel ID information element as part of setup messages. Preferred indicates that the channel is being requested and may be changed by the near end if the channel is already in use. Exclusive indicates that the channel is not negotiable and the call should be rejected by the near end if the channel is already in use. This feature was added to test for rejection of preferred channel IDs and to allow testing for glare. Applies only to channel IDs sent as part of setup messages. Non-setup messages always send exclusive channel IDs. DMS100 always sends exclusive channel IDs regardless of this setting. GR-303 always sends exclusive channel IDs regardless of this setting. Shown for ISDN digital interfaces only.

Non-Setup Channel ID Format: Select NUMBER/SLOT MAP* (default) or SLOT MAP ONLY. This is the format sent by the ISDN channel ID information element in non-setup messages. Number/slot map indicates that number format is used for 64 KBPS calls and slot map format is used for 128 KBPS or higher calls. Slot map only indicates that slot map format is used for all calls. Use number/slot map on older Lucent (AT&T) switches that do not support slot map format for 64 KBPS calls. Slot map only will cause older Lucent

Digital Interface Test Functions

Help for Digital Interface Test Functions (continued)

(AT&T) switches to return cause code 82, channel does not exist, when sent slot map format in call proceeding messages. GR-303 always sends numbered channel IDs regardless of this setting. Shown for ISDN digital interfaces only.

Multirate Slot Assignment: Select FIXED SLOTS* (default) or FLEXIBLE SLOTS. This selection applies to H0, H11, and multirate channels. Fixed slots indicate that slot assignment must be contiguous and end on a slot evenly divisible by the rate multiplier. Example: 6 x 64 KBPS transfer rates must end on slot 6, 12, 18, 24, or 30. Flexible slots indicate that slot assignment may be contiguous or noncontiguous. Shown for GR-303 and ISDN digital interfaces only.

Redirect Elements: Select COPY END TO END ELEMENTS* (default) or DO NOT COPY END TO END ELEMENTS. This selection indicates when the system's redirect facility will copy the display, high layer compatibility, low layer compatibility, progress indicator, transit network, and user-user information elements from the incoming setup message to outgoing setup message. Copy end to end elements allows these elements to be copied. Do not copy end to end elements inhibits these elements from being copied. Shown for ISDN digital interfaces only.

Network Specific Service: Select DO NOT SEND NETWORK SPECIFIC SERVICE* (default), or SEND CODE 0 through SEND CODE 31. Network specific service identifies a specific long distance service. It is sent in the network specific facility information element of ISDN setup messages. Do not send network specific service does not send the network specific facility information element when originating a call and does not check for it when receiving a call. Send code 0 through send code 31 sends the network specific facility information element when originating a call and checks for it when receiving a call. If missing, the call is rejected with cause 96; mandatory information element is missing. Shown for ISDN digital interfaces only.

Data Capture Configuration

This screen configures the way signaling messages and call detail records are captured and displayed. It also starts or stops the call generator. To display captured data, press <ENTER> from this screen.

```
Gordon Kapes, Inc.        System 930 Telephony Simulator       Screen 18

                          DATA CAPTURE CONFIGURATION

 Status:                   NOT RUNNING*
 Screen Mode:              VT100 EMULATION*
 Capture Type:             LAYER 3 WITH INFORMATION ELEMENTS*
 Calls to Capture:         ALL*
 Source:                   INTERFACE 1 AND INTERFACE 2*
 Message Type 1:           ALL MESSAGE TYPES*
 Message Type 2:           NONE*
 Message Type 3:           NONE*
 Message Type 4:           NONE*
 Call Generator Mode:      NOT RUNNING*

 Call Generator Status:    NOT RUNNING




 * Indicates factory default.

       Press Space Bar or Backspace. Press <ENTER> to view results.
                          Press <F1> help, <F2> exit
```

Help for Data Capture Configuration

Status: Select NOT RUNNING or RUNNING. Not running stops data capture and holds the information shown in VT100 emulation mode. Running allows new information to be captured. In VT100 emulation mode, press <ENTER> to display the last page of captured information. In raw ASCII output screen mode, press <ENTER> to display the special raw ASCII output screen.

Screen Mode: Select VT100 EMULATION* (default) or RAW ASCII OUTPUT. VT100 emulation displays captured information using normal screen output. Raw ASCII output sends newly arrived information to the display device with VT100 emulation disabled. This feature is useful for capturing screen output to a file.

Capture Type: Select LAYER 3 WITH INFORMATION ELEMENTS* (default), LAYER 3 – NO INFORMATION ELEMENTS, LAYER 2 & LAYER 3 WITH INFORMATION ELEMENTS LAYER 2 & LAYER 3 – NO INFORMATION ELEMENTS, LAYER 2 ONLY, or CALL DETAIL RECORDS. Layer 3 with information elements shows ISDN layer 3 messages with information elements on separate lines. Layer 3 – no information elements: shows ISDN layer 3 messages without information elements. Layer 2 only: shows ISDN layer 2 frames. Use layer 2 to check layer 2 handshaking with the near end. Layer 2 and layer 3 shows identical information for T1 RBS. Call detail records show basic information such as channel used and called number. They are displayed after a call is disconnected. See help in the Data Capture Display Screen section for more details.

Calls to Capture: Select ALL* (default), NEXT INCOMING, NEXT OUTGOING, or UNTIL INTERNAL ERROR. Indicates whether to capture all calls or only the next incoming or outgoing call. ALL captures all calls. It is undesirable to run data capture with all calls for long periods of time because it may interfere with normal system operation. This selection is preset to ALL when Capture Type is CALL DETAIL RECORDS. NEXT INCOMING captures the next incoming call. Data capture may be retriggered by changing data capture status from running to not running and then running again. NEXT OUTGOING captures the next outgoing

Data Capture Configuration

Help for Data Capture Configuration (continued)

call. Data capture may be retriggered by changing data capture status from running to not running and then running again. UNTIL INTERNAL ERROR captures all calls until an internal ISDN error is reported. Upon detecting an internal ISDN error, data capture status is changed to not running so that data capture is preserved for later analysis.

Source: Select INTERFACE 1 AND INTERFACE 2* (default), INTERFACE 1, INTERFACE 2, or ANALOG PORTS. Indicates the source from which the data will be captured. Analog ports work only with call detail records.

Message Type 1: Select ALL MESSAGE TYPES* (default), SETUP, SETUP_ACK, CALL_PROC, ALERTING, PROGRESS, CONNECT, CONNECT_ACK, DISCONNECT, RELEASE, RELEASE_COMP, INFO, NOTIFY, RESTART, RESTART_ACK, STATUS, STATUS_ENQ, or USER_INFO. Indicates whether to capture all message types or only the selected message type.

Message Type 2: Select NONE* (default), SETUP, SETUP_ACK, CALL_PROC, ALERTING, PROGRESS, CONNECT, CONNECT_ACK, DISCONNECT, RELEASE, RELEASE_COMP, INFO, NOTIFY, RESTART, RESTART_ACK, STATUS, STATUS_ENQ, or USER_INFO. Allows a second message type to be captured.

Message Type 3: Similar to message type 2. Allows a third message type to be captured.

Message Type 4: Similar to message type 2. Allows a fourth message type to be captured.

Call Generator Mode: Select NOT RUNNING* (default), RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING – UNTIL NO CONNECT. This is a global selection that affects all digital interfaces and analog ports. Upon power up or reboot, the call generator is not running. Not running immediately disconnects all calls started by the call generator and prevents the call generator from starting new calls. Running – start new calls allows the call generator to start new calls. By design the call generator waits the selected time between cycles before making the first call. Running – no new calls prevents new calls from starting but does not disconnect calls in process. Running – until no connect prevents new calls from starting after a call has failed to connect but does not disconnect calls in process. <ENTER> must be pressed to change this feature.

Call Generator Status: Shows NOT RUNNING, NOT RUNNING – MAXIMUM NUMBER OF CALLS REACHED, RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING – UNTIL NO CONNECT. Indicates current call generator status.

Data Capture Display

The display screen shows layer 2, layer 3, and call detail records. It is accessed from the configuration screen by pressing <ENTER>.

```
Gordon Kapes, Inc.    System 930 Telephony Simulator       Screen 18-10

                      DATA CAPTURE DISPLAY PAGE 10


 SEND      DSL=1 TIME=02:43 15.850   CRV=00002 CRF=0  SETUP
           BEARER_CAPABILITY         04 03 80 90 A2   SPEECH
           CHANNEL_ID                18 03 A1 83 81   CHANNEL 1
           PROGRESS_IND              1E 02 80 83  ORIGIN ADDRESS NON-ISDN
           CALLED_NUMBER             70 08 C1 "5551212"
 RECEIVE   DSL=1 TIME=02:43 15.900   CRV=00002 CRF=1  CALL_PROC
           CHANNEL_ID                18 03 A9 83 81   CHANNEL 1
 RECEIVE   DSL=1 TIME=02:43 15.900   CRV=00002 CRF=1  ALERTING
           PROGRESS_IND              1E 02 80 88  IN-BAND PROGRESS
 RECEIVE   DSL=1 TIME=02:43 22.050   CRV=00002 CRF=1  CONNECT
 SEND      DSL=1 TIME=02:43 22.050   CRV=00002 CRF=0  CONNECT_ACK
 RECEIVE   DSL=1 TIME=02:43 31.550   CRV=00002 CRF=1  DISCONNECT
           CAUSE                     08 02 80 90  #16:NORMAL CLEARING
 SEND      DSL=1 TIME=02:43 31.550   CRV=00002 CRF=0  RELEASE
 RECEIVE   DSL=1 TIME=02:43 31.600   CRV=00002 CRF=1  RELEASE_COMP


      RUNNING - Press Space Bar to stop. Press C to clear screen.
               Press <F1> help, <F2> exit, <F3> previous
```

ISDN Data Capture Display Screen

```
Gordon Kapes, Inc.    System 930 Telephony Simulator       Screen 18-10

                      DATA CAPTURE DISPLAY PAGE 10









 I/O,CID..,DCN.,CONN,STATE NAME,OUT.,IN..,TIME......,CALLED, (RBS Format)
 OUT,00001,D101,P001,SETUP     ,1111,0000,03:03 12.550,,
 OUT,00001,D101,P001,WINK      ,1111,1111,03:03 12.900,,
 OUT,00001,D101,P001,SETUP ACK ,1111,0000,03:03 13.100,,
 OUT,00001,D101,P001,CONNECT   ,1111,1111,03:03 17.650,5551212,
 OUT,00001,D101,P001,CONNECT   ,1111,0000,03:03 26.500,5551212,
 OUT,00000,D101,    ,NOT IN USE,0000,0000,03:03 27.800,,


      RUNNING - Press Space Bar to stop. Press C to clear screen.
               Press <F1> help, <F2> exit, <F3> previous
```

RBS Data Capture Display Screen

Data Capture Display

```
Gordon Kapes, Inc.    System 930 Telephony Simulator        Screen 18-10

                           DATA CAPTURE DISPLAY PAGE 10

*I/O,CID,DCN,CONN,START_T,CONNECT_T,DISC_T,CALLED,CALLING,NAME, (CDR Format)
*O,00002,D101,P001,01030426,01030430,01030437,5551212,,,

       RUNNING - Press Space Bar to stop. Press C to clear screen.
                  Press <F1> help, <F2> exit, <F3> previous
```

Call Detail Record Data Capture Display Screen

Help for Data Capture Display

In VT100 emulation mode, information is scrolled upward, starting from the bottom of the last page. It is best to start viewing from the last page as it contains the most recent information. In raw ASCII output mode, each line is sent as 8 bit ASCII text followed by carriage return (CR) and line feed (LF).

Select NOT RUNNING and RUNNING by pressing the space bar. Not running stops data capture and holds the information shown. Running allows new information to be captured. If there is a lot of screen activity, the message ***DATA OVERFLOW IN DATA CAPTURE*** will appear in the text. Pressing the space bar also retriggers data capture when configured for next incoming call or next outgoing call. Press C to clear screen. Pressing C also retriggers data capture when configured for next incoming call or next outgoing call. Pressing C has no effect on data capture status.

The following text applies to ISDN layer 2 and layer 3. Robbed bit signaling and call detail records are explained later.

The first word of each line identifies the information source: RECEIVE, SEND, STATUS, and ERROR. Receive indicates the message was received by the system when ISDN signaling is in effect, or this is the terminating side when RBS signaling is in effect. Send indicates the message was sent by the system when ISDN signaling is in effect, or this is the originating side when RBS signaling is in effect. Status indicates the message is internal to the system and typically indicates digital interface shutdown and startup requests due to loss of synchronization. Error indicates an error has occurred within the system. Error messages are captured even when the status of data capture is not running.

ISDN records are displayed in several different formats: layer 2, layer 3, and layer 3 with information elements.

When capture type is layer 3, the format for the first line of ISDN messages that are sent and received is: SEND or RECEIVE, DSL, TIME, CRV, CRF, and ISDN message name. Information elements then follow on separate lines.

When capture type is layer 2 & layer 3, the first line shows ISDN layer 2 frames. Use layer 2 to check layer 2 handshaking with the near end. The format is: SEND or RECEIVE, DSL, TIME, TEI, SAPI, frame type, NS,

Data Capture Display

Help for Data Capture Display (continued)

NR, and PF. SEND or RECEIVE, CRV, CRF, and ISDN message name appear on next line. Information elements then follow on separate lines.

Format of status messages is: STATUS, DSL, TIME, TEI, SAPI, and primitive type.

Format of error messages is: ERROR followed by text describing the error.

Definitions:
CES: Connection Endpoint Suffix. Layer 3 information. Associated with TEI. Private to each side of the digital interface. CES 0 is broadcast, 1 is PRI endpoint.

CID: Call ID. Internal to the system. Decimal number used to identify each call. Private to each side of the system. Associated with Call Reference, which identifies the call between the user and the connected equipment. CID 0 is non-call related, 0001 through 32767 are incoming calls, 32769 through 65535 are outgoing calls. Robbed Bit Signaling CIDs range from 00001 through 32767. GR-303 CIDs range from 00001 through 02048 and are the same as the CRV.

CRF: Call Reference Flag. Layer 3 information. The most significant bit of the call reference. This bit is 0 for the originating side and 1 for the terminating side. Used by ISDN only. Not used by GR-303.

CRV: Call Reference Value. Layer 3 information. Decimal number used to identify a specific call between the system and the connected equipment. Numbers range from 00001 through 32767. CRV is associated with Call ID, which is private to each side of the connection. GR-303 CRVs range from 0001 through 2048 and identify the terminal ID.

DSL: Digital Subscriber Line. Layer 1 information.

SAPI: Service Access Point Identifier. Layer 2 information. SAPI 0 is signaling, 16 is packetized data, and 63 is network management.

TEI: Terminal Endpoint Identifier. Layer 2 information. Identifies the terminating endpoint to the central office. Associated with CES. TEI 0 is broadcast, 1 is PRI endpoint.

TIME: Time shown as hh:mm ss.mmm representing hour, minute, second, and millisecond. Uses current system time.

Frame Type: Layer 2 information:
DISC: Disconnect.
DM: Disconnect mode.
FRMR: Frame Reject.
I: Information; followed by NS and NR.
REJ: Reject.
RNR: Receive Not Ready.
RR: Receive Ready; followed by NR.
SABME: Set Asynchronous Balanced Mode Extended.
UA: Unnumbered Acknowledge.
UI: Unnumbered Information.
XID: Exchange Identification.

NR: Frame Received Number. Layer 2 information.

NS: Frame Sent Number. Layer 2 information.

PF: Poll/Final Bit. Layer 2 information. PF 0 is poll, 1 is final.

Primitives are ISDN layer to layer messages. Primitives are displayed by status messages and are generally useful only to the factory. Although there are too many to fully describe here, the most common are:
MDL_ERR_IND: Layer 2 error indication.
MDL_STARTUP_REQ: Layer 2 startup request.
MDL_SHUTDOWN_REQ: Layer 2 shutdown request.

Data Capture Display

Help for Data Capture Display (continued)

Message Name: Layer 3 information. Although there are too many to fully describe here, the most common are:
SETUP: Begin call setup.
CALL_PROC: Proceeding; near end has received setup.
PROGRESS: Far end is sending a progress tone.
ALERTING: Far end is sending an audible ring progress tone.
CONNECT: Far end has answered.
DISCONNECT: Disconnect far end.
RELEASE: Near end has disconnected.

Information elements contain details about each message. Layer 3 information. Information elements are displayed on separate lines after the message name. There are two information element formats: standard Q.931 and nonstandard. Both formats display the information element name followed by hex octets representing the information element data. Q.931 octet 1 contains the information elements identifier. Q.931 octet 2 shows the number of octets to follow. Some Q.931 information elements only contain one octet. Nonstandard information elements are preceded by a tilde (~) and are for factory use. Nonstandard octets 1 and 2 are the information element identifier. Nonstandard octet 3 shows the number of octets to follow. IA5 (International Alphabet) characters are decoded and shown as readable characters surrounded by quotes (“).

The most common information elements are:
BEARER_CAPABILITY: Encoding method. Transfer capability shown as text.
CALLING_NUMBER: Originating phone number.
CALLED_NUMBER: Destination phone number.
CAUSE: Cause of problem. Cause shown as text.
CHANGE_STATUS: Channel is in or out of service.
CHANNEL_ID: Bearer channel. Channel number shown as text.
DISPLAY: Send text such as an extension number or name.
NETWORK_FACILITY: Type of long distance service.
PROGRESS_IND: Whether or not call is end to end ISDN.
REDIRECTING_NUM: Redirecting phone number.

Call detail records show basic call information such as channel used and called number. Call detail records are displayed after a call is disconnected. The header *I/O,CID,DCN,CONN,START_T,CONNECT_T,DISC_T,CALLED,CALLING,NAME, (CDR Format) is displayed when data capture starts running. Call detail records start with * and commas separate each field.
I/O shows whether the call was inbound or outbound.
CID shows call id (5 decimal digits).
DCN shows the digital channel number (Dnnn) or analog port (Pnnn).
CONN shows the connecting device and is formatted as Annn (connect action number), Dnnn (digital channel number), Gnnn (call generator), Mnnn (message number), or Pnnn (analog port). Blank indicates no device.
Time is shown as ddhhmmss representing day, hour, minute, and second.
START_T shows the time call was started.
CONNECT_T shows the time call was connected.
DISC_T shows the time call was disconnected.
CALLED shows called number.
CALLING shows calling party number. NAME shows calling party name.

Adjacent commas (,,) indicate information was not available.

Example: *O,00003,D201,P016,01143915,,,5551212,,, shows that the call was sent by the system, call id is 00003, uses digital interface 2 channel 1, originated from analog port 16, started at day 01 hour 14 minute 39 second 15, no connect or disconnect time, called number is 5551212, and no calling number.

Quick System Status

This screen shows a simple real time overview of each analog port or digital interface channel. It also shows the audio monitor status and call generator status. It starts or stops the call generator.

```
Gordon Kapes, Inc.      System 930 Telephony Simulator       Screen 19

                          QUICK SYSTEM STATUS

                                1 1 1 1 1 1 1 1 1 1 2 2 2 2 2 2 2 2 2 2 3 3 3
              1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2

Analog 1-32:  - - - - - - - -

Interface 1:  - - - - - - - - - - - - - - - - - - - - - - -

Interface 2:  - - - - - - - - - - - - - - - - - - - - - - -

Call Generator Mode:    NOT RUNNING*

Call Generator Status:  NOT RUNNING

Audio Monitor Status:   ---


               Press Space Bar or Backspace then <ENTER>
                       Press <F1> help, <F2> exit
```

Help for Quick System Status

Analog 1-32: Shows single character status for each analog port: hyphen (-) for on-hook or break-loop current, A for alerting, C for off-hook, or N for no ring current detected (ring fault). If analog card is not present, nothing is shown for its associated ports.

Interface 1 and Interface 2: Shows single character value for each digital interface channel: hyphen (-) for not in use, A for alerting, C for connected, D for disconnect, F for flash, G for progress, K for setup acknowledge, M for multirate, P for proceeding, R for reserved, S for setup, T for transition, and W for wink.

The line may also show CARD NOT PRESENT, DISABLED, RECEIVING BLUE ALARM, RECEIVING YELLOW ALARM, NOT SYNCHRONIZED, or LAYER 2 DOWN indicating framing status of digital interface.

Call Generator Mode: Select NOT RUNNING* (default), RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING – UNTIL NO CONNECT. This is a global selection that affects all digital interfaces and analog ports. Upon power up or reboot, the call generator is not running. Not running immediately disconnects all calls started by the call generator and prevents the call generator from starting new calls. Running – start new calls allows the call generator to start new calls. By design the call generator waits the selected time between cycles before making the first call. Running – no new calls prevents new calls from starting but does not disconnect calls in process. Running – until no connect prevents new calls from starting after a call has failed to connect but does not disconnect calls in process. <ENTER> must be pressed to change this feature.

Call Generator Status: Shows NOT RUNNING, NOT RUNNING – MAXIMUM NUMBER OF CALLS REACHED, RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING – UNTIL NO CONNECT. Indicates current call generator status.

Audio Monitor Status: Shows the specific analog port or digital channel being monitored. Analog ports show PORT. Digital interfaces show DCN. Hyphens (---) indicate no channel is currently being monitored.

Save/Restore System Configuration

This screen loads the system configuration to or from nonvolatile memory. Before installing new program ROM chips to upgrade the system, record the system configuration as all configuration parameters in system memory will be cleared.

```
Gordon Kapes, Inc.     System 930 Telephony Simulator        Screen 20

                   SAVE/RESTORE SYSTEM CONFIGURATION

Action:           --
Selected Profile: 1

No.  Profile Description
1    FACTORY DEFAULTS
2    FACTORY DEFAULTS
3    FACTORY DEFAULTS
4    FACTORY DEFAULTS

Active Profile:   1 - FACTORY DEFAULTS
Active Status:    DOES NOT MATCH SYSTEM CONFIGURATION

Activating an action item stops call processing.




               Press Space Bar or Backspace then <ENTER>
                       Press <F1> help, <F2> exit
```

Help for Save/Restore System Configuration

Action: Select hyphens (--), SAVE SYSTEM CONFIGURATION INTO SELECTED PROFILE, RESTORE SELECTED PROFILE INTO SYSTEM CONFIGURATION, RESTORE FACTORY DEFAULTS AND RESTART SYSTEM, SEND SELECTED PROFILE TO OUTPUT DEVICE, RECEIVE INCOMING DATA AND STORE INTO SELECTED PROFILE, or RESTART SYSTEM. <ENTER> must be pressed to initiate selected action. Hyphens (--) indicate no selection. Save system configuration into selected profile writes the current system configuration into the selected profile. Restore selected profile into system configuration loads the selected profile as the current system configuration. If the version numbers match, the system is restarted and the screen is cleared. If the version numbers do not match, NOT RESTORED – WRONG VERSION NUMBER appears on the action line. Restore factory defaults into system configuration loads the factory default values into the system configuration but not the profile database. The system is then restarted, which clears the screen. Restore factory defaults and profile database loads the factory default values into the system configuration and profile database. The system is then restarted, which clears the screen. Send selected profile to output device displays a special screen that gives further instructions regarding sending a profile to a remote storage device. Receive incoming data and store into selected profile displays a special screen that gives further instructions regarding receiving a profile from a remote storage device. Restart system restarts the system, which clears the screen.

Selected Profile: Select 1 through 4. Selects the profile to be acted upon by the action or new description function.

Profile Description: Enter text to uniquely describe a profile. Hyphens (--) indicate no text has been entered. Press <ENTER> to write the new name into the selected profile. This field can not be changed until the profile has been saved.

Save/Restore System Configuration

Help for Save/Restore System Configuration (continued)

Active Profile: Shows 1 through 4 and its associated description. Indicates the most recently loaded or saved profile. When not factory default, the associated description is also displayed at the top of each screen.

Active Status: Shows MATCHES SYSTEM CONFIGURATION or DOES NOT MATCH SYSTEM CONFIGURATION. Indicates whether the active profile matches the current system configuration.

Note: Profiles saved by version 1.10 and up are upward compatible with this software version. New features may or may not be initialized to their factory default settings. Manually set them to their factory default or strange behavior may result.

Call Generator Configuration—Digital Interface

This screen tells the call generator how to initiate a call through the digital interface, when to terminate it, and what to do after the call is connected. There are two digital interface configuration screens, one for each digital interface. The screens may differ, depending on whether the interface is configured for GR-303 Network Side, GR-303 User Side, ISDN, NFAS, or RBS. Interface 2 is not used when NFAS signaling is enabled.

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 21-1

          CALL GENERATOR CONFIGURATION - DIGITAL INTERFACE 1

Call Generator Mode:         NOT RUNNING*
Number of Active Calls:      0*
Time Between Calls/Cycles:   5       SECONDS
Call Start Interval:         0*      MILLISECONDS
Called Number:               55500[01+99]
                             --
Called Type of Number:       UNKNOWN NUMBER*
Mode:                        ROLLING*
Maximum Number of Calls:     --
Setup Time Limit:            60  SECONDS
Proceed Time Limit:          60  SECONDS
Progress/Alert Time Limit:   60  SECONDS
Connect Action:              CONNECT ACTION 1*

Call Generator Status:       NOT RUNNING

* Indicates factory default.

              Press Space Bar or Backspace then <ENTER>
          Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

ISDN Call Generator Configuration–Digital Interface Screen

```
Gordon Kapes, Inc.    System 930 Telephony Simulator         Screen 21-1

          CALL GENERATOR CONFIGURATION - DIGITAL INTERFACE 1

Call Generator Mode:         NOT RUNNING*
Number of Active Calls:      0*
Time Between Calls/Cycles:   5       SECONDS
Call Start Interval:         0*      MILLISECONDS
Called Number:               55500[01+99]
                               -
Mode:                        ROLLING*
Maximum Number of Calls:     -
Setup Time Limit:            60  SECONDS
Connect Action:              CONNECT ACTION 1*

Call Generator Status:       NOT RUNNING




* Indicates factory default.

              Press Space Bar or Backspace then <ENTER>
          Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

RBS Call Generator Configuration–Digital Interface Screen

Call Generator Configuration—Digital Interface

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 21-1

          CALL GENERATOR CONFIGURATION - DIGITAL INTERFACE 1

Call Generator Mode:          NOT RUNNING*
Number of Active Calls:       0*
Time Between Calls/Cycles:    5        SECONDS
Call Start Interval:          0*       MILLISECONDS
Called Number:                55500[01+99]
                              --
Mode:                         ROLLING*
Maximum Number of Calls:      --
Setup Time Limit:             60   SECONDS
Proceed Time Limit:           60   SECONDS
Progress/Alert Time Limit:    60   SECONDS
Connect Action:               CONNECT ACTION 1*

Call Generator Status:        NOT RUNNING


* Indicates factory default.

              Press Space Bar or Backspace then <ENTER>
          Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

GR-303 IDT Call Generator Configuration–Digital Interface Screen

```
Gordon Kapes, Inc.   System 930 Telephony Simulator        Screen 21-1

          CALL GENERATOR CONFIGURATION - DIGITAL INTERFACE 1

Call Generator Mode:          NOT RUNNING*
Number of Active Calls:       0*
Time Between Calls/Cycles:    5        SECONDS
Call Start Interval:          0*       MILLISECONDS
Called Number:                55500[01+99]
                              --
RDT Terminal Number:          [1001+2048]
Mode:                         ROLLING*
Maximum Number of Calls:      --
Setup Time Limit:             60   SECONDS
Proceed Time Limit:           60   SECONDS
Progress/Alert Time Limit:    60   SECONDS
Connect Action:               CONNECT ACTION 1*

Call Generator Status:        NOT RUNNING

* Indicates factory default.

              Press Space Bar or Backspace then <ENTER>
          Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

GR-303 RDT Call Generator Configuration–Digital Interface Screen

Call Generator Configuration—Digital Interface

Help for Call Generator Configuration—Digital Interface

Call Generator Mode: Select NOT RUNNING* (default), RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING – UNTIL NO CONNECT. This is a global selection that affects all digital interfaces and analog ports. Upon power up or reboot, the call generator is not running. Not running immediately disconnects all calls started by the call generator and prevents the call generator from starting new calls. Running – start new calls allows the call generator to start new calls. By design the call generator waits the selected time between cycles before making the first call. Running – no new calls prevents new calls from starting but does not disconnect calls in process. Running – until no connect prevents new calls from starting after a call has failed to connect but does not disconnect calls in process. <ENTER> must be pressed to change this feature.

Number of Active Calls: Select 0* (default) through maximum number of destination channels. Indicates the maximum number of outgoing calls that can be initiated by the call generator. 0 indicates that no calls are initiated.

Time Between Calls/Cycles: Enter 1 through 999. Factory default is 5 seconds. Indicates the time until the first call begins. Also indicates the time interval after a call is complete until the next call begins (rolling) or time interval after last call is complete until the next cycle begins (burst). A minimum time, a hyphen (-), and a maximum time creates a random time. For example: 10-30 indicates to wait a minimum of 10 seconds and maximum of 30 seconds.

Call Start Interval: Select 0* (default) to 5000. Indicates minimum time in milliseconds between call start ups. Called Number: Enter up to 47 digits and special characters. Digits 0123456789*# are allowed. The factory default is 55500[01+99]. This is the called number sent during call setup. Hyphens (--) indicate no number entered (default on row two). The called number should not contain an access digit (8 or 9) and is not processed by the Analog Port Outbound Call Configuration screen. Two rows are provided to allow more groups (discussed later). See Calling Number Configuration screen for other parameters sent during call setup. The system can send and receive up to 31 digits.

Digits [+-]?; are special characters. Sequential numbers may be created using [min+max]. The sequential number is reset to its minimum value when the call generator is started. Random numbers may be created using [min-max]. Left bracket, plus or minus, and right bracket are required. Example: 55500[01+99] sends a sequential number between 5550001 and 5550099. The minimum and maximum values are limited to four digits each, may range from 0 through 9999, and do not require the same number of digits. The minimum and maximum values may not contain * or #. Brackets may be repeated, but not nested.

Groups may be created using number?certainty;. The number after question mark (?) determines the certainty factor that the number will be sent. Semicolon (;) separates each group. If the certainty factor is missing, it is assumed to be 100 percent. Only one group per call is sent. The system compares the certainty factor to a random number between 1 and 100. If the certainty factor is greater than or equal to the random number then the number is sent. If the certainty factor is less than the random number then the next group is analyzed. No called number is sent if the list of groups has been exhausted. Up to 16 groups are allowed. Example: 911?10;611?30;5551212 sends 911 ten percent of the time, 611 thirty percent of the time, and 5551212 sixty percent of the time. The certainty factor is averaged over many cycles, the number of times a group is chosen is random within a cycle. Random numbers and sequential numbers are allowed within each group.

GR-303 IDT sends the last four digits of the called number as the terminal number. Terminal numbers may range from 0001 to 2048. This identifies the RDT (remote digital terminal) to receive the call. No DTMF digits are sent. GR-303 RDT sends the called number using DTMF dialing.

RDT Terminal Number: Enter up to 15 digits and special characters. Digits 0123456789 are allowed. The factory default is [1001+2048]. This identifies the GR-303 RDT (remote digital terminal) sending the call. Terminal numbers may range from 0001 to 2048. Hyphens (--) indicate no number entered. Shown for GR-303 RDT digital interfaces only.

Call Generator Configuration—Digital Interface

Help for Call Generator Configuration—Digital Interface (continued)

Digits [+-] are special characters. Sequential numbers may be created using [min+max]. The sequential number is reset to its minimum value when the call generator is started. Random numbers may be created using [min-max]. Left bracket, plus or minus, and right bracket are required. Example: 00[01+99] sends a sequential number between 0001 and 0099. The minimum and maximum values are limited to four digits each, may range from 0 through 9999, and do not require the same number of digits. The minimum and maximum values may not contain * or #. Brackets may be repeated, but not nested.

Called Type of Number: Select UNKNOWN NUMBER* (default), INTERNATIONAL NUMBER, NATIONAL NUMBER, SUBSCRIBER NUMBER, or ABBREVIATED NUMBER. Shown for non-GR-303 ISDN digital interfaces only. Called type of number is sent in the called party number information element of the ISDN setup message. Unknown number is either not identified or contains prefix digits that are not part of the called number. Example: *70 to disable call waiting. International number includes a country code. Example: 443125551212. National number includes a national area code. Example: 3125551212. Subscriber number includes a local exchange number. Example: 5551212. Abbreviated number is 3 to 5 digits long. Example: 1212 or 51212.

Mode: Select ROLLING* (default) or BURST. Rolling indicates that a new batch of calls can begin even though calls from the current batch have not ended. The time between cycles must elapse before a new batch of calls are started. Burst indicates that a new batch of calls cannot begin until all calls from the current batch have ended. The time between cycles must elapse before a new batch of calls are started.

Time limits tell the call generator how long to wait for an acknowledgment after a request has been sent before determining that the other side has failed and the call should be disconnected.

Maximum Number of Calls: Enter 0 through 9 digits. Hyphens (--) indicate no number entered (default). This is the maximum number of calls that may be started before the call generator stops running. No number indicates the number of calls is unlimited. When exceeded, the call counters on the Call Counter Screen must be reset and the call generator mode changed to RUNNING – START NEW CALLS or RUNNING – UNTIL NO CONNECT to start new calls.

Setup Time Limit: Enter 1 through 999. Factory default is 60 seconds. Setup time is an ISDN term, but also applies to RBS. This is the time limit that the call generator will wait for a setup ack, proceed, alert, connect, or disconnect response after a setup request is sent. If one is not received, release complete is sent.

Proceed Time Limit: Enter 1 through 999. Factory default is 60 seconds. This is the time limit that the call generator will wait for an alert, connect, or disconnect response after a proceed response is received. If one is not received, a disconnect request is sent. Shown for ISDN digital interfaces only.

Progress/Alert Time Limit: Enter 1 through 999. Factory default is 60 seconds. This is the time limit that the call generator will wait for a connect or disconnect response. After a progress or alert response is received. If one is not received, a disconnect request is sent. Shown for ISDN digital interfaces only.

Connect Action: Select CONNECT ACTION 1 through CONNECT ACTION 3, or NONE. The factory default for digital interface 1 and digital interface 2 is CONNECT ACTION 1* and CONNECT ACTION 2* respectively. Instructs the call generator what to do after a connection has been established with the called party. The connect action is configured on the connect action screen. None indicates that the call is connected but no connect action is performed.

Call Generator Status: Shows NOT RUNNING, NOT RUNNING – MAXIMUM NUMBER OF CALLS REACHED, RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING – UNTIL NO CONNECT. Indicates current call generator status.

Call Generator Configuration—Analog Ports

This screen tells the call generator how to initiate a call through the analog ports, when to terminate it, and what to do after the call is connected. This screen is specific to the analog ports.

```
Gordon Kapes, Inc.    System 930 Telephony Simulator       Screen 21-3

              CALL GENERATOR CONFIGURATION - ANALOG PORTS

Call Generator Mode:         NOT RUNNING*
Number of Active Calls:      0*
Time Between Calls/Cycles:   5     SECONDS
Call Start Interval:         0*    MILLISECONDS
Called Number:               9,55500[01+99]
Call Routing:                ALL PORTS*
Mode:                        ROLLING*
Maximum Number of Calls:     -
Alert Time Limit:            60    SECONDS
Connect Action:              CONNECT ACTION 3*

Call Generator Status:   NOT RUNNING




* Indicates factory default.

               Press Space Bar or Backspace then <ENTER>
         Press Arrow Keys, <F1> help, <F2> exit, <F3> previous
```

Help for Call Generator Configuration—Analog Ports

Call Generator Mode: Select NOT RUNNING* (default), RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING – UNTIL NO CONNECT. This is a global selection that affects all digital interfaces and analog ports. Upon power up or reboot, the call generator is not running. Not running immediately disconnects all calls started by the call generator and prevents the call generator from starting new calls. Running – start new calls allows the call generator to start new calls. By design the call generator waits the selected time between cycles before making the first call. Running – no new calls prevents new calls from starting but does not disconnect calls in process. Running – until no connect prevents new calls from starting after a call has failed to connect but does not disconnect calls in process. <ENTER> must be pressed to change this feature.

Number of Active Calls: Select 0* (default) through maximum number of analog ports. Indicates the maximum number of outgoing calls that can be initiated by the call generator. 0 indicates that no calls are initiated.

Time Between Calls/Cycles: Enter 1 through 999. Factory default is 5 seconds. Indicates the time until the first call begins. Also indicates the time interval after a call is complete until the next call begins (rolling) or time interval after last call is complete until the next cycle begins (burst). A minimum time, a hyphen (-), and a maximum time creates a random time. For example: 10-30 indicates to wait a minimum of 10 seconds and maximum of 30 seconds.

Call Start Interval: Select 0* (default) to 5000. Indicates minimum time in milliseconds between call start ups.

Call Generator Configuration—Analog Ports

Help for Call Generator Configuration—Analog Ports (continued)

Called Number: Enter up to 47 digits and special characters. Digits 0123456789*# are allowed. The factory default is 9,55500[01+99]. Hyphens (--) indicate no number entered. This field is used by the 939 analog card only. It contains the number to be DTMF dialed when the call is initiated by the call generator. The called number may contain an access digit (8 or 9) and is not processed by the Analog Port Outbound Call Configuration screen. The system can send up to 31 DTMF digits.

Digits ,[+-] are special characters. Comma (,) causes DTMF dialing to pause for two seconds. Sequential numbers may be created using [min+max]. The sequential number is reset to its minimum value when the call generator is started. Random numbers may be created using [min-max]. Min and max represent the minimum and maximum values of the number. Left bracket, plus or minus, and right bracket are required. For example: 55500[01+99] indicates to send a sequential number between 5550001 and 5550099. The minimum and maximum values are limited to four digits each, may range from 0 through 9999, and do not require the same number of digits. The minimum and maximum values may not contain * or #. Brackets may be repeated, but not nested.

Call Routing: Select ALL PORTS* (default), or ACD 1 through ACD 32. All ports indicates that calls can be made to all analog ports. ACD 1 through ACD 32 indicates that calls are restricted to analog ports assigned to the ACD on the Analog Port Configuration screen. Analog ports are chosen in ascending sequential order starting from port 1. If no analog ports are available, the call is not made. Resources on the ACD Configuration screen are not used.

Mode: Select ROLLING* (default) or BURST. Rolling indicates that calls are started with a minimum 50ms interval between any two calls. Upon completion, the time between cycles must elapse before the call is restarted. Burst indicates that all calls will start simultaneously. All calls must end and the time between cycles must elapse before a new batch of calls are started.

Maximum Number of Calls: Enter 0 through 9 digits. Hyphens (--) indicate no number entered (default). This is the maximum number of calls that may be started before the call generator stops running. No number indicates the number of calls is unlimited. When exceeded, the call counters on the Call Counter Screen must be reset and the call generator mode changed to RUNNING – START NEW CALLS or RUNNING – UNTIL NO CONNECT to start new calls.

Alert Time Limit: Enter 1 through 999. Factory default is 60 seconds. This is the time limit that the call generator will wait for the call to be answered after being alerted. If the call is not answered within the specified time, alerting stops.

Connect Action: Select CONNECT ACTION 1 through CONNECT ACTION 3* (default), or NONE. Instructs the call generator what to do after a connection has been established with the called party. The connect action is configured on the connect action screen. None indicates that the call is connected but no connect action is performed.

Call Generator Status: Shows NOT RUNNING, NOT RUNNING – MAXIMUM NUMBER OF CALLS REACHED, RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING – UNTIL NO CONNECT. Indicates current call generator status.

Call Counters—Digital Interface

This screen displays the call generator's current status and historical data for each digital interface. Statistics are saved during power down. It also starts or stops the call generator and allows historical data to be reset. <ENTER> must be pressed to changed these features. There are two digital interface call counter screens, one for each digital interface. Screen 2 is not used when NFAS signaling is enabled.

```
Gordon Kapes, Inc.    System 930 Telephony Simulator         Screen 22-1

                 CALL COUNTERS - DIGITAL INTERFACE 1

Call Generator Mode:          NOT RUNNING*
Reset Counters:               NO*
Enable Counters:              ENABLED*
Call Generator Status:        NOT RUNNING
Call Generator Cycle:         0        Call Generator Active Calls:  0

                     Outbound              Incoming
                 Sent    Received      Sent      Received
Setup:            0        ---          ---         0
Setup Ack:        ---      0            0           ---
Proceed:          ---      0            0           ---
Progress:         ---      0            0           ---
Alert:            ---      0            0           ---
Connect:          0        0            0           0
Action:           0        ---          0           ---
Disconnect:       0        0            0           0
Release:          0        0            0           0
Abnormal Release: ---      0            ---         0
              Press Space Bar or Backspace then <ENTER>
          Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

Help for Call Counters—Digital Interface

Call Generator Mode: Select NOT RUNNING* (default), RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING – UNTIL NO CONNECT. This is a global selection that affects all digital interfaces and analog ports. Upon power up or reboot, the call generator is not running. Not running immediately disconnects all calls started by the call generator and prevents the call generator from starting new calls. Running – start new calls allows the call generator to start new calls. By design the call generator waits the selected time between cycles before making the first call. Running – no new calls prevents new calls from starting but does not disconnect calls in process. Running – until no connect prevents new calls from starting after a call has failed to connect but does not disconnect calls in process. <ENTER> must be pressed to change this feature.

Reset Counters: Select NO* (default), THIS SCREEN ONLY, or ALL CALL COUNTER SCREENS. The selections allow counters on this screen or all screens to be reset. <ENTER> must be pressed to reset counters.

Enable Counters: Select ENABLED* (default), or DISABLED. This selection enables or disables counters to accumulate information. By design the counters accumulate information for all calls, not just calls generated by the call generator.

Call Generator Status: Shows NOT RUNNING, NOT RUNNING – MAXIMUM NUMBER OF CALLS REACHED, RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING – UNTIL NO CONNECT. Indicates current call generator status.

Call Generator Cycle: Shows the number of times the call generator has started a new batch of calls. This applies to both rolling and batch modes. Call Generator Active Calls: Shows current number of outgoing

Call Counters—Digital Interface

Help for Call Counters—Digital Interface (continued)

calls that were initiated by the call generator. Burst mode won't start until this number is zero and the time between cycles has elapsed.

The next few status lines shows the number of outbound and incoming calls. Outbound calls are calls initiated by the system. Incoming calls are calls terminated by the system. The status lines show counters indicating the number of requests sent and responses received. The sent count indicates that an attempt was made to send the request but the request may not actually be sent due to the call state. The received count indicates the number of responses accepted by the system. A response may be received but not accepted due to the call state. Counters increment from 0 through 4294967295 then wrap around to 0.

Setup: Shows the number of call setups sent and received by the system. Setup is used to start a call.

Setup Ack: Shows the number of setup acknowledges sent and received by the system. Setup acknowledge is used by GR-303 and Robbed Bit Signaling to indicate that the system is ready to receive a called number.

Proceed: Shows the number of call proceedings sent and received by the system. Proceed indicates that the called number or lack thereof is being processed. Used by ISDN only. Not used by GR-303 or Robbed Bit Signaling.

Progress: Shows the number of progress states sent and received by the system. Progress indicates that the called number is not being alerted. Used by ISDN only. Not used by GR-303 or Robbed Bit Signaling.

Alert: Shows the number of alert states sent and received by the system. Alerting indicates that the called party is being alerted. Used by ISDN and GR-303 only. Not used by Robbed Bit Signaling.

Connect: Shows the number of connect states sent and received by the system. Connect indicates that the called party is connected. Action: Shows the number of times the C command was executed from an outgoing or incoming call connect action.

Disconnect: Shows the number of disconnects sent and received by the system. Disconnect indicates to disconnect the bearer channel.

Release: Shows the number of release states sent and received by the system. Release indicates to terminate the call. It may be initiated by a release or release complete message. Used by ISDN and GR-303 only. Not used by Robbed Bit Signaling.

Abnormal Release: Shows the number of unexpected release completes received by the system. It is used to terminate a call quickly. Used by ISDN and GR-303 only. Not used by Robbed Bit Signaling.

Call Counters—Analog Ports

This screen displays the call generator current status and historical data for the analog ports. Statistics are saved during power down. It also starts or stops the call generator and allows historical data to be reset. <ENTER> must be pressed to changed these features. This screen is specific to the analog ports.

```
Gordon Kapes, Inc.     System 930 Telephony Simulator         Screen 22-3

                     CALL COUNTERS - ANALOG PORTS

Call Generator Mode:   NOT RUNNING*
Reset Counters:        NO*
Enable Counters:       ENABLED*

Call Generator Status:        NOT RUNNING
Call Generator Cycle:         0        Call Generator Active Calls:  0

                   Outbound                Incoming
               Sent      Received      Sent      Received
Alert:          ---        ---          0           0
Connect:        0          0            0           0
Action:         0          ---          0           ---
Disconnect:     0          0            0           0



* Indicates factory default.

             Press Space Bar or Backspace then <ENTER>
        Press Arrow Keys, <F1> help, <F2> exit, <F3> previous
```

Help for Call Counters—Analog Ports

Call Generator Mode: Select NOT RUNNING* (default), RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING – UNTIL NO CONNECT. This is a global selection that affects all digital interfaces and analog ports. Upon power up or reboot, the call generator is not running. Not running immediately disconnects all calls started by the call generator and prevents the call generator from starting new calls. Running – start new calls allows the call generator to start new calls. By design the call generator waits the selected time between cycles before making the first call. Running – no new calls prevents new calls from starting but does not disconnect calls in process. Running – until no connect prevents new calls from starting after a call has failed to connect but does not disconnect calls in process. <ENTER> must be pressed to change this feature.

Reset Counters: Select NO* (default), THIS SCREEN ONLY, or ALL CALL COUNTER SCREENS. The selections allow counters on this screen or all screens to be reset. <ENTER> must be pressed to reset counters.

Enable Counters: Select ENABLED* (default), or DISABLED. This selection enables or disables counters to accumulate information. By design the counters accumulate information for all calls, not just calls generated by the call generator.

Call Generator Status: Shows NOT RUNNING, NOT RUNNING – MAXIMUM NUMBER OF CALLS REACHED, RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING – UNTIL NO CONNECT. Indicates current call generator status.

Call Generator Cycle: Shows the number of times the call generator has started a new batch of calls. This applies to both rolling and batch modes.

Call Counters—Analog Ports

Help for Call Counters—Analog Ports (continued)

Call Generator Active Calls: Shows current number of outgoing calls that were initiated by the call generator. Burst mode won't start until this number is zero and the time between cycles has elapsed.

The next few status lines shows the number of outbound and incoming calls. Outbound calls are calls initiated by the system. Incoming calls are calls terminated by the system. The status lines also show counters indicating the number of requests sent and responses received. Counters increment from 0 through 4294967295 then wrap around to 0.

Alert: Shows the number of alert states sent by the system. Alert indicates the presence of ring voltage.

Connect: Shows the number of connect states sent and received by the system. Connect indicates the off-hook condition.

Action: Shows the number of times the C command was executed from an outgoing or incoming connect action.

Disconnect Sent: Shows the number of disconnects sent and received by the system. Disconnect sent indicates the break loop condition. Disconnect received indicates the on-hook condition.

Connect Action Configuration

This screen instructs the system what to do after a call is routed to connect action and a connection has been established with the called party. It also starts or stops the call generator. There are three screens, one for each connect action.

```
Gordon Kapes, Inc.    System 930 Telephony Simulator       Screen 23-1

                    CONNECT ACTION 1 CONFIGURATION

Call Generator Mode:    NOT RUNNING*
Connect Action:
  T19,1;W10;C;H
  --
  --
  --
  --

Call Generator Status:  NOT RUNNING







* Indicates factory default.

                 Press Space Bar or Backspace then <ENTER>
             Press Arrow Keys, <F1> help, <F2> exit, <F4> next
```

Help for Connect Action Configuration

Call Generator Mode: Select NOT RUNNING* (default), RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING – UNTIL NO CONNECT. This is a global selection that affects all digital interfaces and analog ports. Upon power up or reboot, the call generator is not running. Not running immediately disconnects all calls started by the call generator and prevents the call generator from starting new calls. Running – start new calls allows the call generator to start new calls. By design the call generator waits the selected time between cycles before making the first call. Running – no new calls prevents new calls from starting but does not disconnect calls in process. Running – until no connect prevents new calls from starting after a call has failed to connect but does not disconnect calls in process. <ENTER> must be pressed to change this feature. Connect Action: Instructs the system what to do after a connection has been established with the called party. Enter up to five lines of commands. Factory default (T19,1;W10; C;H) sends tone 19 (440Hz) for one second, waits 10 seconds, increments the action counter on the call counter screen, then disconnects. Hyphens (--) indicate no entry has been made. Eight commands are available: C for count, D for dial, H for hangup, M for message connect, S for silence detection, T for tone connect, V for voice detection, and W for wait. Use semicolon (;) to append commands together. A single command may not be spread over two lines. Commands are executed starting from the top line.

Count: C. This command increments the action counter by one on the call counter screen.

Dial: D followed by the DTMF digits to dial. There are no default DTMF digits. This command sends in-band DTMF tones. Each tone is on for 100 ms, with 100 ms between tones. Digits 0123456789*# are allowed. The number of digits is limited only by the line length. Sequential numbers may be created using [min+max]. The sequential number is reset to its minimum value when the call generator is started. Random numbers may be created using [min-max]. Min and max represent the minimum and maximum

Connect Action Configuration

Help for Connect Action Configuration (continued)

values of the random number. A left bracket, plus or minus, and right bracket are required. For example: D55512[00-99] indicates to dial a number between 5551200 and 5551299. The minimum and maximum values are limited to four digits each, may range from 0 through 9999, and do not require the same number of digits. The minimum and maximum values may not contain * or #. Brackets may be repeated, but not nested.

Hangup: H followed by the disconnect certainty factor in percent. The disconnect certainty factor may range from 0 to 99 percent. If certainty factor is not specified, command will always cause disconnect, canceling any remaining commands. The disconnect certainty factor is used to create randomization. Sometimes the call will disconnect and sometimes it won't. If the call doesn't disconnect, the next command is allowed to execute. Obviously, if the call disconnects the remaining commands are not executed. For example: H50 indicates to disconnect 50 percent of the time. Message Connect: M followed by the message number, comma, and time in seconds. This command sends an in-band voice message using the recorder/announcer resource. The message number may be 1 or 2. If message number is not specified, the default is message 1. If message time is not specified, the specified message is started immediately and plays once until the message ends. Each message has a maximum length of 20 seconds. Time may range from 1 through 999 seconds. If the message is shorter than the specified time, the message will simply repeat until the time expires. A hyphen (-) between two times creates a random time. For example: M1,10-30 causes message 1 to play for a minimum of 10 seconds and a maximum of 30 seconds.

Silence Detection: S followed by time in seconds, comma, and audio threshold. This command waits for silence to be detected for the specified time. If time is not specified, the default is 3 seconds. Time may range from 1 through 999 seconds. Audio threshold may range from 1 through 127. If the audio threshold is not specified, the default is 89 (59 hex) for mu-law and 42 (2A hex) for A-law. This corresponds to 30% of the maximum audio level. Silence is detected when the absolute value of the digital sample is above the audio threshold. It is recommended that audio be sent for a minimum of 3 seconds in order for the S command to detect it. Audio is sampled 20 times per second and if the time duration is too short, energy may not be detected. Example: S10 waits for 10 seconds of silence. If silence is not detected for the specified time, no further commands are processed. A hyphen (-) between two times creates a random time. For example: S10-30 waits for silence to be detected for a minimum of 10 seconds and maximum of 30 seconds.

Voice Detection: V followed by time in seconds, comma, and audio threshold. This command waits for voice energy to exceed the minimum audio threshold followed by an interval of silence for the specified time. If time is not specified, the default is 1 second. Time may range from 1 through 999 seconds. Audio threshold may range from 1 through 127. If the audio threshold is not specified, the default is 76 (4C hex) for mu-law and 29 (1D hex) for A-law. This corresponds to 40% of the maximum audio level. Audio is detected when the absolute value of the digital sample is below the audio threshold. It is recommended that audio be sent for a minimum of 3 seconds in order for the V command to detect it. Audio is sampled 20 times per second and if the time duration is too short, energy may not be detected. Example: V10 waits for voice energy, followed by 10 seconds of silence. If voice energy is not detected followed by silence for the specified time, no further commands are processed. A hyphen (-) between two times creates a random time. For example: V10-30 waits for voice energy to be detected followed silence to be detected for a minimum of 10 seconds and a maximum of 30 seconds.

Tone Connect: T followed by the tone number, comma, and time in seconds. If tone number is not specified, the default is tone 19 (440Hz). Time may range from 1 through 999 seconds. If time is not specified, the default is 1 second. This command sends an in-band tone using the tone facility. A hyphen (-) between two times creates a random time. For example: T19,10-30 indicates to send a 440Hz tone for a minimum of 10 seconds and maximum of 30 seconds. See tone list. Wait: W followed by the time in seconds. If time is not specified, the default is 1 second. While waiting, silence is sent on the associated channel. A hyphen (-) between two times creates a random wait time. For example: W10-30 indicates to wait for a minimum of 10 seconds and a maximum of 30 seconds. Time values may range from 1 through 999 seconds.

Connect Action Configuration

Help for Connect Action Configuration (continued)

Command Separator: Semicolon (;). Used to append commands on a line. For example: D123;M1;H indicates to dial DTMF digits 123, play message 1 until the end of the message, and then disconnect.

Tone List

0 - DTMF 0	4 - DTMF 4	8 - DTMF 8	16 - Dial Tone
1 - DTMF 1	5 - DTMF 5	9 - DTMF 9	17 - Audible Ring
2 - DTMF 2	6 - DTMF 6	14 - DTMF *	18 - Busy/Reorder
3 - DTMF 3	7 - DTMF 7	15 - DTMF #	19 - 440Hz

Call Generator Status: Shows NOT RUNNING, NOT RUNNING – MAXIMUM NUMBER OF CALLS REACHED, RUNNING – START NEW CALLS, RUNNING – NO NEW CALLS, or RUNNING UNTIL NO CONNECT. Indicates current call generator status.

Time & Date Configuration

This screen sets the time and date used by data capture display screens and for sending Caller ID.

```
Gordon Kapes, Inc.    System 930 Telephony Simulator          Screen 24

                          TIME & DATE CONFIGURATION

Current Time Zone:  -
Current Year:       03
Current Month:      01
Current Day:        01
Current Hour:       01
Current Minute:     02










* Indicates factory default.

                         Enter Time Zone or Backspace.
                          Press <F1> help, <F2> exit
```

Help for Time & Date Configuration

Time Zone: Enter time zone abbreviation, up to 5 characters. Shows hyphens (--) if no time zone has been entered. Example: UTC for Universal Time Coordinated. Just a reminder and not used by any screens or resources.

Current Year: Select 00 through 99. Current time appears in data capture records. Revision C processor card does not advance time during power down. Revision F and higher processor board maintains time during power down through use of a battery powered clock. Changing any time field resets seconds to 00.

Current Month: Select 01 through 12.

Current Day: Select 01 through 31. Upper limit depends on year and month.

Current Hour: Select 00 through 23.

Current Minute: Select 00 through 59.

939 Analog Card Configuration

This screen configures parameters for the 939 analog card. The 939 analog card emulates eight FXS (subscriber) loop start circuits.

```
Gordon Kapes, Inc.    System 930 Telephony Simulator        Screen 25

                     939 ANALOG CARD CONFIGURATION

Inbound Analog Call Configuration:
Number of Rings:         1*
Route Action:            CONNECT ACTION 3*
Called Number:           55500[01+99]
Called Mode:             SEND CALLED NUMBER*
Type of Number:          UNKNOWN NUMBER*

Outbound Analog Calls Initiated from a Digital Interface:
Analog Dialed Number:    9,55500[01+99]
Analog Dialed Mode:      SEND ANALOG DIALED NUMBER*

General Parameters:
939 Receive Loss:        0dB*
939 Reference Tone:      OFF*
Forced Disconnect Time:  0  SECONDS

* Indicates factory default.

               Press Space Bar or Backspace then <ENTER>
                Press Arrow Keys, <F1> help, <F2> exit
```

Help for 939 Analog Card Configuration

The 939 analog card allows the system to answer an incoming analog call. Incoming analog calls are detected by the presence of alerting voltage. The system then initiates an outbound call according to the selected route action. The calls are then connected together. Call teardown is initiated by a break in analog loop current or a disconnect from the outbound side. The Analog Port Dialing Configuration screen and Analog Port Outbound Call Configuration screen are not used.

The 939 analog card also allows the system to initiate an outbound analog call. Calls may be initiated from one of three sources; an incoming call from a digital interface, the call generator, or an outgoing call from another analog port. To initiate an outbound analog call the 939 analog port goes off-hook, DTMF dials the analog dialed number, and connects the calls together. Digital interface calls are routed to the 939 analog card when route if match on the Inbound Match Configuration screen is set to an ACD or PORT containing a 939 analog port. Call generator calls are routed to the 939 analog card when call routing on the Call Generator Configuration – Analog Ports screen is set to an ACD containing a 939 analog port. Analog port calls are routed to the 939 analog card by dialing the extension of a 939 analog port. When going off-hook, if the 939 analog port does not detect loop current within the outbound first digit timeout time as configured on the Analog Port Dialing Configuration screen, the port goes on-hook and the call is terminated.

Number of Rings: Select 1* (default) through 9. Number of incoming alerting cycles before going off-hook and processing the call.

Route Action: Select CONNECT ACTION 1 through CONNECT ACTION 3* (default), DIGITAL INTERFACE 1, DIGITAL INTERFACE 2, ALTERNATE INTERFACES REORDER, BUSY, MSG 1 PLAY CONTINUOUS, MSG 2 PLAY CONTINUOUS, or ACD 1 through ACD 32. Selects the action taken after the 939 analog port answers a call. Connect action 1 through connect action 3 routes the call to the specified connect action. Digital

939 Analog Card Configuration

Help for 939 Analog Card Configuration (continued)

interface and alternate interfaces sends the call to the specified digital interface. Alternate interfaces uses digital interfaces 1 & 2, alternating between them on every other call. Reorder and busy routes the call to the specified progress tone and remains connected until the caller disconnects. Msg 1 play continuous and msg 2 play continuous routes the call to the specified voice message and plays the message until the caller disconnects. ACD routes the call to the specified ACD group.

Called Number: Enter up to 31 digits and special characters. Digits 0123456789*# are allowed. The factory default is 55500[01+99]. Hyphens (--) indicate no number entered. Used when an inbound call is routed to a digital interface. Determines part of the called number sent to the digital interface. An access digit (8 or 9) is not required. The system can send and receive up to 31 digits.

Digits [+-] are special characters. Sequential numbers may be created using [min+max]. Random numbers may be created using [min-max]. Min and max represent the minimum and maximum values of the number. A left bracket, plus or minus, and right bracket are required. For example: 55512[00-99] indicates to send a random number between 5551200 and 5551299. The minimum and maximum values are limited to four digits each, may range from 0 through 9999, and do not require the same number of digits. The minimum and maximum values may not contain * or #. Brackets may be repeated, but may not be nested.

Called Mode: Select SEND CALLED NUMBER* (default), SEND CALLED NUMBER WITH 1 DIGIT EXT OVERLAY through SEND CALLED NUMBER WITH 4 DIGIT EXT OVERLAY, SEND CALLED NUMBER WITH EXTENSION OVERLAY, or SEND NO NUMBER. Used when an inbound call is routed to a digital interface. Determines the called number sent to the digital interface. Send called number sends the called number. Send called number with 1 digit ext overlay through Send called number with 4 digit ext overlay sends the called number and overlays the last one to four analog port extension digits at the end of the called number. The digits are overlaid from left to right. Send called number with extension overlay sends the called number and overlays up to five analog port extension digits at the end of the called number. Send no number means that there is no number. Example: If this selection is send called number with 2 digit ext overlay, the called number is 5550000, and the extension is 1234, then the number sent is 5550034.

Type of Number: Select UNKNOWN NUMBER* (default), INTERNATIONAL NUMBER, NATIONAL NUMBER, SUBSCRIBER NUMBER, or ABBREVIATED NUMBER. Used when an inbound call is routed to an ISDN digital interface. The digital interface type of number is sent in the called party number information element of the ISDN setup message. Unknown number is either not identified or contains prefix digits that are not part of the called number. Example: *70 to disable call waiting. International number includes a country code. Example: 443125551212. National number includes a national area code. Example: 3125551212. Subscriber number includes a local exchange number. Example: 5551212. Abbreviated number is 3 to 5 digits long. Example: 1212 or 51212.

Analog Dialed Number: Enter up to 31 digits and special characters. Digits 0123456789*# are allowed. The factory default is 9,55500[01+99]. Hyphens (--) indicate no number entered. This determines part of the number to be DTMF dialed by the 939 analog card when calls are initiated from a digital interface. Digital interface calls are routed to the 939 analog card from the Inbound Match Configuration screen when route if match is set to an ACD or PORT containing a 939 analog port. This selection is not used for calls initiated by the call generator or when dialing a 939 analog port from a non-939 analog port. Each DTMF tone is on for 100 ms, with 100 ms between tones. The system can send up to 31 DTMF digits.

Digits ,[+-] are special characters. Comma (,) causes DTMF dialing to pause for two seconds. Sequential numbers may be created using [min+max]. Random numbers may be created using [min-max]. Min and max represent the minimum and maximum values of the number. A left bracket, plus or minus, and right bracket are required. For example: 55512[00-99] indicates to send a random number between 5551200 and 5551299. The minimum and maximum values are limited to four digits each, may range from 0 through 9999, and do not require the same number of digits. The minimum and maximum values may not contain * or #. Brackets may be repeated, but may not be nested. Analog Dialed Mode: Select SEND ANALOG DIALED NUMBER* (default), SEND ANALOG DIALED NUMBER WITH CALLED NUMBER OVERLAY,

939 Analog Card Configuration

Help for 939 Analog Card Configuration (continued)

SEND ANALOG DIALED NUMBER FOLLOWED BY CALLED NUMBER, or SEND NO NUMBER. This determines the complete number to be DTMF dialed by the 939 analog card when calls are initiated from a digital interface. Digital interface calls are routed to the 939 analog card from the Inbound Match Configuration screen when route if match is set to an ACD or PORT containing a 939 analog port. This selection is not used for calls initiated by the call generator or when dialing a 939 analog port from a non-939 analog port. Send analog dialed number allows the analog dialed number to be DTMF dialed. Send analog dialed number with called number overlay DTMF dials the analog dialed number with the called number overlaid at the end of the analog number. The digits are overlaid from left to right. The called number is the called number received from the digital interface. Send analog dialed number followed by called number DTMF dials the analog dialed number followed by the called number. Send no number does not DTMF dial a number.

939 Receive Loss: Select 0dB* (default) or –6dB. 0dB sends the audio level unattenuated. –6dB reduces the audio level sent to all 939 card analog ports.

939 Reference Tone: Select OFF* (default) or ON. Sends a 1 kHz 1 milliwatt (0dB) reference tone to all 939 card analog ports.

Forced Disconnect Time: Enter 0* (default) through 999. Maximum amount of time in seconds that the 939 card will keep a call connected before forcing a disconnect. 0 disables this feature.

Security Configuration

This screen configures access security features. These can be useful when accessing the menu system through a local area network. The security configuration is saved in nonvolatile memory, but not in the profile area and is not affected by the Save/Restore System Configuration screen.

```
Gordon Kapes, Inc.    System 930 Telephony Simulator          Screen 12

                        SECURITY CONFIGURATION

Login Required:            NO*
Password:                  SYS930
Password Reminder:         Default password is SYS930
VT100 Compatibility Test:  NO*
Automatic Logoff:          NO*
Inactivity Time:           120  MINUTES









* Indicates factory default.

                Press Space Bar or Backspace to select
                Press Arrow Keys, <F1> help, <F2> exit
```

Help for Security Configuration

Login Required: Select NO* (default), or YES. No allows the operator to immediately access the main menu. Yes requires the operator to enter the password and, if selected, VT100 compatibility test before accessing the main menu.

Password: Enter password, up to 10 characters, using alphabetic letters, numbers, or punctuation characters. This is the password entered upon login that allows access to the main menu. Factory default password is SYS930.

Password Reminder: Enter password reminder, up to 39 characters. This text is displayed during login. Example: Enter childhood phone number. Factory default is Default password is SYS930.

VT100 Compatibility Test: Select NO* (default), or YES. No skips this feature. Yes asks user to press function keys F1, F2, F3, F4, and backspace after successful password entry. This test ensures that correct keyboard entry can be performed.

Automatic Logoff: Select NO* (default), or YES. No disables automatic logoff, allowing continuous access to the screens. Yes automatically logs the user off when the inactivity time is reached.

Inactivity Time: Enter number from 1 to 999 in minutes. Amount of time that the keyboard must remain inactive before the system automatically logs off. Factory default is 120 minutes.