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U.S. Department of Transportation
Federal Aviation Administration

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Standard

Use of Selective Signaling Standard
for
Voice Communication Systems

FOREWORD

This standard defines the selective signaling (SS) interface protocol requirements between Federal Aviation Administration (FAA) voice communications systems or between FAA voice communications systems and non-FAA voice communications systems (e.g., a military or foreign switch).

A "voice communications system" in this standard is typically a voice switch, but may be a "system", with only one local user interface, which is not a voice switch in the common sense.

This standard identifies (1) general requirements, (2) essential SS functional requirements, (3) additional requirements, (4) physical requirements, (5) optional SS features, and (6) obsolete requirements.

The FAA has three designations for SS circuit operation as described in Section 3.3.4.

This standard shall apply to any new or existing (as required) FAA voice communication system (voice switch) required to use SS. Examples of voice switches currently used by the FAA that employ SS are listed in Appendix 20.

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1. SCOPE

1.1 Scope. This standard defines the selective signaling (SS) interface protocol requirements between FAA-to-FAA voice communications systems or between FAA voice communications systems and Central Telephone Office as illustrated in Figure 1.1.

1.2 Purpose. The purpose of this standard is to define the SS protocol for FAA voice communications systems.

1.3 Introduction. Selective signaling is a signaling system designed for multipoint circuits. SS can be used on point-to-point circuits, but is usually employed on multipoint circuits that allow a caller to dial a two or three-digit dial code corresponding to the destination party. The destination party is alerted of the incoming SS call by ringing or chiming. Additional codes can be dialed to add other parties to the conversation. Any party who accesses a SS circuit currently in use becomes a party to the conversation. Leased line costs are reduced by having multiple parties sharing the same circuit.

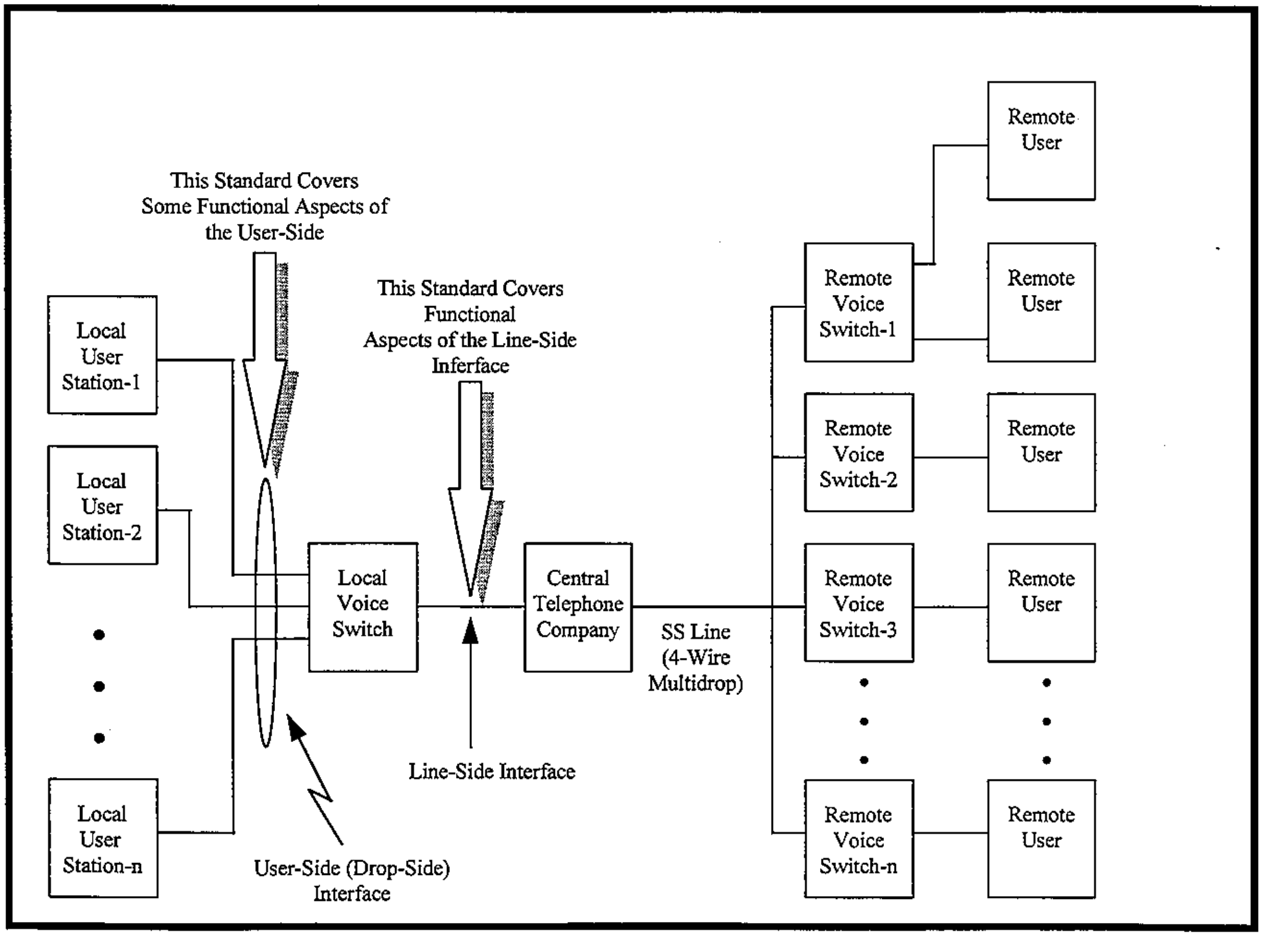

Figure 1-1 Line-Side and User-Side Interfaces

2. APPLICABLE DOCUMENTS

Unless otherwise specified, the following documents form a part of this standard to the extent specified herein. In the event of conflict between the documents referenced herein and the contents of this standard, the contents of this standard shall be considered the superseding requirement.

2.1 Government documents.

PUBLICATIONS

Federal Aviation Administration (FAA) Interface Requirements Documents

NAS-IC-42004200	Voice Switch/Voice Switch Interface, March 4, 1991
NAS-IC-44010002	Transmission Equipment Analog Interface, April 11, 1988

Copies of specifications, standards, drawings and publications required by suppliers in connection with specified procurements functions should be obtained from the procuring activity or as directed by the contracting officer.

2.2 Non-government documents.

STANDARDS

American Telephone and Telegraph (AT&T) Practice Standards:

982-325-100	June 1983, SS-1 Selective Signaling System, General Descriptive Information, Private Line Telephone Service
480-621-502	October 1972, SS-1 Selective Signaling System Decoder and Keyer Detailed Tests, Private Line Telephone Service
982-326-100	June 1983, SS-1A Selective Signaling System, General Descriptive Information, Private Line Telephone Service
982-329-100	June 1983, SS-4 Selective Signaling System, General Descriptive Information, Private Line Telephone Service, (Note: This document is labeled as a "Bell System Practice, AT&T Co. Standard", not an "AT&T Practice Standard".)

480-623-001	June 1983, SS-4 Selective Signaling System With or Without Off Network Access, Identification, Installation and Connections, 4-Wire Private Line Telephone Service

2.3 Document sources. Technical society and technical association specifications and standards are generally available for reference from libraries. They are also distributed among technical groups and users in Federal agencies.

2.3.1 Source of FAA documents. Copies of FAA specifications, standards and publications may be obtained from the Contracting Officer, Federal Aviation Administration, 800 Independence Avenue, S.W., Washington, DC. 20591. Requests should clearly identify the desired material by number and date, and state the intended use of the material.

2.3.2 AT&T documents. Copies of AT&T standards and publications may be obtained from the AT&T Customer Information Center, 2855 N. Franklin Road, Indianapolis, IN. 46219-1385 or by calling the AT&T Customer Information Center at 1-800-432-6600.

3. REQUIREMENTS

3.1 General requirements. SS is employed between some FAA voice switches (intra-FAA interfaces) and between some FAA voice switches and external voice switches/communication systems for Ground/Ground (G/G) air traffic control (ATC). This standard describes the interface between a voice communications systems; Figure 3-1. Typically a voice switch interface is a 4-wire line to which one or more other voice communications systems are connected. The voice communications systems on these lines employ either SS or voice paging in order to establish calls.

Functional requirements are listed below in Section 3.2. Additional requirements beyond those derivable from the AT&T practices are described in Section 3.3. Physical requirements are described in Section 3.4. The SS interface protocol shall conform to the AT&T practices as described in the five AT&T documents which apply explicitly for the Optional and Excluded SS features described in Section 3.5. Optional SS features will be explained in the governing voice switch document. Requirements in AT&T practice standards which do not apply to FAA are listed in Section 3.6.

3.2 Functional requirements. The interface described herein is analog.

The requirements of SS-1, SS-1A and SS-4 as described in the AT&T practices are summarized in Table 3-1.

Table 3-1 SS Requirements

	SS Type (AT&T Practice paragraph numbers are in parentheses)		
TYPES	SS-1 GDI	SS-1A	SS-4
# Digits in Dial Code	2	2	2 or 3 (2.13)
# Dial Codes	81	81	81 or 729 (2.13)
Restricted Digits	1	1	1 (2.13)
Special Dial Codes			"999" is the universal broadcast code (2.43)
Signaling on Line	2400/2600 Hz FSK	2400/2600 Hz FSK	2400/2600 Hz FSK (2.17)
Station Equipment Interface (drop side)	Rotary or TOUCH-TONE	Rotary or TOUCH-TONE	Rotary or TOUCH-TONE (2.08, 2.35)
Location of SS Equipment	CO or Customer Premises (2.02-2.04)	Customer Premises Only	Customer Premises. (GDI 2.01) (IIC 1.09)
Transmission Level Adjustment	No reference	Yes	Yes (2.02)
Interarea Switching	Yes		No (GDI 2.05)
Busy Tone Frequency	No reference	2150 Hz (GDI 2.)	2170 Hz (GDI 2.13, 2.29) 2600 Hz (GDI 2.29, 2.33)
Busy Tone Function	During dialing (2.16). Adjacent system is dialing (2.3.2). Privacy lockout (2.35)		During dialing (GDI 2.29, 2.13, 2.18)

3.2.1 SS-1. This older standard provides speed dialing and network connection for up to 81 user utilizing two-digit codes. This equipment is normally installed at central offices or at the customers premises.

3.2.2 SS-1A. This standard provides the same features as SS-1, except that it may only be installed at the customers premises. Any new voice switch using SS shall conform to the SS-4 specifications listed in Section 2.2 unless otherwise specified in this standard. When replacing existing voice switches, the replacement switch shall be upgraded to SS-4 protocol or it shall continue to conform to its current protocol.

3.2.3 SS-4. SS-4 provides two or three-digit dial codes for up to 729 users and is ostensibly compatible with SS-1 and SS-1A. SS-1, SS-1A and SS-4 drops coexist and communicate among each other on existing SS circuits. However, in order for these systems to communicate with each other, some fine tuning had to be done in the field to adjust for differences in tolerances for signaling characteristics. See Section 3.4.1.2 for signaling tolerance requirements that permit communication among SS-1, SS-1A and SS-4. (Note: The AT&T specifications for SS-4 do not provide sufficient detail to specify the SS protocol. Much of the detail can be found in the AT&T SS-1 and SS-1A practices).

3.2.4 SS reference sources. Various AT&T practices were written to describe the AT&T's specific implementation of SS but these practices were not intended to be used as generic requirements to be followed by a developer of a system that interfaces to an SS circuit. Nonetheless, most requirements for an SS interface are derivable from the AT&T practices. This standard includes these AT&T practices by reference, and modifies the requirements derivable from these practices by adding some requirements, declaring some features to be optional, and excluding some features. Other requirements in this standard were derived from Tellabs technical manuals which have more detail on SS tolerances than the AT&T practices.

The AT&T practices describe:

a) Signaling on the 4-wire multi-station line between Central Offices (CO).
b) Signaling between the CO and the customer premises SS equipment (i.e., the voice switch).
c) Signaling between the customer premises SS equipment and the user's position or station equipment.

Items a) and b) above were described as separate types of signaling because the CO could provide a SS interface, whereby SS tone equipment at the CO would convert signaling from the customer premises equipment (CPE) into a SS format. However, the Leased Interfacility NAS Communications System (LINCS) will not provide equipment to convert CPE signaling into SS. Therefore the CPE must provide a direct SS interface to the multipoint circuit.

3.2.5 Line-side signaling. Line-side signaling interfaces shall be to analog 4-wire multi-station lines in accordance with AT&T SS practices as shown in Figure 3-1. Dial signaling on the line side shall be 2400/2600 Hz FSK tones. There will be no DX, loop, DTMF or 4-wire E&M signaling. See Section 3.4.1.2 for details of signaling characteristics.

3.2.5.1 User-side signaling and functions. The functional operation of the user-side equipment on the voice switch shall conform to the AT&T practices, although the physical circuitry may differ from that described in the AT&T practices. In this context, "functional operation" refers to functions such as:

1) seizing a circuit
2) releasing a circuit
3) dialing on a circuit
4) alerting the user of an incoming call (e.g., ringing, chiming, flashing light or voice call to headset or loudspeaker)
5) establishing talk and receive paths (and push-to-talk, if applicable)
6) provision of side tone
7) quantities of local users that can access the same SS circuit

The AT&T practices refer to the above functions, but any specific requirements for the above functions will be stated in the governing voice switch document.

3.3 Additional requirements. The following sections describe requirements that are not described (or not described in sufficient detail) in the AT&T practices.

3.3.1 Call-Progress tones.

3.3.1.1 Dial confirmation tones. The voice switch shall provide audible confirmation to the calling party that the dialed digits are being outpulsed, e.g., by providing the FSK tones to the calling party's station. The level of these tones shall be specified by the governing voice switch document, e.g., 16 dB attenuation.

3.3.1.2 Busy tone. Busy tone shall be applied in accordance with the governing voice switch document. (Note: Some existing SS equipment generates a busy tone on the drop side to the local user if that local user has accessed the circuit when dialing is in progress by another local or far-end user. See Appendix 10 for additional notes on busy tone).

3.3.1.3 Audible ring tone. The voice switch shall not provide audible ring tone. Audible ring tone is the tone generated by the destination switch as confirmation to the caller that the destination party is being rung. (Audible ring tone is often referred to as "ringback" tone.)

3.3.2 Dialing additional dial codes. This section addresses the dialing of additional dial codes after the first dial code, e.g., dialing codes to add other parties to the call.

3.3.2.1 Dialing additional dial codes without additional operation. In the old AT&T implementation of SS-1A, the station operator must repeatedly seize the register sender to dial successive station codes (paragraph 7.01 of the AT&T SS-1A GDI). However, the voice switch shall not require re-seizure of equipment or any additional operation by the station user prior to dialing additional digits.

3.3.2.2 Additional dial codes without reset or time-out. Both the calling and the receiving switch shall allow additional dial codes to be redialed without requiring the reset digit (i.e., the digit '1') to be dialed and without requiring the six-second dialing time-out to elapse.

3.3.3 Incoming dial code translation. The voice switch shall be capable of providing an incoming dial code translation capability (typically a "translation table") that is unique to each SS circuit when the voice switch interfaces to multiple SS circuits. This will allow flexibility in dial code assignments as described in the following sections and illustrated in Figure 3-1.

3.3.3.1 Dial code validity on a per-circuit basis. The voice switch shall allow a dial code to be a valid local user code on a specific SS circuit and treat the same code as an invalid local user dial code on another SS circuit(s). Basically, when on the same system, you can not use the same dial code assignments.

3.3.3.2 One position with multiple dial codes. The voice switch shall allow any one local position to have different dial codes on different SS circuits. See Note 2 on Figure 3-1.

3.3.3.3 Two positions with the same dial code. The voice switch shall allow any two positions on any two SS circuits to have the same dial code. See Note 3 on Figure 3-1.

3.3.3.4 Unassigned dial code or inactive position. If the dial code received is not assigned to a local user, or if it is assigned but the destination position is inactive (with no call forwarding, if applicable), the voice switch shall not prevent other local users from accessing the circuit. In other words, the voice switch shall not "lock-up" an SS circuit in these circumstances.

3.3.4 Voice paging on SS circuits. The voice switch shall allow outgoing or incoming voice paging on SS circuits on a per circuit basis.

In voice call operation, the SS interface is accessed by the calling party who will announce the name of the destination party. The destination party will hear the announcement (generally on a loudspeaker) and then access the SS interface to communicate with the caller.

The FAA uses the following circuit type designations for SS circuits:

1) Type 4 (SS inbound and outbound)
2) Type 4/5 (SS inbound and has one or more remote users that are dialed plus one or more remote users on the same circuit that are voice paged. The station that is voice paged will hear all FSK dialing on the circuit unless it is filtered out)
3) Type 5 (SS inbound and voice call outbound)

3.3.4.1 Outgoing calls. For outgoing voice calls, the voice switch shall provide cut-through to the circuit so that the user can voice page the far-end user.

3.3.4.2 Incoming calls. For incoming calls on SS circuits designated to receive voice paging, the voice switch shall alert the user of the incoming call by the means defined in the governing voice switch requirements document.

3.3.5 Two and Three digit dial codes on the same SS-4 circuit. Two-digit and three-digit dialing should not be used on the same SS circuit as implied by paragraph 2.13 of the AT&T SS-4 GDI practice. However the voice switch shall allow two and three-digit dialing on a SS-4 circuit during transition periods when voice switch stations on a SS circuit are migrating from two to three-digit dialing. During transition there will be "2-digit switches" and "3-digit switches". It is understood that there is a possibility of misinterpretation of dial codes during transition, e.g., 2 consecutive three-digit codes would be interpreted as 3 two-digit codes by switches that have not yet transitioned to three-digit dialing. Likewise, 3 consecutive two-digit codes intended for "two-digit switches" would be interpreted as 2 three-digit codes by a three-digit switch.

3.3.6 Non-senderized operation. The voice switch shall operate in a non-senderized dialing mode whereby the voice switch signals each digit dialed on the line as soon as the local user dials the digit. In other words, the voice switch shall not use "store and forward" dialing whereby the digits are stored until completion of dialing at which time the dialed digits are signaled on the line. The problems with senderized (store and forward) dialing are twofold: (1) Call setup time is delayed because the first digit is not signaled on the line until the last digit has been dialed; and (2) The end of dialing cannot be easily determined during transitionary periods from two-digit to three-digit dialing.

3.3.7 Automatic clearing digit. The voice switch shall be capable, on a per circuit basis, of automatically transmitting the clearing digit (the digit '1') as a prefix to the digits dialed by the user. Some existing switches automatically send the digit '1' as a prefix to the digits dialed by the controller. The intent of this was to clear the digit decoders of all stations on the line in case the decoders had not reset themselves for some reason. In some cases air traffic controllers routinely dial the digit '1' prior to dialing the dial codes. By not automatically sending the clearing digit, on a per circuit basis, the voice switch can accommodate the mode of operation in which the digit '1' is a valid dial code digit. See Section 3.5.1.3. In cases where sending a leading digit '1' is desirable, this feature frees the controller from having to key that digit.

3.3.8 The "clear" digit. The AT&T practices discuss the use of the clear digit in the context of two-digit dialing, but not in the context of three-digit dialing. When the digit '1' is treated as the "clear" digit, the SS voice switch, upon receipt of the digit '1', shall clear all digits already collected, not just the last digit.

3.3.9 Transmit and receive capability upon access. Upon accessing a SS circuit, the user shall have audio transmit and receive capability (i.e., the user can talk and listen and conduct a conversation with any other party that may be on the line). Depending on the requirements of the governing voice switch document the user may hear a busy tone if dialing is in progress.

3.4 Physical requirements.

3.4.1 Electrical/electronic requirements. See the governing switch document for requirements for electrical interfaces to the SS circuits. (Note: For example, the ETVS Specification, FAA-E-2894, July 26, 1994, references Title 47 CFR Part 68 of the FCC regulations.)

3.4.1.1 Transmission level adjustability range. Voice transmission level requirements are specified in the governing voice switch document. (Note: For example, Section 60.2.6 of the ETVS Specification, FAA-E-2894, July 26, 1994, specifies audio transmission requirements.)

3.4.1.2 Signaling tolerances. In order to help compatibility with existing equipment in the field, the voice switch shall meet the tolerances as listed in Table 3-2.

Table 3-2 Signaling Tolerances

Signaling Characteristic	Standard	Tolerance
Level for FSK tones	-8dBm0	±2dB or tighter Tx. -3dBm0 to -17dBm0 Rx; however a wider tolerance of +3dBm0 to -24dBm0 Rx is preferred.
FSK "break" frequency, nominal 2600Hz	2599-2601Hz	±2 Hz from the standard Tx frequency, i.e. 2597-2603 Hz. 2600 ±30 Hz ≤ Rx tolerance ≤ 2600 ±37 Hz; ±30 Hz is the preferred tolerance.
FSK guard (or "make") frequency, nominal 2400Hz	2396-2404Hz	±4 Hz from the standard Tx frequency, i.e. 2392-2408 Hz. It is not necessary and not recommended to detect the reception of the 2400 Hz tone, but if the voice switch relies on it, it shall accept 2400 ±34 Hz.

(table is continued on the next page)

Table 3-2 Signaling Tolerances (continued)

Signaling Characteristic	Standard	Tolerance
Duration (t) of first 2600 Hz break pulse of a digit.	100 ms	$95 \leq t \leq 105$ ms Tx. $95 \leq t \leq 160$ ms Rx preferred; however $95 \leq t \leq 150$ ms is allowed.
Duration (t) of subsequent 2600Hz break pulses	58 ms	$55 \leq t \leq 65$ ms Tx. $30 \leq t \leq 120$ ms Rx.
Duration of the 2400 Hz guard tone between 2600 Hz pulses	42 ms	$35 \leq t \leq 45$ ms Tx. 20 ms $\leq t \leq$ 90 ms Rx. It is recommended that this interval be treated as absence of the 2600 Hz tone and it is not recommended that the voice switch detect the presence of the 2400 Hz tone.
Duration of the 2400 Hz guard tone after the last 2600 Hz pulse of a digit	42 ms	$35 \leq t \leq 45$ ms Tx. 20 ms $\leq t <$ 6 sec Rx. It is recommended that this interval be treated as absence of the 2600 Hz tone and it is not recommended that the voice switch detect the presence of the 2400 Hz tone. Some voice switches transmit a constant 2400 Hz tone in between digits, therefore the 2400 Hz tone can be nearly six seconds long, which is the interdigit time-out.
Time (t) break between digits. This time starts after the end of the nominal 2400 Hz guard tone. However, if the 2400 Hz tone is continuous throughout the inter-digit period, this time (t) starts 20-90 ms after the end of the 2600 Hz tone.	225 ms	200 ms $\leq t <$ 300 ms Tx (or longer if the caller has not yet dialed the next digit). 150 ms $\leq t <$ 6 sec Rx.

3.5 SS features. Section 3.5.1 identifies and describes Optional SS features.

3.5.1 Optional SS features. AT&T practices listed in Section 2.2 refer to various options or features that are optional. The governing voice switch requirements specification shall state whether they are required. These optional features are described in subsequent paragraphs and summarized in Table 3-3.

Table 3-3 Optional SS Features

Optional Features	AT&T Practice References (Paragraph #)		
	SS-1	SS-1A	SS-4
1) Interarea Switching	GDI 1.05, 2.11, 2.30, 2.31	1.06(b), 3.05	Not allowed. GDI 2.04
2) Three-digit interarea switching	GDI 2.11, 2.32	1.06(a), 3.06	
3) Use of digit '1' as a valid dial code (other than for interarea dialing)	No reference	No reference	No reference
4) Incoming code groups	GDI 3.01		
5) Broadcast code use			GDI 2.43
6) On-premises dialing	GDI 3.04, 4.02	2., 4.07	GDI 2.13e
7) Ring time-out	No reference found	No reference found	IIC 1.08(f); 5.33, Table F, option M
8) Miscellaneous control circuit			IIC 2.26, 5.14, 5.41
9) Loop-back		GDI 1.04	

3.5.1.1 Interarea switching. If required by the governing voice switch document, the voice switch shall join two or more separate SS circuits and allow users on one SS circuit to dial users on another SS circuit when a special two-digit dial code is dialed on a per call basis. Regular two-digit dialing can be used among all stations on the two SS circuits as long as station dial codes in the two joined SS systems are unique. Unique numbering will prevent both stations in the originating and the adjacent system from being simultaneously rung with a single dial code.

3.5.1.1.1 Two-digit "disconnect" dial code. The voice switch shall disconnect the SS circuits when a special two-digit dial code is dialed.

3.5.1.2 Three-digit interarea switching. If required by the governing voice switch document, the voice switch shall accommodate three-digit interarea switching as described in the AT&T practices. This feature allows the two separate SS circuits to be joined without concern for duplication of dial codes. Calls for the attached SS circuit are dialed using three-digit dialing wherein the digit '1' is dialed in between the regular two-digit dial code. This is accomplished by opening up the talking path between the two circuits after the first digit is dialed and then closing the talking path when the second digit ('1') is dialed.

3.5.1.2.1 Local user dialing. With three-digit interarea switching, the originating switch shall disconnect the adjoining switch after the first digit is dialed by a local user and reconnect the adjoining switch after the second digit, if the second digit is a '1'.

3.5.1.3 Use of digit 1 as a valid dial code. The digit '1' is generally not a valid dial code digit. Instead, as described in SS-1 GDI paragraph 2.10, the digit '1' is intended to clear an erroneously dialed first digit. For example, assume a user has established a call and decides to add another party by dialing the code 23. If the user dials '4' as the first digit of the dial code instead of '2', the user can then dial '1' to clear the last digit (i.e. the '4'), and then dial 23. For this reason, there are only 81 valid two-digit dial codes and 729 valid three-digit dial codes.
Certain Canadian switches were implemented to treat the digit '1' as a valid dial code digit. If required by the governing voice switch document, the voice switches shall, on a per SS circuit basis, allow either the standard treatment of the digit '1' or allow the digit '1' to be a valid dial code digit incoming and outgoing. If the digit '1' is used as a valid dial code digit, then there will be 100 valid two-digit dial codes and 1000 valid three-digit codes.

(Note: At least one center, Cleveland, is considering using the digit '1' as a valid dial code digit for positions on the VSCS.)

3.5.1.4 Incoming code groups. If required for incoming code groups, a single incoming dial code shall be interpreted by the voice switch as a dial code corresponding to a group of local users (typically not used at ARTCCs). The maximum number of code groups and the maximum number of stations in each code group shall be specified by the governing voice switch document.

3.5.1.5 Broadcast code use. The AT&T SS-4 GDI practice refers to the dial code "999" as the universal broadcast code. The governing voice switch may require that the dial code "999" be assignable (on a per switch or per circuit basis) as a regular dial code digit and not be treated as the broadcast code.

3.5.1.6 On-premises dialing. Per the AT&T practices, on-premise dialing is the normal mode of operation. With on-premises dialing, two users on the same voice switch can call each other via

SS. The AT&T practice mentions that on-premises dialing can be disabled, in which case more than 81 individual station selections can be obtained on a point-to-point SS circuit. In this mode, the voice switch shall interpret all dial codes dialed by local users as calls to the other switch, not as "intercom" calls to other users at the same site. This allows 162 different users on a SS circuit with two-digit dialing, 81 at each site. Since most FAA voice switches have intercom calling that is integral to the fabric of the switch, the governing voice switch document may declare that on-premises dialing via SS is not required. Also, if the voice switch provides an on-premises SS dialing capability, the governing voice switch document may require the voice switch to disable it.

3.5.1.7 Ring time-out. Incoming ring time-out capability on incoming SS calls shall be required in accordance with the governing voice switch document.

3.5.1.8 Miscellaneous control circuit. If required by the governing voice switch document, the voice switch shall provide a miscellaneous control circuit capability. Per the AT&T practices, the miscellaneous control circuit can be used to disable the privacy override keys or to control various relay-operated devices such as loudspeakers or lamps. Applications cited in the AT&T Practice include "dialup and dialdown" or "turnon and turnoff" arrangements. An FAA application of this is an SS circuit that serves as a backup line to a remote radio site. When the primary line fails, a controller can access the SS circuit and dial a code that will operate relays at both ends of the circuit to switch the equipment at both ends of the circuit to the SS circuit.

3.5.1.9 Loop-back. This is described in the SS-1A practice as a feature that allows someone at a test desk in a CO to order the customer premises SS equipment to loop back the circuit to the CO so that the maintenance personnel at the CO can test the overall loss between the CO and the customer premises. If required in the governing voice document, the voice switch shall provide a remotely activated line-side loopback capability on the SS circuit. The governing voice switch document will specify. The control signal or mechanism for ordering the loopback as well as the signal or mechanism for deactivating the loopback shall be in accordance with the governing voice switch document.

3.6 Obsolete requirements.

3.6.1 Old FAA requirements that are no longer applicable. For the purpose of ensuring compatibility when a new voice switch is installed, the following paragraphs list old FAA requirements that are no longer applicable. Any FAA facility that still employs the interfaces or features described below will have to change upon installation of a new voice switch that complies with this standard.

3.6.2 Ringing on the line side. The ARTCC Center Interphone Switching System specification, FAA-S-2010, November 27, 1964, stated that manual ringing for outbound calls is used in some cases on Type 4 and 5 circuits. A survey of SS circuits at the ARTCCs revealed some SS circuits with 20 Hz ringing outbound or inbound. This standard does not require manual ringing nor 20 Hz ringing, but the governing voice switch specification may require it.

3.6.3 Emergency call. The #300 Interphone Switching System Handbook, 6530.2, 5/9/66, paragraph 32 has a discussion of dial code assignments, presumably for SS circuits since it mentions that the digit '1' is a clearing code. It states that the following dial codes shall be assigned on a national basis:

Dial Code	Position
22	Watch Supervisor
33	Watch Supervisor (emergency)

There is a presumption that the emergency call functions as an override call to the watch supervisor, but the concept of overriding an existing SS call has no meaning on a SS circuit without privacy since anyone who accesses the circuit is automatically cut through for both transmit and receive.

Another possible interpretation is that the override call goes directly into the watch supervisor's headset without the supervisor having to answer the call. However, such a feature apparently does not currently exist in the NAS for SS calls, nor does there appear to be a requirement for this feature.

There is another presumption that the emergency call simply provides a different chime, but this capability has not been found in a poll of the ARTCCs. (There is an emergency call/chime capability on intercom calls on the WECO 300, but evidently not on SS calls.)

Therefore, the "emergency call" feature is not required by this standard.

4. QUALITY ASSURANCE PROVISIONS. This section is not applicable to this standard.

5. PREPARATION FOR DELIVERY. This section is not applicable to this standard.

6. NOTES

6.1 Definitions. The following definitions apply to the terms and acronyms used in this standard:

6.1.1 Glare. A situation where both switches seize the same line at the same time (or nearly the same time). Since both ends of a line are seized simultaneously, the line becomes hung up.

6.1.2 Interface. A common functional and/or physical boundary where hardware, software and/or personnel interact.

6.1.3 Line-side interface. Refers to the interface between the voice switch and the four-wire multipoint circuit.

6.1.4 Multipoint circuit. A single communications channel (typically, a leased telephone circuit) to which more than two stations or other devices are attached.

6.1.5 Selective Signaling. A signaling system designed for point-to-point or multipoint circuits that allows a caller to dial a two or three-digit dial code corresponding to the destination party.

6.1.6 Type 4 circuit. A 4-wire SS circuit type that is SS inbound and outbound.

6.1.7 Type 4/5 circuit. A 4-wire SS circuit type that is SS inbound and voice call or SS outbound.

6.1.8 Type 5 circuit. A 4-wire SS circuit type that is SS inbound and voice call outbound.

6.1.9 Voice communications equipment. The devices that transfer, interpret or process information among people, places or machines.

6.2 Abbreviations and acronyms. The following abbreviations and acronyms are used in this standard.

AFSS	Automated Flight Service Station
ARTCC	Air Route Traffic Control Center
ATC	Air Traffic Control or Air Traffic Controller
ATCT	Air Traffic Control Tower
CO	Central Office
CPE	Customer Premises Equipment
DT	Detailed Test (an AT&T document type)
DTMF	Dual Tone Multifrequency
DX	Duplex
E&M	Ear & Mouth
ETVS	Enhanced Terminal Voice Switch
FAA	Federal Aviation Administration
FCC	Federal Communications Commission
FSK	Frequency Shift Keying
GDI	General Descriptive Information (an AT&T document type)
G/G	Ground/Ground
Hz	Hertz
ICSS	Integrated Communications Switching System
IIC	Identification, Installation and Connections (an AT&T document type)
LINCS	Leased Interfacility NAS Communications System
PBX	Private Branch Exchange
RDVS	Rapid Deployment Voice Switch
Rx	Receive
SS	Selective Signaling
STVS	Small Tower Voice Switch
TMVS	Traffic Management Voice Switch
TRACON	Terminal Radar Approach Control
Tx	Transmit
VSCS	Voice Switching and Control System
WECO	Western Electric Company

APPENDIX 10

SELECTIVE SIGNALING TONE APPLICATIONS

10. SELECTIVE SIGNALING TONE APPLICATIONS

10.1 Frequency Shift Keying (FSK). Signaling on SS-1, SS-1A, and SS-4 circuits is done by FSK. Digit pulses are 2600 Hz tones of 58 ms duration. A guard tone of 2400 Hz is inserted between pulses and at the end of the last pulse (SS-1 GDI paragraphs 2.16-2.18).

In some cases, SS circuits operate in a "voice call" mode rather than by dialing and ringing. In voice call operation, the calling party accesses the SS circuit and announces the name of the destination party. The destination party hears the announcement (generally on a loudspeaker) and then accesses the SS circuit to communicate with the caller.

10.2 Busy tone. Busy tone specifications shall be part of the governing voice switch document. The recommendation of this standard is to apply a 2600 Hz busy tone at -24dBm0 to the local user who accesses a SS circuit when dialing is in progress and when, after a user has already accessed the circuit, dialing is initiated by another party on the circuit. At -8dBm0, the FSK tones would be uncomfortably loud, so a busy tone is recommended instead. A standard busy tone such as that applied on Private Branch Exchange calls may confuse the caller who may think that the tone means that the SS circuit cannot be accessed.

The AT&T Practice Standards are not clear regarding the requirements for the busy tone (see Table 3-1). References to busy tone include:

a) An oscillator is provided to generate 2150 Hz (SS-1A practice, paragraph 2.) and 2170Hz (SS-4 practice, paragraph 2.13).
b) The SS-4 system transmits a busy tone to all stations during the dialing interval (SS-4 GDI 2.13).
c) Upon receipt of the first pulse of any code, the SS-4 System applies tone as a busy signal indication to all telephone receivers until the second digit has been dialed or until the end of the 6-second time-out period when a call is abandoned (SS-4 GDI 2.18).
d) Tone will be present if a private conversation is in progress or another station is dialing (SS-1 2.06).
e) Busy tone is used to indicate if the adjacent system (in the case of interarea dialing) is in the process of dialing or is engaged in a private conversation (SS-1 2.3.2).
f) A low busy tone of 2600 Hz is provided by the oscillator when the privacy feature is used or when a station goes off-hook after another station has started dialing. The oscillator control circuit also provides a low level 2170-Hz tone on the line when an intrusion is made to request release of the system (SS-4 GDI 2.29).
g) 2600-Hz busy tone to indicate privacy or dialing in progress (SS-4 GDI 2.33).
h) Strapping options from SS-4 IIC Table F (p 66) include:
   - SS-4 busy lamp when using automatic or manual privacy system
   - SS-4 privacy busy lamp indicates system busy and in privacy

The Tellabs 334 can be optioned to lockout additional stations and to pass either the busy tone or the FSK tones to the additional stations that accesses a SS circuit while dialing is already in progress.

The VSCS filters FSK tones (per Attachment D, page 23, #19 of the March, 1994 Trunk Anomaly Report, but another source indicated that the VSCS neither filters the 2400 Hz tones nor provides a busy tone.)

It is unclear whether the user is supposed to hear the 2150/2170 Hz busy tone instead of, or in addition to, the FSK. The 2600-Hz Detector Circuit blocks the 2600-Hz tone signal (SS-4 GDI 2.28).

10.3 Glare. There are no specific requirements for handling glare situations.

APPENDIX 20

SS EQUIPMENT MANUFACTURERS

20. SS EQUIPMENT MANUFACTURERS

Denro:	Built their own SS card. SS has always been done internally in the Denro switches. In the Denro Small Tower Voice Switch (STVS), changing from SS-1 or SS-1A to SS-4 is a simple change that can be done at no extra contract cost. It involves changing a hex switch setting and reprogramming an EPROM to go to 3-digit dialing. The switches will not support SS-3. [VSCS Trunk Anomaly Rpt 3/94, Attachment C, p7]
Litton-Amecom:	Used Wescom equipment in older ICSS switches. RDVS has L-A's own SS card.
Tellabs	Tellabs 334 system supports SS-1, SS-1A and SS-4. As of January, 1995 Tellabs has enough parts to build about 300 more systems (where one system connects to one SS circuit) but may not manufacture any more after that, depending on the demand. Tellabs will continue to support SS systems for approximately 10 years.
Wescom:	Manufactures the type 490 SS system. Wescom's 490 Selective Signaling System supports SS-1, SS-1A and SS-4. The 490 system can accommodate one, two, and three-digit codes, but the FAA requirement is only for two or three digit dial codes. When a subsidiary of Rockwell in May 1994, Wescom had discontinued their SS equipment. However, as a subsidiary of Charles Industries, Wescom is still actively selling its SS equipment as of November, 1994 and plans to continue the product as long as sales continue.
AT&T	AT&T declared that it would discontinue support of its SS-1 and SS-1A equipment at the end of 1994. The FAA has bought old AT&T SS equipment from AT&T for the cost of $1.
"Telco" (AT&T):	Telco is buying Tellabs gear for new installations.
Canada:	Uses Tellabs 334 SST for newer applications. Used Wescom on older switches (ICCS) which was built by Rockwell Collins.
Ameritec:	Their AM8a SS text set has programmable tolerances.

Examples of voice switches (that was referred to by forward) currently used by the FAA that employ SS include:

a) The Western Electric Company (WECO) 300 switch used at ARTCCs. The WECO 300s are scheduled to be replaced by the VSCS, which also supports SS.
b) WECO 301A used at Terminal Radar Approach Control facilities (TRACONs).
c) ICSSs used in TRACONs and Air Traffic Control Towers (ATCTs).

Table 1-1 below lists FAA voice communications systems that do require and do not require support of the SS protocol.

Table 1-1 FAA Voice Communications Systems Requirement to Support SS

Required to support SS	Not required to support SS
Enhanced Terminal Voice Switch (ETVS)	Some administrative Private Branch Exchanges (PBXs)
Integrated Communications Switching System (ICSS)	
Rapid Deployment Voice Switch (RDVS)	
Small Tower Voice Switch (STVS)	
Traffic Management Voice Switch (TMVS)	
Voice Switching and Control System (VSCS)	
WECO (Western Electric Company) 300, 301A	