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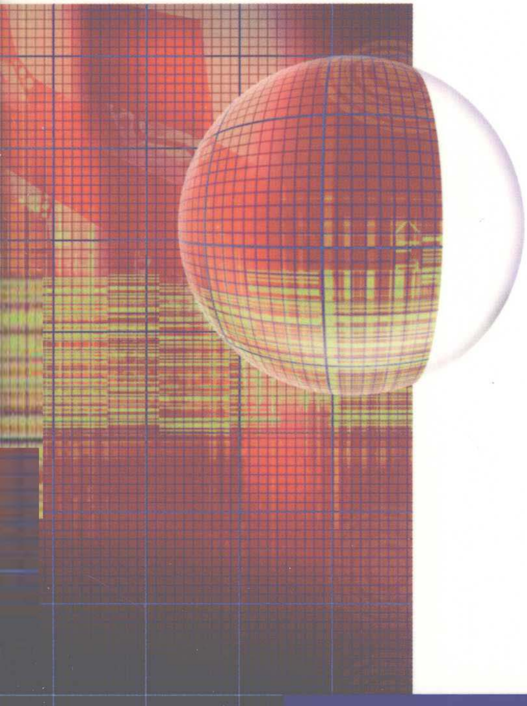
PBX SYSTEMS FOR IP TELEPHONY

Migrating Enterprise Communications



- Tap into the lost art of PBX system design
- Maximize performance capabilities of telephones, messaging, call distribution management, and networking
- Access valuable product feature and function matrices

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PBX Systems for IP Telephony

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From time to time solutions providers change the names of their products or bundle several products as a new offering. A new name does not imply that the original product is no longer available or is no longer supported by the manufacturer/developer. Since this book went to print Avaya has announced a number of name changes reflecting the evolution of their technologies, but will continue to use both old and new names for the convenience of long-time customers. If you have any questions about products mentioned in this book, please contact your Avaya sales representative.

CONTENTS

CHAPTER 1

Enterprise Communications Systems Today

Today's enterprise communications market is in a considerable flux caused by major ongoing changes in the technology of core products and the network infrastructure. Notably, voice communications systems are migrating from a time- to a packet-based switching and transmission design. The last major market and product shift occurred in the mid-1970s when the first computer stored program control (SPC) and digital switching communications systems were announced and shipped to replace older generation electromechanical systems. Although every generation believes that product upgrades and enhancements occurring in their prime are the most significant ever, telecommunications managers who remember the limited feature and function capabilities available on communications systems 30 years ago may be less impressed with the current market upheaval than industry newcomers who have learned to expect a new generation of products every 18 months from the data networking world.

Today's typical enterprise voice communications network includes many, if not all, of the following ingredients:

1. A core communications *switching system* (Private Branch Exchange [PBX] system, Key Telephone System [KTS], or KTS/Hybrid system) that provides dial tone, call setup and teardown functions, and more call processing features than any one customer is likely to use
2. A *management system* to support fault and configuration operations
3. A *call accounting system* that analyzes and processes call detail records to generate billing and traffic reports
4. A *voice messaging system* that offers a wide array of services far beyond a basic answering system

Other widely used products that support basic voice applications in the enterprise include automated attendants, paging systems, and voice announcers. It is naturally assumed, but sometimes overlooked, that each network system user has some type of desktop or mobile telephone to access the core communications switching system. Other stand-alone desktop equipment scattered around the enterprise is likely to include facsimile (fax) machines and modems for dial-up data network access.

Customers with call center system requirements will install, at a minimum:

1. An Automatic Call Distributor (ACD)
2. A Management Information System (MIS)

As the call center requirements become more sophisticated, subsystems and options will be added to the basic ACD. These might include an Interactive Voice Response (IVR) system, an Automatic Speech Recognition (ASR) system, or a Computer Telephony Integration (CTI) application server. Users now routinely expect that all of these call center system elements will gradually merge with the Web server and e-mail server to form a mixed media e-contact center.

Twenty years ago almost none of these products existed beyond the core communications switching system. Small-line-size customers during the early 1980s with basic voice communications requirements would have a KTS or perhaps one of the recently introduced KTS/Hybrid systems. Intermediate and large-line-size customers with more advanced requirements preferred a PBX system, although what counted as advanced capabilities at the time would include features and functions considered basic today, such as Direct Inward Dialing (DID), Call Detail Recording (CDR), and Automatic Route Selection (ARS). These features were once available on only large, sophisticated, and relatively expensive PBX systems, but they can now be found on KTS products targeted at very small customer locations. The trickle-down theory of KTS/PBX feature and function options says that optionally priced advanced features and functions designed primarily for customers of large PBX systems eventually become standard offerings on entry-level KTSs.

The number of available features on PBX systems has increased exponentially since the first SPC models were introduced in the 1970s. A leading-edge PBX system marketed in 1980 had a software package with about 100 features for station, attendant, and system operations. By 1990 the number of features had more than doubled. Today a typical PBX system boasts more than 500 features, including optional hospitality, networking, and ACD options, and today's typical KTS/Hybrid system offers more performance options than any PBX system in 1980. Despite the significant increase in features designed for desktop access and implementation, the majority of PBX station users (i.e., people with phones on their desks) use fewer than ten features on an everyday basis. Ironically, today's typical station user may use fewer features than he would have used 20 years ago because many once-popular features, such as call pick-up and automatic callback, are rarely implemented. One reason for the decline in use of once common desktop features is the prevalence of voice messaging systems that preclude the use of many manually operated features for call coverage situations.

As a result, today's PBX developers continue to write new feature software programs for the non-typical station user. Studies show that most station users implement about six features on an everyday basis, and features in general use are limited to hold, transfer, conference, and a few others. However, system designers cannot assume that the set of features in general use will be the same for every station user. Many features are used by a small number of system subscribers, but they are no less important than those used by the majority. For example, a feature such as Flexible Night Service may be used only by the system's sole attendant console operator, and the Recent Change History feature may be of value only to the system administrator, but these features are as vital to the few individuals who implement them as Call Forwarding is to a typical desktop telephone station user. Many of the hundreds of PBX features introduced during the past 20 years were developed at the explicit request of customers. When a customer or a small group of customers demanded a new feature, a PBX manufacturer first determined that anticipated demand justified its development. Once offered by a major manufacturer, the new feature soon became available on systems from most competitors.

It's important to note that some perfectly viable features are unique to special categories of customers or station users and may be used by as few as one system subscriber per enterprise. A feature's value is not determined solely by how many individual station users implement the feature, but also by potential cost savings and productivity improvement at a station, system, or network level.

Of course, most customers do not have stringent demands on PBX system design architecture attributes; they're looking for basic growth and redundancy requirements. A station user who doesn't have telecommunications system acquisition or management responsibilities cares nothing about the technical underpinnings of the telephone system he's using. He picks up the telephone handset, listens for dial tone, punches a number or activates a feature, and is satisfied by the experience almost every time. As long as that's true, the station user won't be asking whether the system has analog or digital transmission, circuit-switched or packet-switched connections, a proprietary software operating system, or a standard off-the-shelf Windows solution. People who *should* care about PBX system technology and reliability standards, applications support, and future product direction are the telecommunications manager, voice/data networks director, or CIO. This book is written for the individual who must know more about PBX systems than basic calling procedures, who must be able to configure and reconfigure the system when need demands, and who must know the answers when questions are asked.

The Fundamental Enterprise Communications Systems

Current voice communications systems comprise five fundamental product categories:

- Key Telephone System (KTS)
- Private Branch Exchange (PBX)
- Automatic Call Distributor (ACD)
- Voice Messaging System (VMS)
- Interactive Voice Response (IVR)

The first three categories support call processing and switching functions to enable telephone calls between two or more station users. The latter two product categories are designed to work in conjunction with one of the three core communications switching systems to provide optional services beyond a basic station-to-station call. It's possible to integrate voice messaging or IVR capabilities into a KTS, PBX, or ACD product design, but most companies don't. Instead, customers have traditionally chosen to install external, stand-alone systems for messaging and voice response applications. (Many small KTS products have an integrated voice messaging function, but the messaging features are typically not as robust as stand-alone offerings.)

KTS

A KTS is a customer-premises communications system designed to support basic voice applications and relatively small station user requirements. KTS got its name from its historical use with proprietary telephones, known as key telephones, which interface with a central control cabinet known as a Key Service Unit (KSU). The KSU is equipped with the system's call processing control, port interface cards, and a variety of system/service circuit cards, such as Dual Tone Multi Frequency (DTMF) receivers and Input/Output (I/O) interfaces. The KSU performs central office line connection, intercom functions, paging, and station connections. Its common control elements include a call processor and system memory databases, and its most important function is the provisioning of dial tone. Other basic call processing functions are call answering, dialing, and transaction features such as hold, transfer, call forward, and conference.

Oddly, not all KTS products require a KSU. Instead, the intelligence needed to perform call processing and switching can be built into the circuitry of each key telephone instrument. Such systems are easy to install and maintain, but usually have limited feature and function capabilities, and are acceptable only for customers with modest port-size requirements. The KSU-less system is usually installed to work behind Centrex services offered by local telephone operating companies, which provide the more advanced features and functions through their central office communications switching system.

Common to KTS telephone instruments are designated, programmable key buttons for making and receiving off-premises calls over telephone company line circuits (trunks). The term *KTS* takes its name from the telephone instrument keys (analogous to typewriter keys) integrated into the product design. The line keys for each telephone instrument have direct access to off-premises telephones. Because trunks are distributed over and shared across groups of select telephones, each station user's desktop instrument must be provisioned with multiple-line access keys. *This is the distinguishing characteristic of a KTS: station user selection and access to designated telephone company lines.* A pure KTS product supports only multi-line key telephone instruments and is incapable of supporting industry standard 2500-type single-line analog telephones unless a designated line circuit is dedicated to a single analog telephone instrument. Needless to say, it's unusual for a customer to configure a KTS in this manner. Small system customers who want station users with 2500-type telephones to have shared access to a pool of trunks do have an alternative: a KTS/Hybrid product (see below), designed to support a mix of proprietary multiple-line button phones and nonproprietary single-line analog phones.

PBX

PBXs aren't just big KTSs with more features and functions. The two share architecture elements, but PBX systems are designed for more robust functionality, greater growth capacities for ports, traffic, and call processing, and more levels of redundancy. PBX basic architecture includes the common control carrier/cabinet, port carriers/cabinets, and port circuit cards. Peripheral equipment support includes proprietary telephones, both single and multiple line, and industry standard single-line analog telephones. A critical discussion of PBX system design and feature capabilities is the primary objective of this book, and we'll return to it in subsequent chapters.

First, let's look at the major design and operational difference between a KTS and PBX. PBX stations can place calls over telephone company trunks only with the shared pool access method. The sole exceptions are telephones equipped with an optional private line for inbound and outbound calls. [Note: for reasons too complex to explain, the term *line*, when used in relation to a PBX system, usually refers to all customer premises equipment peripheral endpoints and is not a reference to a trunk circuit, as it would be with a KTS. The terms *station* and *line* are both used for PBX endpoints (telephones, modems, fax terminals) that are not off-premises trunk circuits.]

Hybrid System

Hybrid communications systems also share operating functions common to standard KTS and PBX systems. Original Hybrid systems were designed to more closely resemble KTS rather than PBX systems, although differences between the product categories are diminishing. Like KTS, Hybrid systems are based on a control cabinet similar to a KSU and can support a variety of port circuit boards for interfacing to station and trunk circuit equipment. All Hybrids support multiple-line proprietary key telephones and industry-standard single-line analog telephones. In a Hybrid system, phone access to line circuits is identical to that of a pure KTS, but single-line analog phones access a defined pool (group) of telephone company line circuits. *This latter design capability is what distinguishes a pure KTS from a Hybrid system.* The Hybrid's port-oriented architecture design permits custom configurations to suit specific business applications. The architecture and technology design foundations of current Hybrid systems are more similar to PBXs than to KTSs. In fact, features and functions are sometimes difficult to distinguish from more expensive PBX systems.

Unfortunately confusion reigns when it comes to product category typing a communications system as KTS, Hybrid, or PBX. Some manufacturers call their Hybrid offering a KTS/Hybrid and others may refer to it as a Hybrid/PBX. The naming issue gets interesting in the United States because the Federal Communications Commission classifies customer premises communications systems as either KTS or Hybrid based on how single-line telephones access the central office. If the phone can access only one line as programmed by the system administrator, the system is a KTS; if it can access a pooled group of lines, the system is considered a Hybrid. Some manufacturers may even register a single system as both because the call processing software allows configuration flexibility for either pooled- or single-line access from a single-line telephone. Note that designating the product as a KTS, Hybrid, or PBX system may have financial consequences based on telephone company jurisdiction because trunk tariffs for linking a customer premises communications system to the central office can differ between KTS and PBX. (This was more common 15 years ago than it is today.) Ultimately it is the local telephone company that defines the type of system the customer is seeking to connect to the central office.

ACD Systems

The central component of a customer call center is an ACD. ACD systems were originally developed to handle large volumes of incoming calls and automatically route them to designated answering positions. ACD systems are designed and customer programmed to satisfy higher quality of service standards than PBX systems for the following call processing functions:

- Screening
- Routing
- Queuing
- Answering

Most PBX systems can be programmed to function as ACD systems, but few ACDs can be programmed to function as PBXs and continue to support most of the latter product's standard or optional features and functions. Nevertheless, an ACD system shares many of the architecture and feature capabilities of a PBX system. You can think of it as a PBX designed for a very specific application—to distribute incoming calls equitably to a group or groups of answering stations. We usually call ACD answering stations *agents*, and *this is the fundamental difference between PBX and ACD system service: calls handled by a PBX are routed to a specific station user*, whereas ACD calls are routed to a group of stations, although call analysis programs can be used to route the call to a specific agent in a group.

ACDs exhibit several architecture design and feature standards that are often not adhered to for PBXs. A true ACD system is based on a nonblocking switch network design, has sufficient call processing power to handle a large volume of complex call types, and has software program-