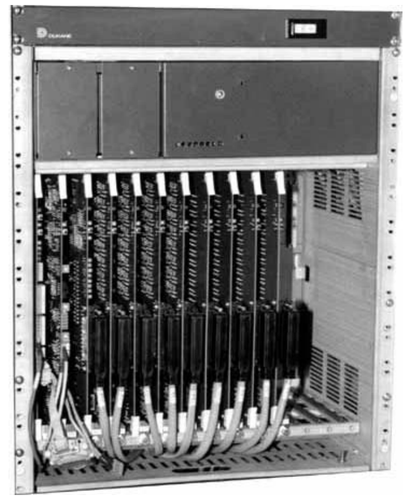
StarCall Feature Packages



StarCall Integrated Intercom System

Overview

StarCall Integrated Intercom System

The Edwards StarCall System is a Local Operating Network (LON®) based, multichannel, microprocessor-controlled communications system. The system is capable of simultaneously handling telephone conversations, intercom, program, and page distribution using standard DTMF telephones with electronic ringers, Edwards Model 7A1110 Administrative Telephones (ATELs), and standard speakers. Configuration versatility is built into the StarCall System, allowing different speaker-to-telephone ratios. The maximum system capacity is 512 speaker stations in increments of 16, or up to 512 standard telephones in increments of 16. The system also accommodates up to 320 administrative telephones with additional features, in increments of eight. Administrative telephones can be mixed with standard telephones and speaker stations in any combination of speaker stations and phones up to a maximum capacity of 512 total ports. The use of option cards that provide additional system features will affect overall system capacity.

Enhanced Intercom Feature Package

The Enhanced Intercom Feature Package provides the following system features:

- Timed distribution of event types that pertain to program sourc-such as music
- Scheduled time event delay during page
- Selectable intercom listen frequency response
- Automatic redirection of call-ins based on time of day (night mode)
- Temporary speaker exclusion control from an associated telephone

Call Management Feature Package

Call Management Feature Package for the StarCall® System is a software feature package that provides area code restriction that can be enabled or disabled for each individual trunk port. Area code restricted trunk ports can be configured with up to 16 area codes in the school's surrounding area to allow faculty and staff telephone contact with parents at home or at work. This area code list is global and applies to all trunk ports. Toll-free 800 and 888 numbers must be included within this list of 16 area codes if calls to those area codes will be allowed. Account codes are provided for calls outside the list of permitted area codes. Call Management Feature Package 1 accommodates up to 64 different account codes for use by faculty and/or staff.

437-00120A Remote Programming And Diagnostics Package (RAPID)

The Edwards Model 437-00120A Remote Programming and Diagnostics software allows configuration of system operational parameters using a PC. The program has an intuitive Windows-based graphical user interface. All system programming can be done off-line and saved to a DOS file. The RAPID software program requires an IBM or compatible PC running on Windows 3.1 or higher. System configuration and diagnostics can also be accomplished from a remote PC via modem. Both the local (PC) and remote modems must be Hayes compatible.

Telephone Feature Package

Telephone Feature Package allows standard telephones to place station-to-station calls, place three-party conference calls, transfer a call to another party, place calls on hold, and place emergency voice pages. These telephones must be standard 2500-set DTMF telephones with electronic ringers, such as the Edwards Model 7A1111A. This feature package also allows the telephones to receive speaker station call-ins. Three-party conference calls require that the Edwards Model 110-3524C Audio Routing Card already be installed in the system. Telephone Feature Package provides the following system features:

- Telephone-initiated voice page (by zones, by rooms, or by all call)
- Automatic associated room speaker muting during telephone voice paging
- Automatic or manual transfer of intercom between telephone and associated speaker
- Telephone-initiated tone distribution
- Telephone-initiated program distribution

Security Feature Package

Security Feature Package provides the following features and requires the presence of the Telephone Feature Package:

- Standard telephone-initiated emergency voice page
- Automatic muting of associated room speaker during telephone emergency voice paging
- Standard telephone off-hook duress call-in
- Standard telephone off-hook telephone call
- Standard telephone-initiated off-hook page
- Automatic call-in activation of event (Requires CPC-E or CPC2)
- Manual activation of remote event (Requires CPC-E or CPC2)

Standard Features

Enhanced Intercom Feature Package

- Timed distribution of program sources
- Automatic time scheduled event delay during page
- Loudspeaker audio reception filter setting
- Automatic night-mode call-in coverage
- Temporary page, program, and tone speaker exclusion

Telephone Feature Package

- Call holding
- Call transfer
- Station-to-station dialing
- Three-party conference calls
- Speaker station call-in to telephone
- Voice paging by: all call; zones; rooms
- Speaker mute at page source location
- Speaker intercom-to-phone toggle
- Initiate tone or program distribution to zones, rooms, or all

StarCall Integrated Intercom System

- Advanced multiprocessor design
- System configuration by PC terminal
- Optional built-in user-programmable master clock
- Modular software feature packages
- Automatic daylight savings time and leap year adjustments
- Simultaneous phone, intercom, program, and page distribution
- PBX and CO interface
- Compatible with standard 2500 DTMF phone instruments with electronic ringers
- Rack-mounted central equipment cabinet

Management Feature Package

- Area code restriction
- Account codes
- Attendant rollover
- Direct dial to foreign systems
- Telephone call pickup
- Message waiting indication
- Privacy indication

437-00120A Remote Programming And Diagnostics Package (RAPID)

- Windows®-Based
- Off-Line Programmable
- Remote Programming via Modem
- Intuitive Graphical User Interface (GUI)
- Tools for monitoring and diagnosing system activity

Security Feature Package

- Telephone-initiated emergency page
- Off-hook duress call-in
- Off-hook telephone call
- Off-hook page
- Automatic call-in activation of event
- Manual activation of remote event

Application

StarCall Integrated Intercom System

System traffic is managed over global telephonic links (16 basic, 32 expanded), one to four high power intercom channels, and one or two page/program distribution channels. StarCall allows selection of up to 26 distinct tone events for system use, each of which can be programmed from a selection of up to 22 tone types and two program sources. A paging microphone input is also provided. Each Intercom Amplifier Module (IAM) is capable of 15 watts output at 25 volts.

The StarCall System is available in a rack-mount configuration. This configuration includes a primary shelf (Model 110-3546A) and up to two expansion shelves (Model 110-3547A). The rack-mount primary shelf has 14 card slots. Slot 1 is reserved for the Central Processor Card (CPC) and slot 2 for the Audio Routing Card (ARC), leaving 12 slots for line and option cards. The expansion shelf, which does not require a CPC or ARC, has 14 slots available for line and option cards. Both primary and expansion shelves have space for up to two IAMs. The expansion shelf (Model 110-3547A) comes with cables for interconnection to a primary shelf or to another expansion shelf.

StarCall's extensive hardware modularity is accomplished through plug-in circuit cards. The CPC manages the connections between the different types of line cards and the ARC. Data communication between cards is managed over a LON. Each card slot has a fixed node address that is constructed during system configuration. This provides "plug and play" installation of the line cards. The CPC also has an RS-232 port to configure the system through a PC. A modem may also be used for this purpose. System configuration is accomplished by using an IBM PC or compatible PC running Windows version 3.1 or higher with the RAPID (Remote Programming and Diagnostics) PC programming application. The remote modem is required to be Hayes compatible.

The CPC contains the system software, which controls all StarCall functions and features. System software can be upgraded from a PC, either directly or via modem, using the RMU (Remote Maintenance Utility) PC application. RMU runs on an IBM PC or compatible running Windows version 3.1 or higher connected to the CPC via the RS-232 port. System software upgrades can also be accomplished from a remote PC via modem. The remote modem is required to be Hayes compatible.

All of the following features as well as Feature Packages are included in the base system software.

- Set system time, date, and schedule
- All administrative telephone intercom features
- Alphanumeric room numbers
- Normal call to emergency call upgrade
- Seven call-in priority levels plus privacy
- Distinctive ring tones for standard and emergency calls
- Emergency call-in to specified emergency stations
- Thirty-two multipurpose zones
- Thirty-two call destination groups
- Remote zone page microphone
- User-selective call answer from ATEL
- Up to three call-in switches plus privacy and call assurance LED on a single pair of wires
- An RS-485 port for interface to external devices
- A total of 12 I/O ports arranged as follows:
- Two SPDT relay ports
- Two multifunction open collector driver outputs (can be configured for digital secondary clock control)
- Four multifunction dry contact input ports
- Four ports that can be used as open collector driver outputs or as additional input ports
- Attendant call waiting
- Trunk access by trunk group
- Optional daylight savings time
- Dial access using password
- Call-in recall
- RDU350 tone enable
- Programmable hookswitch flash

When attendant call waiting is enabled and all attendants are busy, the primary attendant receives a call-waiting ring burst on their ATEL when another incoming call arrives. This feature requires a StarCall Administrative Telephone and a Trunk Interface Card (TIC). Attendant call waiting is activated using the RAPID software.

Trunk access by trunk group provides users with the option of dialing 9 or a series of two digits, 80 through 89, to access a trunk port. This allows users greater flexibility to specify which trunk port they wish to access. A typical use is selecting a specific facility for least cost routing when calling a nearby community. This capability functions when the StarCall trunk ports are connected directly to the public telephone network. It also functions when StarCall is interconnected through a KSU or PBX when those systems are configured to access, either physically or by dial access, a specific port from a specific extension. By selecting a specific trunk port through StarCall, the user selects a specific extension on the KSU or PBX.

Optional daylight savings time allows users in locations where daylight savings time (DST) is not used to disable the automatic switchover to DST. The default configuration is automatic DST switchover enabled.

Dial access using password allows users to override dial access restrictions that are set using RaPiD by entering a password.

Call-in recall allows the programming of call-in reminder time when the call-in point remained active after a call-in was answered. If this occurs, the call-in automatically reappears at the assigned administrative phones. This function can also be disabled system-wide. The call-in point must be cleared before a new call-in can be placed. The default is system-wide automatic recall enabled. Programming of this feature requires RAPID.

The RDU350 remote display unit's tone enable provides control of its internal tone generator. The RDU350 can sound when a call-in is received at the ATEL or standard telephone associated with the RDU. Additional devices may be controlled by an RDU350 relay contact independent of the internal tone generator.

Programmable hookswitch flash allows the programming of momentary hookswitch actuation ("flash") duration. This is the maximum time the handset can be on hook or the hookswitch manually actuated and still be interpreted as a hookswitch flash signal. The range is adjustable from 0.8 to 1.5 seconds. Programming of this feature requires RAPID.

The Audio Routing Card (ARC) card manages the system's common shared resources.

- Four DTMF receivers
- Four call progress tones
- Twenty-five tone types, including wail, warble, and chime
- Four intercom amplifier module
- Two paging/program source inputs
- One microphone input
- Two 3-party phone conference circuits
- Thirty two global telephonic links

The Administrative Telephone Card (ATC) manages data and audio for the ATELs. The ATC-E4 supports four ATELs on 32 system links. The field wiring is terminated at the administrative telephones through a customer-provided distribution punch block. Each ATC includes a 12-foot (3.7 m) pigtail cable with connector. The connector plugs into the receptacle on the ATC. Power (+24Vdc) for the ATELs is bussed separately, not within the pigtail cables. Each ATEL requires two shielded 22AWG pairs for data and an 18AWG pair for power. DC power for the ATEL is brought to the punchblock by a separate pair of customer-provided wires. Each shelf power supply can support up to eight ATELs. Every additional group of 24 ATELs requires an additional power supply (Model 17A365). The use of one or more remote display units (RDUs) reduces this capacity. An RDU uses approximately twice the power of an ATEL. For example, each shelf power supply can support up to six ATELs and one RDU.

The STC-E Standard Telephone Card manages ring state and off-hook detection for up to 16 standard 2500 DTMF telephones with electronic ringers. The field wiring is terminated at the telephones through customerprovided distribution punch blocks. Each STC includes a 12-foot (3.7 m) pigtail cable with a connector that plugs into the receptacle on the STC. The STC-E supports 32 system links. The telephone wiring must be shielded twisted pair.

The Balanced Telephone Card (BTC) provides the same functionality as the STC, supporting up to 16 standard 2500 DTMF telephones with electronic ringers. The BTC-E supports 32 system links. BTCs provide superior noise rejection due to the balanced nature of the signal pair. Each BTC includes a 12-foot (3.7 m) pigtail cable with a connector that plugs into the receptacle on the BTC. BTC telephone wiring requires only twisted pairs.

The Trunk Interface Card (TIC) provides StarCall with trunk ports for connection to loop start central office trunks or to KSU/PBX extension ports. The TIC-E4 supports four TIC ports on 32 system links. The field wiring is terminated at the TIC via RJ-11 telephone jacks mounted on the card edge.

The Audio Switching Card (ASC) manages 16 speaker station ports providing call-in and privacy. There are two 25-volt high-level audio buses on the ASC-E card. Each card requires two pairs of wire per speaker station port, one for audio and one for signaling. The ASC-E supports three call-in switches, privacy, and a call assurance LED. Although not required, it is recommended that the audio pair from the ASC to the speaker be shielded, 22AWG twisted pair. Field wiring connects to an ASC with a 12-foot (3.7 m) pigtail-to-connector cable assembly.

Each call switch can be assigned one of seven priority levels and up to 32 call-in destination groups. There are three priority levels for emergency calls and three for normal calls. The seventh priority allows remote call cancel. A normal call-in may be upgraded to an emergency call by rapidly pressing the call button twice. Each speaker station can be a member of up to 32 multipurpose zones through system configuration programming. There is also a permanent global page/program distribution room speaker exclusion table. This table allows the exclusion of certain speaker stations from all call, page, time signal, and program distribution.

A speaker station may be temporarily excluded from all paging, timed events, and program distribution by telephone command for up to 24 hours. Emergency pages cannot be excluded. Exclusion can be canceled manually using the telephone, or will be automatically canceled by the system on the change of day. Temporary speaker exclusion from a classroom telephone requires installation of Enhanced Intercom Feature Package 1.

The IAM provides 15 watts of intercom to a 25-volt speaker. Audio direction switching can be VOX or push-to-talk. A frequency response setting of the intercom listen is also provided to filter low frequencies and improve talkback from acoustically active environments. Each speaker station can be individually configured for either a high or low frequency response. This listen tone filter requires installation of the Enhanced Intercom Feature Package 1. All IAMs include two cables for input and output connections.

An Edwards Model 1A4060 Power Amplifier is used to provide 60 watts of power for paging and program distribution using 25-volt distributed audio. Larger amplifiers such as the Edwards Model 1A4125 (125 watts) or 1A4250 (250 watts) will be substituted where required. All page/program amplifiers require two cables each. Factorybuilt systems include these cables. They can be ordered separately for field-assembled systems. Each cable includes a connector on one end to mate with the corresponding connector on the primary or expansion shelf.

Enhanced Intercom Feature Package

The feature package provides for the distributing of a program source through an automated time event. This feature can be used to play music during preprogrammed time periods. Time events are programmed as either a high or low priority. A time event programmed as a low priority will always be delayed until after a paging event concludes. The "Scheduled time event delay during page" option delays high priority events until the completion of a page. For example, an announcement can be completed before a programmed tone or bell sounds.

Each loudspeaker port can be individually programmed for the best intercom listening quality. Bass audio from the speaker can be automatically removed whenever the desired speaker is selected for intercom handset operation. This feature limits low frequencies and improves the listen intelligibility of audio received from locations with poor acoustics, including echo and reverberation.

The enhanced intercom feature package also allows call-in coverage to be redirected to an assigned coverage group at a programmed time of day. This provides the capability for an automatic night-mode operation.

Standard telephone users may temporarily exclude the associated loudspeaker from time, paging, or program distributed events. This temporary exclusion capability lasts up to 24 hours, can be manually canceled by dial command, and automatically cancels with the change of day. Emergency priority pages override all exclusions.

Telephone Feature Package

Telephone Feature Package allows the system to be programmed so a telephone will toggle an associated speaker intercom call to that phone when it goes off hook. This can also be done by DTMF dial command. The user can transfer the connection back from the phone to the loudspeaker by dial command. A telephone is associated with a speaker when both are programmed with the same dial number.

Standard telephones can be used to initiate a voice page to specific rooms, to specific zones, or to all rooms. The loudspeaker associated with a telephone is automatically muted when that telephone initiates a voice page. A telephone can also be used to initiate a tone or program source to specific rooms, to specific zones, or to all rooms. Standard telephones must be 2500-set DTMF telephones with electronic ringers, such as the Edwards Model 7A1111A.

Call Management Feature Package

Attendant rollover automatically directs attendant telephone traffic to the next designated telephone. All telephone calls from all Trunk Interface Card (TIC) ports are delivered to the same attendant telephone. If the attendant telephone is busy, incoming calls will rollover to the next designated telephone. If that telephone is busy, the call rolls over to the next designated telephone, etc. Up to eight telephones can be designated. This feature applies to calls from central office lines only and requires connection to a TIC.

Direct dial to foreign systems allows any StarCall telephone to dial any extension, regardless of whether the called telephone resides on a StarCall system, a key system unit (KSU), or a private branch exchange (PBX). Duplicate extension numbers across systems are not allowed and the number of digits must be the same in both systems. Call Management Feature Package allows the StarCall system to automatically route the call through TIC ports to the correct extension on the other system.

Telephone call pickup is either by telephone extension or by call destination group. For example, if there are two StarCall telephones in the same office area and a telephone call rings one of those telephones and that person is not at their desk, another person in the office can receive that call, eliminating the need to physically move to the other desk to answer the call.

Message waiting indications can be sent to specific call assurance LEDs using a dialup code. Upon seeing the message waiting indication, a user can dial the office and retrieve their message. This function is a global system attribute and requires the presence of optional Call Assurance indicators in the individual rooms.

The privacy indication flashes the Call Assurance LED whenever there is an open intercom path to that room. The privacy tone is also periodically sounded. This function requires the presence of optional Call Assurance indicators in the individual rooms.

Security Feature Package

Emergency voice paging allows any standard telephone to initiate an emergency voice page. Any loudspeaker associated with a telephone is automatically muted to eliminate feedback when that telephone initiates an emergency voice page.

Any telephone port can be configured to automatically generate a duress call-in after the telephone has been left off-hook for the programmed period of time. When any designated Administrative Telephone answers an off-hook duress call, an intercom connection is established with the calling telephone's associated speaker. This allows hands-free response from the origination point of the call-in.

Telephone ports can be configured to automatically generate a telephone call to a designated telephone extension upon going off hook. Off-hook calling can be globally configured or individually configured by port. A telephone port can be configured to automatically initiate a voice page to zones when off-hook. This voice page can be configured as a normal or emergency priority.

The security package allows automatic generation of dry contact closures by programmed call-in events and/or manual generation of dry contact closures from administrative or standard telephones. These contact closures can be used to activate events such as remote door lock control or security camera control. Standard telephones must be 2500-set DTMF telephones with electronic ringers, such as the Edwards Model 7A1111A.

System Functions

StarCall Integrated Intercom System

All system functions are software controlled and are included in the basic StarCall System. Some system functions require installation of one or more of the expanded circuit cards.

All audio functions in the system operate within the following priority scheme. Emergency intercom has the highest priority. Additional audio functions in descending order of priority are: emergency page, civil emergency, manual time tone and high priority event tone, all call and zone page, intercom, custodial tone, low priority event tone, and audio program distribution. A lower priority event may be interrupted by a higher priority event if necessary. Interrupted lower priority functions will be restored after the higher priority function ends. Telephone conversations will not be interrupted by the above listed functions.

A telephone is associated directly with a speaker station by assigning the same dial number to both.

A KSU/PBX interface with StarCall can be achieved through one of two methods. The first method is through a loop start trunk port on the KSU/PBX to a StarCall STC or BTC port. The second method is through an extension port on the KSU/PBX to a trunk port on a StarCall TIC.

The system is capable of terminating 32 loop start central office trunks or KSU/PBX extensions using TICs.

A real-time clock is capable of maintaining system time for 10 years in the event of a power failure. The real-time clock automatically adjusts for daylight savings time and leap years. All ATELs display the time when there are no call-ins present and the unit is not in use. This display may be configured for 12 or 24-hour mode. The day of the week and date are also displayed.

A StarCall System with the CPC-E has 12 I/O ports configurable for special purpose input and output functions. Two ports are SPDT relay contacts. Two ports are open collector output ports. Four ports are dedicated dry contact inputs. The remaining four ports are open collector output ports that are user-configurable to loop back as an input, or to act as additional dry contact input ports. Either relay contact port can be configured for use as an analog clock control. Any of the open collector output or I/O ports defined as an output can be configured for digital clock control (requires two).

High-level intercoms to 25-volt loudspeaker assemblies are supported by the IAM. This module supplies 15 watts of power with voice operated switching (VOX) or push-to-talk operation. It is recommended that each speaker port use one shielded pair, 22AWG minimum for the audio wire pair. Up to three call-in switches, privacy, and a call assurance LED are supported over the call-in wire pair. Each of the call-in switches can be assigned one of seven priorities.

StarCall has a page microphone input configurable for one to 32 possible zones. ATEL or standard telephone initiated voice paging can be distributed via all call or any combination of 32 possible multipurpose zones. Standard telephone initiated paging is part of Telephone Feature Packages.

Each standard telephone location can be customized by programming the following attributes:

Telephone Extension Number—3, 4, or 5-digit alphanumeric numbering can be used

Dial Access—Determines if calls can be initiated from this telephone.

Ring Access—Specifies whether this telephone annunciates calls by ringing.

Page Access—Specifies whether all call and zone pages can be made from this telephone. (Paging from a standard telephone is part of Telephone Features Package.)

Emergency Ring Access—Determines whether this telephone can receive emergency call-ins from speaker stations.

Normal Ring Access—Determines whether this telephone can receive normal call-ins from speaker stations.

Trunk Access—Determines whether this telephone can call outside StarCall with or without dialing a password.

Off-Hook Duress—Specifies whether this telephone initiates a call-in for its associated speaker station when going off-hook for a programmable period of time without dialing. Off-hook duress feature is part of Classroom Telephone Feature Package and Security Feature Package.

Intercom Off-Hook Answer—Specifies whether this telephone answers a call-in by going off-hook or whether an answer code must be dialed after going off-hook.

Intercom Speaker to Handset Transfer—Specifies whether the call at the speaker station is transferred to this telephone immediately when telephone goes off hook, or whether transfer code must be dialed after going off-hook.

- Attendant call waiting
- Trunk access by trunk group
- Optional daylight savings time
- Dial access using password
- Call-in recall
- RDU350 tone enable
- Programmable hookswitch flash

The ATEL has all the features of the standard telephone plus the following: special function keys, 16-character alphanumeric display, and hands free operation. Each displays the day of week, date, and time in the idle state, or displays the dial number, priority level, position, and number of calls in the queue when call-ins are present. The ATEL produces a distinctive ring for emergency calls.

All port locations may be assigned on a system-wide basis and can be assigned three-, four-, or five-character full alphanumneric dial numbers.

Engineers' Specifications

Call Management Feature Package for the StarCall System

It shall be possible to order the call management feature package as a part of a new system or add it to an existing system.

The call management feature package shall allow call restrictions that can be enabled or disabled for each individual trunk port by area code. It shall be possible to configure up to 16 area codes that can be dialed from the system. This area code list shall be global and apply to all trunk ports. Toll-free 800 and 888 numbers must be included within the global list of 16 area codes if calls to those area codes will be allowed. Account codes shall be provided for calls outside the list of permitted area codes.

The call management feature package shall accommodate up to 64 different account codes for use by faculty and/or staff. The call management feature package shall provide an attendant rollover feature to automatically direct attendant telephone traffic to the next designated telephone. All telephone calls from all TIC ports delivered to the same attendant telephone shall rollover to the next designated telephone if the attendant telephone is busy. If the next telephone is busy, the call rolls over to the next designated telephone, etc. It shall be possible to designate up to eight attendant rollover telephones.

The call management feature package shall allow any telephone in the system to direct dial any extension, regardless of whether the called telephone resides on a StarCall system a KSU, or a PBX. Duplicate extension numbers across systems shall not be allowed and the number of digits must be the same in both systems. The call management feature package shall allow the StarCall system to automatically route the call through a TIC port to the correct extension on the other system.

The call management feature package shall provide telephone call pickup either by telephone extension or by call destination group. It shall be possible to receive a call ringing on another telephone in the same area, eliminating the need to physically move to the other telephone to answer the call.

The call management feature package shall send message waiting indications to specific call assurance LEDs. This function shall be a global system attribute and require the presence of optional Call Assurance indicators in the individual rooms.

The call management feature package shall flash the Call Assurance LED whenever there is an open intercom path to the room. The privacy tone shall also sound periodically. This function shall require the presence of optional Call Assurance indicators in the individual rooms.

StarCall Integrated Intercom System

The facility communications system shall be an Edwards StarCall System or approved equal. The system shall be a local operating network (LON) based, multichannel, microprocessor-controlled communications system. The system shall be capable of simultaneously handling telephone conversations, intercom, program, and page distribution using standard DTMF telephones with electronic ringers, Edwards Model 7A1110 Administrative Telephones (ATELs), and standard 25V speakers. The system shall allow different ratios of speaker stations to telephones and not require that a speaker station be associated with a telephone to provide two-way voice communication.

The system shall have a maximum capacity of 512 speakers in increments of 16 with one Administrative Telephone Card, or up to 512 standard telephones in increments of 16. The system also shall accommodate up to 320 administrative telephones in increments of eight. The system shall allow up to eight administrative telephones to be mixed with standard telephones and speaker stations in any combination of speaker stations and telephones up to a maximum system capacity of 512 ports.

The system shall provide up to 32 global telephonic links, up to four high power intercom channels, and two page/program distribution channels. A paging microphone input shall also be provided. The system shall provide up to 26 distinct tone events, the tones for which are chosen from up to 22 tone types and two program sources.

The system shall be rack mounted and have a capacity of 14 printed circuit cards. The system shall reserve slot 1 for a Central Processor Card and slot 2 for an Audio Routing Card. The system shall contain a shelf assembly with a backplane, power supply, and card cage. The power supply shall be capable of powering a fully populated shelf, plus the equivalent of up to eight ATELs. Additional ATELs shall be powered by 24Vdc, provided by power supplies included with the expansion shelves or auxiliary power supply. Both primary and expansion shelves shall have space for up to two Intercom Amplifier Modules (IAMs). The expansion shelf shall provide cables for interconnection to a primary shelf or another expansion shelf. Each expansion shelf shall provide additional capacity for 14 printed circuit cards. Each expansion shelf power supply shall be capable of supporting the equivalent of up to eight ATELs. A remote display unit (RDU) shall consume approximately the same power as two ATELs.

The system shall have a Central Processor Card (CPC) managing the connections between the different types of line cards and an Audio Routing Card (ARC). Data communication between cards shall be managed over a LON network. Each card slot shall have a fixed node address, constructed during system configuration, to provide “plug and play” installation of the line cards. The system shall be configured through the CPC, either by a PC connected to the RS-232 port or by modem. System configuration can be done using an IBM PC or compatible running Windows version 3.1 or higher. The PC programming application shall be RAPID (Remote Programming and Diagnostics). The system configuration software must support Hayes modems or equivalent for communication with a remote PC.

The CPC shall have a total of 12 I/O ports arranged as follows:

- Two SPDT relay output ports

Two multifunction open collector (O.C.) driver outputs (shall be usable for digital secondary clock control)

Four multifunction dry contact input ports

Four ports that can be used as open collector driver outputs or as additional input ports

The CPC shall contain the resident system software and provide the features listed below:

Set system time, date, and schedule

All administrative telephone intercom features

Alphanumeric room numbers

Normal call to emergency call upgrade

Seven call-in priority levels plus privacy

Distinctive ring tones for standard and emergency calls

Emergency call-in to specified emergency station

Thirty-two multipurpose zones

Thirty-two call destination groups

Remote zone page microphone

User selective call answer from ATEL

Telephone interface capability

Attendant call waiting

Trunk access by trunk group

Optional daylight savings time

Dial access using password

Call-in recall

RDU350 tone enable

Programmable hookswitch flash

System software shall be capable of being upgraded from a PC, either directly or by modem, using the PC application RMU (Remote Maintenance Utility). RMU shall run on an IBM PC or compatible running Windows version 3.1 or higher connected to the CPC using the RS-232 port. It shall also be possible to perform system software upgrades from a remote PC via modem. The remote modem must be Hayes compatible.

The ARC shall manage the system's common shared resources.

Four DTMF receivers

Four call progress tones

Twenty-five tone types, including wail, warble, and chime

Four intercom amplifier module

Two paging/program source inputs

One microphone input

Two 3-party phone conference circuits

Thirty two global telephonic links

The Administrative Telephone Card (ATC) shall manage data, audio, and power for up to four Edwards Model 7A1110 Administrative Telephones (ATELs). Each ATC shall have a connector and include a 12-foot (3.7 m) pigtail-to-connector cable assembly.

The Standard Telephone Card (STC) shall manage ring state and off-hook detection for up to 16 standard 2500 DTMF telephones with electronic ringer. Each STC shall have a connector and include a 12-foot (3.7 m) pigtail-to-connector cable assembly.

The Balanced Telephone Card (BTC) shall provide the same functionality as the STC, supporting up to 16 standard 2500 DTMF telephones with electronic ringers. BTCs shall provide a balanced signal pair with superior noise rejection. Each BTC shall include a 12-foot (3.7 m) pigtail cable with a connector that plugs into the receptacle on the BTC. BTC telephone wiring shall require only twisted pair.

The Trunk Interface Card (TIC) shall provide StarCall with trunk ports for connection to loop start central office trunks or to KSU/PBX extension ports. A basic TIC-B shall support two TIC ports on 16 system links, while the expanded TIC-E4 shall support four TIC ports on 32 system links, and the expanded TIC-E8 shall support eight TIC ports on 32 system links. The field wiring shall be terminated at the TIC via RJ- 11 telephone jacks mounted on the card edge.

The Audio Switching Card (ASC) shall manage 16 speaker station ports and provide call-in and privacy. The ASC shall provide two
25-volt high level audio buses. The speaker station line connections shall be supervised for detection of three levels of user-configurable call switch signaling in addition to privacy. The ASC shall allow each call switch to be assigned one of seven priority levels and up to 32 call-in destination groups. The ASC shall provide three priority levels for emergency calls and three for normal calls. A seventh priority shall be provided to allow remote call cancel. The system shall upgrade a normal call-in to an emergency call when the call button is pressed twice within two seconds. It shall be possible to assign speaker stations to any one or more of up to 32 multipurpose zones through system configuration programming. The system shall provide a permanent global page/program distribution room speaker station exclusion table to allow the exclusion of certain speaker stations from all call, page, and program distribution.

Systems shall provide for the installation of up to four IAMs in the shelves, with each IAM providing 15 watts of intercom to a 25-volt speaker. The IAMs shall be compatible with both VOX and push-to-talk audio direction control.

Systems shall allow Edwards power amplifiers to be mounted for paging and program distribution using 25-volt distributed audio.

The system port capacity shall be 1024 ports for CPC2 based systems, and 640 ports for CPC-E based systems, according to the card type:

STC, BTC, and ASC cards, 16 per card

ATC cards, 4 per card

CTC cards, 2 or 4 per card

ICC and OCC cards, 48 per card

The system shall have 32 trunk ports, according to the card type:

TIC and COC cards, 4 or 8 per card

A system shall provide 40 available card slots in a three-shelf rack system.

A system shall have two paging/program channels, and may select from two program sources.

A system shall provide 26 programmable tone event types.

The system shall provide the following audio function priorities listed in descending order of priority:

Emergency intercom

Emergency page

Civil emergency

Manual time tone and high priority event tone

All call and zone page

Intercom

Custodial tone

Low priority event tone

Program distribution

A lower priority event shall be interrupted by higher priority event. An interrupted lower priority function shall be restored after the higher priority function ends. Telephone conversations shall not be interrupted by the above listed functions.

The system shall allow a telephone to be associated directly with a speaker station by assigning the same dial number to both.

A KSU/PBX intercom/paging interface to the system shall be achieved through a KSU/PBX loop start trunk port to an STC or BTC extension port.

A KSU/PBX telephonic-only interface to the system shall be achieved through a KSU/PBX extension port to a TIC trunk port.

The system shall provide the capacity to terminate up to 32 central office (CO) loop start trunks.

System time shall be maintained by a real-time clock capable of maintaining time for 10 years in the event of a power failure. The system real-time clock shall automatically adjust for daylight savings time and leap years. All ATELs shall display the time when no call-ins are present and the telephone is not in use. This display shall be configurable for 12 or 24-hour mode. The day of the week and the date shall also be displayed.

An expanded system shall provide 12 I/O ports configurable for special purpose input and output functions. Two ports shall be SPDT relay contacts. Two other ports shall be open collector output ports. Either of the relay contact ports shall be configurable for analog clock control. Four ports shall be dedicated dry contact inputs. The remaining four ports shall be open collector output ports, which are user-configurable to loop back as an input or act as additional dry contact input ports. Any of the open collector output ports or I/O ports defined as an output shall be configurable for digital clock control.

The system shall provide a page microphone input configurable for any combination of one to 32 zones. ATELs and STELs shall be capable of initiating voice paging that can be distributed through all call or any combination of 32 possible multipurpose zones.

The system shall allow each telephone's class of service to be custom configured with the following on/off attributes:

Dial access

Telephone ring access

Direct off-hook page or telephone call

Timed off-hook duress call-in

Page access

Direct off-hook intercom speaker to handset transfer

Emergency ring access

Normal ring access

Trunk access

Intercom off-hook answer

The system shall require a password to access restricted functions. The system shall also allow the following attributes to be individually configured for each telephone port:

Call destination group

Telephone extension number

The system shall establish an intercom path to the associated speaker (if present) of the off-hook telephone when a duress call-in is answered. The system shall establish an intercom path to the associated speaker (if present) of the telephone from which a call-in was made. All port locations shall be assigned on a system-wide basis and can be assigned three, four, or five-digit full alphanumeric dial numbers.

The system shall offer optional feature software packages for future upgrades.

Specifications for StarCall Integrated Intercom System	
Rated Audio Output	25V Line, 15W Intercom, 60W to 250W Program/Page
Interfaces	RS-232: PC or modem interface I/O Ports: Clock correction and special audio functions KSU/PBX: Loop start CO port to STC for telephone interface functionality, loop start TIC to KSU/PBX telephone port for voice communications
Terminations	All telephone and speaker station wiring terminates to a customer-provided punch block. Every line card except the TIC includes a 12-foot (3.7 m) pigtail-to-connector cable assembly. TICs have RJ-11 receptacles. Other line cards have receptacles and terminal strips where applicable.
Tone Events	Emergency page preannounce; Civil emergency; Auxiliary alarm; Emergency reminder tone; Page preannounce; Normal reminder tone; Custodial tone; Door tone; Intercom preannounce; Privacy tone; Event tones 1 through 16; Each tone event uses one of 22 tone types or two program sources.
Capacity	1024 ports for CPC2 based systems (640 ports for CPC-E based systems) total, according to the card type: STC, BTC, and ASC cards, 16 per card ATC cards, 4 per card CTC cards, 2 or 4 per card ICC and OCC cards, 48 per card 32 trunk ports, according to the card type: TIC and COC cards, 4 or 8 per card 4 IAMs

Ordering Information

Model	Description
StarCall Integrated Intercom System & Associated Equipment	
StarCall Integrated Intercom System	StarCall Integrated Intercom System
110-3547A	Expansion Shelf
110-3527A	ATC-E4 Administrative Telephone Card (4 ports)
110-3531A	STC-E Standard Telephone Card (16 ports, 32 links)
110-3533B	ASC-B Audio Switching Card (16 stations, 1 path, 2 channels)
110-3534A	ASC-E Audio Switching Card (16 stations, 2 paths, 6 channels)
110-3543	Ring Supply Module
110-3544C	IAM Intercom Amplifier Module
110-3551A	TIC-E4 Trunk Interface Card (4 trunks, 32 links)
110-3552B	TIC-E8 Trunk Interface Card (8 trunks, 32 links)
110-3592	Assembly – 70 in (178 cm) StarCall Floor Rack
110-3593	Assembly – 53 in (135 cm) StarCall Floor Rack
437-00131	StarCall Remote Maintenance Utility (RMU)
110-3521A	CPC-E Central Processor Card
110-3524C	ARC-E Audio Routing Card
110-3542	PSM Power Supply
110-3546A	Primary Shelf
110-3763A	CPC2 Central processor Card
110-3851A	CTC-2 Call Notification Telephone Card (2 ports)
110-3852A	CTC-4 Call Notification Telephone Card (4 ports)
110-3889A	NIC Network Interface Card
110-4001	COC-4 Central Office Card (4 ports)
110-4002	COC-8 Central Office Card (8 ports)
110-4003	TCM Trunk Caller-ID Module
437-00141	StarCall Operating Firmware Service Update
1A4060	60-Watt Power Amplifier
1A4125	125-Watt Power Amplifier (optional)
1A4250	250-Watt Power Amplifier (optional)
7A1110	Administrative Telephone
7A1111A	Classroom Telephone
9A4300	Call-In Switch with Call, Emergency, Cancel, and Privacy buttons, and Assurance LED
9A4301	Call-In Switch with Call button and Assurance LED
9A4302	Call-In Switch with Call and Cancel buttons, and Assurance LED
9A4303	Call-In Switch with Call and Privacy buttons, and Assurance LED
17A365	Power Supply (for Administrative Telephones and Remote Display Units)
77A1000	33.6K External Modem Kit (includes cable)
RTC350/TC350	AM-FM Tuner Cassette/Player
RTC350P/TC350P	AM-FM Tuner Cassette/Player with Mixer/Preamp
RDU350	Remote Display Unit

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